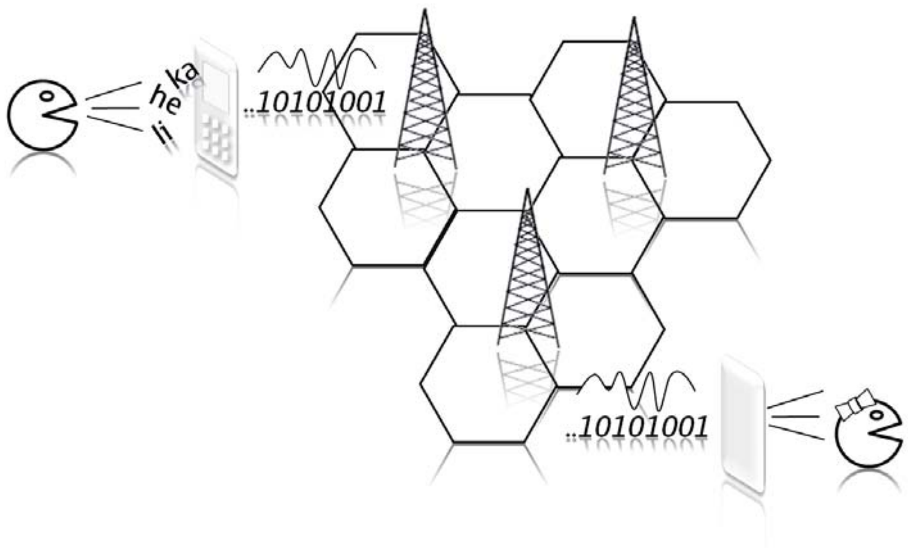
JYVÄSKYLÄ STUDIES IN COMPUTING
126

Kari Aho

Enhancing System Level Performance of Third Generation Cellular Networks Through VoIP and MBMS Services



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ABSTRACT

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The purpose of this thesis is to address variety of performance enhancements and configurations in terms of 3G/3.5G cellular networks. *Multimedia Broadcast Multicast Service (MBMS)* and *Voice over IP (VoIP)* are used to benchmark different features as they are undeniably key concepts also in the wireless cellular networks and thus securing their performance can be considered as critical. From practical perspective, wireless networks set a very strict environment for different services through additional delays, packet loss, jitter, corruption and so forth. This thesis addresses those challenges by evaluating variety of diversity options, mobility performance aspects and robustness, battery saving opportunities, interference coordination and finally applicability of dual carrier. Studies are conducted with the help of time driven cellular system simulators where, e.g., fading, propagation and radio management functionalities are explicitly taken into account.

Keywords: Mobility, Handover, Receive Diversity, Transmit Diversity, Macro Diversity, Interference Coordination, Dual Carrier, Capacity, Battery Savings, Quality of Service, MBMS, VoIP, HSDPA, HSUPA.

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I would also like to rise up and thank non-profitable foundations who have provided extremely valuable funding to complete this thesis. In general, that kind of funding is imperative to support academic research and to keep Finnish know-how one of the most valued in the world. Foundations that have supported the competition of this thesis are listed in the following by their Finnish names in alphabetical order:

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ACRONYMS

1G	First Generation
1Rx	One Receive Antenna
2G	Second Generation
2Rx	Two Receive Antennas
3G	Third Generation
3GPP	Third Generation Partnership Project
ACM	Association for Computing Machinery
AMR-WB	Adaptive Multi-Rate - Wideband
AS	Active Set
AVI	Actual Value Interface
BCH	Broadcast Channel
BER	Bit Error Rate
BLER	Block Error Rate
BM-SC	Broadcast-Multicast Service Centre
BS	Base Station
BSC	Base Station Controller
C1	Carrier 1
C2	Carrier 2
C/I	Carrier to Interference Ratio
CAPEX	Capital Expenditure
CBR	Constant Bit Rate
CBS	Cell Broadcast Service
CDMA	Code Division Multiple Access
CPICH	Common Pilot Channel
CSCF	Call Session Control Function
CL	Closed Loop
CN	Core Network
Codec	Coder Decoder
CPC	Continuous Packet Connectivity
CPCH	Uplink Common Packet Channel
CS	Circuit Switched
CS	Combining Set

CRC	Cyclic Redundancy Check
DC	Dual Carrier / Dual Cell
DCCH	Dedicated Control Channel
DCH	Dedicated Channel
DSCH	Dedicated Shared Channel
DSSS	Direct Sequence Spread Spectrum
DTCH	Dedicated Traffic Channel
DTX	Discontinuous Transmission
DL	Downlink
DPCCH	Dedicated Physical Control Channel
DPDCH	Dedicated Physical Data Channel
DRX	Discontinuous Reception
E_b/N_o	Energy per Bit to Noise Ratio
E_s/N_o	Energy per Symbol to Noise Ratio
E_c/I_{or}	Energy per Chip to Interference Ratio
E-AGCH	E-DCH Absolute Grant Channel
E-DCH	Enhanced Dedicated Channel
E-DPCCH	Enhanced Dedicated Physical Control Channel
E-DPDCH	Enhanced Dedicated Physical Data Channel
E-HICH	E-DCH HARQ Indicator Channel
E-RGCH	E-DCH Relative Grant Channel
E-TFCI	Enhanced Transport Format Combination Indicator
FACH	Forward Access Channel
FER	Frame Error Rate
FO	First Order
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSTN	General Switched Telephony Network
GSM	Global System for Mobile Communications
HARQ	Hybrid Automatic Repeat Request
HLR	Home Location Register
HO	Handover
HoL	Head-of-Line

HS-DSCH	High Speed Downlink Shared Channel
HS-SCCH	High Speed Common Control Channel
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
HSUPA	High Speed Uplink Packet Access
I-CSCF	Interrogating Call Session Control Function
IC	Interference Coordination
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
ILPC	Innerloop Power Control
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IP-MS	IP Multicast Service
IR	Initialization and Refresh
ISD	Inter-site Distance
ITU	International Telecommunication Union
L1	Layer 1
LA	Link Adaptation
LTE	Long Term Evolution
LMMSE	Linear Minimum Mean Squared Error
MAC	Medium Access Control
MBMS	Multimedia Broadcast Multicast Service
MCCH	MBMS Control Channel
MEGACO	Media Gateway Control Protocol
MGCF	Media Gateway Control Function
MGW	Media Gateway
MIMO	Multiple Inputs Multiple Outputs
MRC	Maximum Ratio Combining
MSCH	MBMS Scheduling Channel
MTCH	MBMS Traffic Channel
Node-B	Base Station
NMT	Nordic Mobile Telephone
NST	Non-Scheduled Transmissions

OL	Open Loop
OLPC	Outerloop Power Control
OPEX	Operating Expense
P-CSCF	Proxy Call Session Control Function
p-t-m	point-to-multipoint
p-t-p	point-to-point
PA	Power Amplifier
PC	Power Control
PCH	Paging Channel
PedA	Pedestrian A (channel profile)
PDCP	Packet Data Convergence Protocol
PDU	Protocol Data Unit
PF	Proportional Fair
PG	Processing Gain
PHY	Physical Layer
PS	Packet Switched
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RACH	Random Access Channel
Rel'5	Release 5
Rel'6	Release 6
Rel'7	Release 7
Rel'8	Release 8
Rel'9	Release 9
Rel'99	Release 99
RFC	Request for Comments
RLC	Radio Link Control
RNC	Radio Network Controller
ROCH	Robust Header Compression
RoT	Rise over Thermal (RoT)
RR	Round Robin
RRM	Radio Resource Management
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol

RTT	Round-trip time
Rx	Receive
S-CCPCH	Secondary Common Control Physical Channel
S-CSCF	Serving Call Session Control Function
SCS	Scheduling Candidate Set
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SID	Silence Indicator Frame
SIP	Session Initiation Protocol
SINR	Signal to Interference and Noise Ratio
SMS	Short Message Service
SO	Second Order
SRB	Signaling Radio Bearer
SSRC	Synchronization Source
STTD	Space Time Transmit Diversity
TTI	Transmission Time Interval
TU	Typical Urban
Tx	Transmit
UAC	User Agent Client
UAS	User Agent Server
UE	User Equipment
UL	Uplink
UTMS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
VehA	Vehicular A (channel profile)
VoIP	Voice over IP
WA	Wrap-Around
WCDMA	Wideband Code Division Multiple Access
WLAN	Wireless Local Area Network

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LIST OF INCLUDED ARTICLES

PI K. Aho, J. Kurjenniemi, V. Haikola, and T. Ristaniemi. System Level Performance of Multimedia Broadcast Multicast Services (MBMS) with Macro Diversity. *Proceedings of IEEE Wireless Communications and Networking Conference (WCNC), Hong Kong, China*, 2007.

PII K. Aho, J. Kurjenniemi, V. Haikola, and T. Ristaniemi. Performance enhancement of Multimedia Broadcast Multicast Services (MBMS) with Receive Diversity. *Proceedings of the Sixth International Conference on Networking (ICN), Martinique, France*, 2007.

PIII K. Aho, J. Kurjenniemi, V. Haikola, C. Callender, and T. Ristaniemi. Multimedia Broadcast Multicast Service performance and its enhancements in WCDMA networks. *Wireless Personal Communications, ISSN 0929-6212*, 2007.

PIV K. Aho, J. Äijänen, and T. Ristaniemi. Impact of mobility to the VoIP over HSUPA system level performance. *Proceedings of the 67th IEEE Vehicular Technology Conference (VTC), Singapore*, 2008.

PV P. Lunden, J. Äijänen, K. Aho, and T. Ristaniemi. Performance of VoIP over HSDPA in mobility scenarios. *Proceedings of the 67th IEEE Vehicular Technology Conference (VTC), Singapore*, 2008.

PVI T. Nihtilä, K. Aho, and I. Repo. On Serving Cell Change Reliability. *Proceedings of the Wireless Communications and Networking Conference (WCNC), Budabest*, *Hungary*, 2009.

PVII K. Aho, I. Repo, T. Nihtilä, and T. Ristaniemi. Analysis of VoIP over HSDPA Performance with Discontinuous Reception Cycles. *Proceedings of International Conference on Information Technology (ITNG), Las Vegas, USA*, 2009.

PVIII I. Repo, K. Aho, T. Nihtilä, and T. Ristaniemi. Analysis of RRM Limitations and Restricted Transmission Periods for VoIP over HSDPA. *Proceedings of Asian Internet Engineering Conference (AINTEC), Bangkok, Thailand*, 2009.

PIX I. Repo, K. Aho, S. Hakola, T. Chapman, and F. Laakso. Enhancing HSUPA System Level Performance with Dual Carrier Capability. *Proceedings of IEEE International Symposium on Wireless Pervasive Computing (ISWPC), Modena, Italy*, 2010.

PX J. Puttonen, I. Repo, K. Aho, T. Nihtilä, J. Kurjenniemi, T. Henttonen, M. Moisio, and K. Chang. Non-regular Network Performance Comparison between HSDPA and LTE. *Proceedings of IEEE International Symposium on Wireless Pervasive Computing (ISWPC), Modena, Italy*, 2010.

PXI K. Aho, I. Repo, J. Puttonen, T. Henttonen, M. Moisio, and K. Chang. Benchmarking of VoIP over HSDPA and LTE Performance with Realistic Network Data. *Proceedings of IEEE International Symposium on Wireless Pervasive Computing (ISWPC), Modena, Italy*, 2010.

PXII F. Laakso, K. Aho, T. Chapman, and T. Ristaniemi. Applicability of Interference Coordination in Highly Loaded HSUPA Network. *Proceedings of IEEE 71st Vehicular Technology Conference (VTC), Taipei, Taiwan*, 2010.

PXIII F. Laakso, K. Aho, I. Repo, and T. Chapman. Extended HSUPA Coverage and Enhanced Battery Saving Opportunities with Multiple TTI Lengths. *Proceedings of IEEE 21st International Symposium on Personal Indoor and Mobile Radio Communications, (PIMRC), Istanbul, Turkey*, 2010.

The author of this thesis was the first author of MBMS related articles **[PI]**, **[PII]** and **[PIII]**. For those articles the author took the main responsibility of the simulation work and participated in the implementation work by adding necessary performance indicator statistics to the research tool. In addition, the author was the main contributor for the analysis and article writing process.

For the dynamic system level VoIP over HSUPA article **[PIV]** the author was responsible of the implementation and modeling work of VoIP, QoS monitoring and handover delays. The author carried out also the majority of the simulations as well as took the main responsibility of the result analysis and writing. The article **[PXIII]** covering quasi-static evaluation of VoIP over HSUPA was prepared with the guidance, analysis and writing support from the author. In addition, the author was involved in the implementation work by covering VoIP modeling to the quasi-static research tool.

For the articles **[PV]** and **[PVI]** dealing with downlink counterpart of the dynamic VoIP and mobility performance evaluation the author participated in simulation scenario planning, analysis and writing of the articles. The article **[PVII]** further deepening the HSDPA performance in the respect of VoIP QoS and battery savings included significant contributions from the author on analysis, writing, simulation work and on the implementation work resulting into DRX and related functions for the research tool. For the final HSDPA VoIP mobility article **[PVIII]** the author was involved in guiding the simulation and implementation work, analyzing the results and writing of the article.

The articles **[PIX]** and **[PXII]** covering quasi-static system level evaluation of dual carrier HSUPA and interference coordination were completed under the guidance of the author. The author was also heavily involved in the analysis and writing work.

Finally, apart from analyzing and writing, the author of this thesis was involved in various ways for the articles **[PX]** and **[PXI]** comparing HSDPA and LTE performance. The author took the responsibility of planning the different types of scenarios for the articles, parameter harmonization, conducting the VoIP over HSDPA and LTE simulations and proving guidance for the work on the CBR traffic HSDPA simulations.

1 INTRODUCTION

Wireless telecommunications industry has undergone many even revolutionary steps within time span of just a few decades. From the perspective of commercial wireless cellular technologies, to which this thesis is related to, the beginning of this evolution dates back to early 1980's. During that time so called *First Generation (1G)* networks also known as *Nordic Mobile Telephone (NMT)* networks [41] offering voice services were taken into use. First generation networks, based on analogue technology, were replaced by digital *Global System for Mobile Communications (GSM)* [37] networks roughly a decade later. On top of providing voice services, GSM, i.e., *Second Generation (2G)* systems introduced a new low-cost alternative to communicate, *Short Message Service (SMS)* [37]. Along with SMS technology and capability to provide other small bit rate data services GSM reached huge level of success and is still dominating technique on the markets. However, growing interest towards more innovative services such as rich calls, mobile-TV and music streaming in the wireless domain acted among the pushing forces to improve the packet data capabilities beyond GSM and its packet data evolutions. This need was met for the first time in 2001 when the first commercial *Third Generation (3G)* networks utilizing *Wideband Code Division Multiple Access (WCDMA)* [45] technique were taken into use. With the highest commercially implemented data rate of 384 kbps [45] the deployment of large variety of new services requiring higher data rates was now possible. The scope of services was broadened even further when *High Speed Packet Access (HSPA)* evolutions [46] to 3G networks were introduced. HSPA, i.e., 3.5G networks can go even beyond 100 Mbps theoretical peak data rates in the downlink if the latest enhancements such as multi-carrier and multiple inputs multiple outputs are taken into account. These higher bit rates accompanied by the new range of services have already attracted over 600 million WCDMA-HSPA subscribes around the word. When taking account GSM the amount of subscribers reaches staggering amount of 4.5 billion [84].

Some of those new services consist of sending identical data to vast amount recipients, for instance mobile-TV can be considered as one of those services. Sending identical information from single source to vast amount of recipients can consume a lot of radio resources and thus can easily limit the number of users which can be served. One innovative solution to improve the performance in previously presented situation is *Multimedia Broadcast Multicast Service (MBMS)* [14]. In order to meet the requirements set by various factors such as network operators, service providers and subscribers MBMS transmissions aim to deliver services with as high data rate that is possible by using minimum amount of radio resources.

Even though the increase in the number of subscribers is nowadays fueled by various data services, voice and especially *Circuit Switched (CS)* voice still remains as the main source of revenue for the cellular operators. From operator perspective having both CS network elements for voice and *Packet Switched (PS)* network elements for data services can be rather expensive and thus the point of interest has moved toward All-IP networks. In other words, in the future also voice would be carried via PS elements which is proved by the fact that *Long Term Evolution (LTE)* systems, which are to follow 3G/3.5G networks, support only PS voice [20]. All-IP network would lead inevitability to cost savings as the CS related part of the core network would not be needed anymore thus eliminating/reducing related *Capital Expenditure (CAPEX)* and *Operating Expense (OPEX)*. Additionally, by having services such as *Voice over Internet Protocol (VoIP)*, i.e., voice carried over PS elements, simpler implementation of rich call services would be possible as then voice and data would be then carried via same network elements.

This thesis aims to address variety of performance related aspects in terms of 3G/3.5G networks. As MBMS and VoIP are undeniably key concepts in wireless cellular networks those services are used in most of the studies to benchmarking different features. In the following sections, the research problem, related studies and motivation for this thesis will elaborated further.

1.1 Research Problem

When considering satisfactory user experience in the cellular network, various factors ranging all the way from optimizing the *Radio Resource Management (RRM)* functions to battery consumption need to be addressed. Addressing these different factors in detail will also result into to preventing performance losses from the network point of view. From the RRM functions, mobility and *Handovers (HO)* are very important and thus covered in detail in this thesis, especially in the respect of HSPA downlink where the lack of soft handover sets some challenges to the performance. Besides mobility performance, this thesis covers issues such as HSPA uplink interference coordination aimed to aid provide enhancements for power restricted cell edge terminals, dual carrier HSPA uplink used, e.g., to improve the peak rates in the uplink and *Continuous Packet Connectivity (CPC)* improving the downlink and uplink power saving opportunities. With the respect of the studies presented in this thesis, MBMS and VoIP are used in general as use cases due to their intolerance to excessive packet losses and delays. Moreover, as mentioned earlier MBMS and VoIP type of technologies are among the key services in cellular networks and thus addressing the performance of them is crucial.

Originally, MBMS was designed to use only *Downlink (DL)* (from the *Base Station (BS) User Equipment (UE)*) transmissions to the without any *Uplink (UL)* (from UE to BS transmissions) in order to save radio resources. Later there has been discussion to use MBMS principles with HSPA evolutions with some UL control information but those are out of scope in terms of this thesis. MBMS over WCDMA performance is analyzed with and without different performance enhancements considered for MBMS. Thus, apart from optimizing the performance in the respect of handover parameters, this study investigates whether MBMS performance can potentially be enhanced with, e.g., macro diversity combining, *Receive (Rx)* and *Transmit (Tx)* diversity.

Macro diversity combining means, roughly speaking, that transmissions coming from different cells/sectors are combined. MBMS is likely to be available through large parts of the operator network due to its targeted usage which includes services such as mobile-TV. From operator perspective it is not very feasible that there would be service interruptions, thus high availability. Hence, macro diversity is one of the most likely options for performance enhancement.

Receive diversity, on another hand, is one of the most efficient diversity techniques since, in addition to the diversity gain, the received combined signal power is theoretically doubled with two receive antennas when compared to single antenna reception [69]. However, the size, cost and power resources in the UE can limit the applicability of Rx-diversity and thus transmit diversity, i.e., deploying additional Tx-antennas to the base station can be considered as a more attractive solution.

From the perspective of VoIP, the *Third Generation Partnership Project (3GPP) Release 99 (Rel'99)* introduced WCDMA which enabled for the first time to run VoIP over cellular networks with reasonable quality, though with lower spectral efficiency than CS voice. The situation changed when HSPA evolutions to the 3G networks were introduced in *Release 5 (Rel'5)* and *Release 6 (Rel'6)* [46]. Both high speed evolutions, namely *High Speed Uplink Packet Access (HSUPA)* and *High Speed Downlink Packet Access (HSDPA)*, were conceived in order to achieve higher capacity and coverage for high transmission rates by the means of techniques such as *Hybrid Automatic Repeat Request (HARQ)*, fast scheduling and shortened *Transmission Time Interval (TTI)* [46]. However, studies regarding the HSPA evolutions soon proved that also delay critical small bit rate services, such as VoIP, can benefit from the new features that HSPA evolutions introduce. This thesis aims deepen the knowledge of VoIP performance over HSDPA and HSUPA networks by testing various RRM settings and techniques applicable for VoIP as well as to other services. These issues are, for instance, handover settings and performance, continuous packet connectivity and HSUPA TTI length selection.

CPC introduces possibility in HSPA networks to eliminate the continuous transmission and reception. From the uplink perspective this means that control channel used, e.g., for power control is not transmitted continuously, thus reducing interference levels in the system. In the downlink CPC introduces cycles that UE and network can agree upon so that UE will listen incoming transmissions only part of the time. Thus, CPC introduces situations where the UE can turn of the receive/transmit circuitry for power saving purposes. However, miss-configured CPC can potentially lead to severe performance losses due to the absence of power control and increased delays.

Taking previously mentioned aspects into account this thesis aims to answer to the following questions:

- Is macro diversity combining able to benefit the MBMS performance?
- How do the combining threshold settings impact the performance of macro diversity combining?
- What kind of performance enhancement can be achieved with receive and transmit diversity?
- Are there power savings opportunities available when network contains terminals with and without the capability to receive diversity?
- How reliable serving HSDPA cell change procedure is and is there possible enhancements to it?
- Do additional delays in HSPA handovers hinder the performance?
- What kind of impact user velocity and active set size poses?
- Is continuous packet connectivity able to provide battery saving opportunities without compromising the capacity and quality of service?
- Does interference coordination provide added value in the HSPA uplink?
- Are terminals able to utilize dual carrier HSUPA regardless of the limited available power?
- Does dual carrier bring benefit over two single carrier systems?
- How does HSDPA perform against next generation LTE networks?

To answer above mentioned questions, fully dynamic and quasi-static system level simulations are carried over. In general, there are at least two factors advocating the usage of system simulations: on the one hand, the complexity of the network behavior makes it virtually impossible to calculate accurate analytical results, and on another hand, physical network trials would be time consuming and expensive. Thus, system level simulations where RRM functionalities and their inter-actions are taken explicitly into account are essential to the performance evaluations. The used tools and their types are presented in detail in Chapter 6.

1.2 Related Studies

The purpose of this section is to point out some of the most important studies related to this thesis to provide further understanding of the different topics. Furthermore, this section elaborates in addition to Section 1.1 how this thesis deepens previous studies.

1.2.1 Multimedia Broadcast Multicast Service

Extensive capacity analysis for MBMS in 3G networks was carried out in [42], where the existence of multiple radio links was considered and evaluated under the assumption of soft combining principle. The study showed that without combining roughly 13 % of the Node-B power was required for good-enough coverage but with soft combining the same number was only 6 %.

MBMS was additionally studied in [34], where multi-resolution broadcast systems such as *Multiple Inputs Multiple Outputs (MIMO)* were studied in the respect of MBMS. Authors showed, for instance, that MIMO with two transmitting and two receiving antennas can double the spectral efficiency in comparison with single resolutions systems with corresponding *Block Error Rate (BLER)* target.

In [65] the trade-offs between resources used for physical layer and resources used for application layer with MBMS were considered. The results indicated that the balancing of the overheads is necessary in contrast to just focusing on the physical layer aspects.

MBMS counting mechanisms are addressed for instance in [25] and later extended to cover also macro diversity concepts in [77]. Counting mechanisms are used to determine the trade-off point when the transmission mode should be changed from point-to-point (dedicated resources) to point-to-multipoint (shared resources) mode. Those studies revealed that dedicated and shared channels could efficiently be used to deliver the MBMS services, selection of which depending on the number of users that desire the service and their requirements.

In addition, studies like [32] address MBMS performance analysis on such issues as longer *Transmission Time Interval (TTI)*, *Space Time Transmit Diversity (STTD)* and power and channel allocation with MBMS whereas in the same time [32] indicates that more work on the mobility aspect should be done.

Majority of the existing research is aimed, for instance, to reduce the required Node-B's transmission power for MBMS. This thesis deepens the MBMS performance analysis in many respects and thus gives power saving opportunities for Node-B as well as for the UE. First of all, different combining principles at the UE side are considered and carefully analyzed with respect of key parameters which have to be set in practice. Those parameters include, e.g., threshold values for adding, dropping and replacing a radio link from the set of combined radio links as well as timers related to those thresholds. In addition, the effects of typical non-idealities like the network non-synchronism and decision delays are considered. Finally, both single and dual receive antenna configurations are considered as well as adaptive branch activation/deactivation in the dual antenna case for the sake of UE battery power savings.

1.2.2 Voice over IP

One of the most critical parts of the downlink traffic, when VoIP or real-time traffic is considered, is how the scheduling works. Tight *Quality of Service (QoS)* requirements combined with small packets size with VoIP necessitate to fine tune scheduling from traditional *Round Robin (RR)* and *Proportional Fair (PF)* schedulers. Scheduling related issues with VoIP are studied closely in, e.g., [82], [66] and [30]. In [82] scheduling algorithm for VoIP traffic is presented which is extended in [66] with dynamic transmission (code and power) resource allocation method. According to the results presented in [66] with both of those scheduling related enhancements VoIP performance can be improved in terms of capacity even 115 % when compared to traditional RR scheduler. Finally [30] deepens the scheduling algorithm studies from the perspective of a situation where different penetration of other services in addition VoIP compete on the same radio resources. The conclusion of that paper is in line with [82] and [66] by indicating that QoS aware algorithm which takes, for instance, delay into account provides highest performance, even though mixture of different traffics would exist in the network.

Apart from scheduling related issues, impact of mobility and thus serving *High Speed Downlink Shared Channel (HS-DSCH)* change procedure is very crucial, especially for services, such as VoIP, that are highly sensitive to excess delay and packet losses. Mobility aspect in terms of HSDPA are addressed, e.g., in [81]. According to that study, even though cell change would have a long execution time, low speed users will not suffer (or suffer very slightly) from the lost of packets due to handovers. High speed users were, on the other hand, noticed to have interruptions up to 1 second without proper parameter tuning and thus impacting the overall VoIP performance.

Power consumption and possible power savings are almost as important as the radio network performance when small hand-held devices are in question. In the downlink discontinuous reception is considered as one technique to address that. Discontinuous reception has been previously studied in [86], where the effects of DRX cycles and related timers to the queue lengths, packet waiting times, and the power saving factor were studied in WCDMA. The study quantitatively showed how to select appropriate DRX cycle values and the related inactivity timer for various traffic patterns.

As coverage is typically limited by the uplink, studies regarding VoIP over HSUPA are at least equally important to evaluate the overall performance of VoIP. The capacity in HSUPA networks depends on number of parameters. These parameters include such settings as TTI length, number of retransmissions and relative channel power allocation.

VoIP over HSUPA system level performance with Rel'6 features is evaluated in [31]. According to the simulation results presented by that paper, VoIP over

HSUPA can offer an attractive solution compared to normal CS voice. However, like with HSDPA, careful parameter optimization is needed as for instance by setting parameter related to combining, i.e., active set update procedure can affect the performance nearly 10 % according to that paper.

HSUPA introduces two new code-multiplexed physical uplink channels: *Enhanced Dedicated Physical Data Channel (E-DPDCH)* for user data and *Enhanced Dedicated Physical Control Channel (E-DPCCH)* for control data. Introduction of these channels increases the peak to average power ratio and thus power amplifier back-off must be increased to fulfill the requirements of adjacent channel leakage ratio. Increasing the back-off results into reduced efficiency and maximum transmission power. This has impact on the VoIP capacity and it is addressed in [73]. Central finding of that paper is that if more power is allocated to the data channel, VoIP coverage can be enhanced but total power of UE is very limited and if less power is reserved for control channel it, for instance, easily leads to increased error rate.

Power savings at the UE side in the uplink can be achieved as a by-product when applying Rel'7 discontinuous uplink transmissions, also referred as uplink gating. The idea of uplink gating is to halt the control channels when the UE is in the idle mode, thus decreasing the amount interference. The performance together with gating is addressed, e.g., in [27], [39] and [40]. The first paper addresses the performance from the point of view of link performance and concludes that the stability of power control can be maintained and additional advantage in terms of performance can be achieved. The latter two papers deepen the analysis in terms of system simulations. In [39] it is shown that substantial gains can be seen in terms of VoIP performance. The highest absolute performance can be achieved with 2 ms TTI and gating according to that paper. In [40] static and dynamically activated MAC layer bundling of two VoIP packets in a single frame is proposed and compared against the single packet per frame transmission. Studies are conducted with 10 ms TTI and the results indicate noticeable gains of bundling when gating is used.

Finally, overview of the VoIP performance over both HSDPA and HSUPA networks is addressed in [44] with the respect of packet optimization features introduced in Rel'5, '6 and '7. The paper indicates that when all enhancements are included, the VoIP spectral efficiency is expected to go beyond twice the circuit switched voice capacity.

This thesis deepens the knowledge of VoIP over HSPA performance in various areas. For instance, existing studies in the respect of power saving opportunities concentrate only on the system performance point of view whereas this thesis addresses the performance also from the point of view of the user device. Moreover, mobility of users and interactions of the radio resource management functionalities are explicitly taken into account in this thesis providing irreplaceable analysis on the mobility aspects. By doing this, the thesis is able to point out some aspects that need to be paid special attention without having costly and time consuming physical network trials. This thesis will increase the knowledge, e.g., on the impact of hard and soft handovers, handover/active set update delays, power saving opportunities and velocity. Thus, this thesis provides further knowledge for the conditions where users are able better maintaining good quality of service and network will have better radio resources utilization.

1.2.3 Interference Coordination

There are several approaches for interference cancellation and coordination in wireless networks. Previously interference cancellation has been considered for HSUPA as well (see, e.g., [74] but interference coordination mainly for HSDPA and LTE from the cellular network point of view.

For instance, in [64] a simple interference mitigation scheme to enhance cell edge user data rate through network-wide coordination is introduced for HSDPA. The results of that paper show noticeable gains in terms of the cell edge user throughput without compromising the total system capacity and spectral efficiency.

In LTE interference coordination is quite widely studied through different coordination mechanisms, see, e.g., [67], [29], [85] and [76]. In one of the approaches, since the frequency band is divided into resource blocks and the same blocks are used in each cell, some of the blocks are restricted to be scheduled to only UEs near to the cell center. The pattern of blocks in which the scheduling is restricted is arranged between the cells according to a certain frequency reuse pattern. Considering neighboring cells, this arrangement allows for cell edge UEs to be scheduled into different blocks, thus reducing the inter-cell interference and improving user (and cell) throughput and availability of higher data rates.

In this thesis existing studies are broadened through investigating possibilities to apply the LTE-type interference coordination principles to HSUPA, by the means of time domain coordination. For the coordination to operate, the base stations need to be synchronized so that coordinated scheduling of cell center users and cell edge users in the neighboring cells can be implemented properly. Furthermore, an advanced timing mechanism needs to be introduced in order to synchronize uplink receive timings at each Node-B. This could be achieved, for example, by alignment of the downlink timings

1.2.4 Dual Carrier HSUPA

In Releases 5 and 6 HSDPA and HSUPA were defined to utilize 5 MHz bandwidth. However, in Release 8 dual carrier functionality (5+5 MHz) was specified for HSDPA. Performance study of the dual carrier/cell operation for HSDPA was carried out in [26]. The results indicate twice the peak rate for a single user utilizing dual carrier when compared to a single carrier performance. In addition, different sources for achieved gain, such as multi-user diversity and frequency selectivity on top of additional bandwidth, were identified. In [62] the dual carrier HSDPA studies are broadened further to cover multi-carrier (1-4 carriers) evolution of HSDPA performance. The study shows that the multi-carrier HSDPA system configurations with N carriers can bring the N-fold gain in average user throughput when compared to the single carrier HSDPA system, providing that the system is not fully saturated.

This thesis deepens the existing studies by evaluating the motivation for having dual-carrier also in the uplink. Studies were used to support the 3GPP standardization work on improving the uplink data rates to meet end user expectations of future mobile broadband services.

1.3 Other Articles

In addition to the included articles, the author of this thesis has published number of conference and journal articles dealing with long term evolution cellular networks. Those articles are as follows:

- K. Aho, T. Henttonen, J. Puttonen, and T. Ristaniemi, 'Trade-off Between Increased Talk-time and LTE Performance', Proceedings of IARIA Ninth International Conference on Networks (ICN), Menuires, The Three Valleys, France, April, 2010. Best paper award.

- J. Puttonen, H-H. Puupponen, K. Aho, T. Henttonen, and M. Moisio, 'Impact of Control Channel Limitations on the LTE VoIP Capacity', Proceedings of IARIA Ninth International Conference on Networks (ICN), Menuires, The Three Valleys, France, April, 2010.

- K. Aho, T. Henttonen, J. Puttonen, and L. Dalsgaard, 'Channel Quality Indicator Preamble for Discontinuous Reception', Proceedings of IEEE 71st Vehicular Technology Conference (VTC), Taipei, Taiwan, May, 2010.

- K. Aho, T. Henttonen, J. Puttonen, L. Dalsgaard, and T. Ristaniemi, 'Trade-offs Between Increased Talk-time and LTE Voice over IP Performance', International Journal On Advances in Telecommunications, December 2010, accepted for publication.

- K. Aho, O. Alanen, and J. Kaikkonen, 'CQI Reporting Imperfections and their Consequences in LTE Networks', Proceedings of IARIA Tenth International Conference on Networks (ICN), St. Maarten, The Netherlands Antilles, January, 2011, accepted for publication.

  The rest of this thesis is organized so that in the following sections the principles of WCDMA and HSPA are presented. Those are followed by introduction to MBMS and VoIP services after which research tools and modeling aspects are covered briefly. Finally, the achieved results and conclusion is drawn at the end of the thesis.

2 WIDEBAND CODE DIVISION MULTIPLE ACCESS

WCDMA is the most commonly adopted radio interface in third generation *Universal Mobile Telecommunications System (UMTS)* networks. UMTS and WCDMA are widely described in [45] and in 3GPP specifications [1]. Uplink and downlink packet evolutions, HSUPA and HSDPA respectively, to WCDMA are introduced in detail in [45], [46] and [35]. The purpose of this section is to present briefly key issues related to WCDMA, HSDPA and HSUPA in order to give the reader adequate knowledge to read and review the studies included in this thesis.

2.1 Radio Access Network Architecture

New logical radio access network was needed mainly due to new radio access technology but nevertheless it contains comparable elements to second generation GMS/*General Packet Radio Service (GPRS)* networks. The *UMTS Terrestrial Radio Access Network (UTRAN)* architecture, illustrated in Figure 1, consists of such logical elements as *Radio Network Controller (RNC)* and *Node-B* [45].

UTRAN can contain multiple RNCs which are responsible of controlling the radio resources and Node-Bs in its domain. RNCs are connected to each other via Iur interface. When compared to second generation systems RNC corresponds roughly to the GSM *Base Station Controller (BSC)*. Node-B, owned by RNC, corresponds roughly to the *Base Station (BS)* in GSM systems. Node-B or BS is responsible of handling the data flow between Uu, i.e., between UE and Node-B and Iub, i.e., between Node-B and RNC interfaces. In this thesis Node-B and base station terminology will both be used to mean solely UMTS network element and not GSM network element unless it is separately specified.

Finally, UTRAN is connected to *Core Network (CN)* via Iu interface. CN, based on GSM/GPRS systems, is responsible of routing connections between external networks and UMTS.

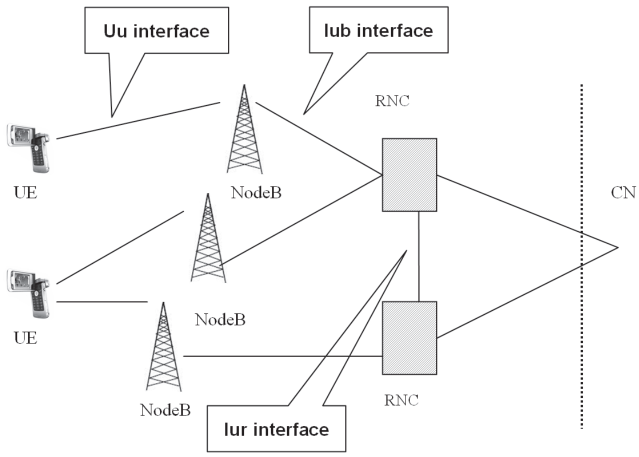

FIGURE 1 UMTS Architecture

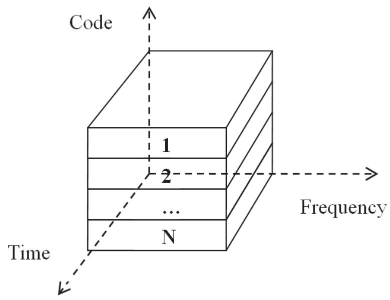

FIGURE 2 CDMA Principle

2.2 Introduction to WCDMA

WCDMA is based on *Code Division Multiple Access (CDMA)* technique where users share the same frequency and time plane. Instead of frequency or time, user entities are identified from each other by using codes, as illustrated in Figure 2. Usage of codes in WCDMA is two-fold: spreading, i.e., channelization codes are used to spread the information signal and scrambling codes are used on top of spreading codes to identify different sources from each other [45]. Main functionality of both channelization and scrambling codes are listed in Table 1. The principle of spreading operation is presented more closely in the following section.

TABLE 1 Functionality of Codes in WCDMA, [45]

	Channelization codes	Scrambling codes
Usage in UL	Separation of physical data and control channels from same terminal	Separation of terminals
Usage in DL	Separation of downlink connections to different users within one cell	Separation of sectors (cells)
Spreading	Yes, increases transmission bandwidth	No, does not affect transmission bandwidth

2.3 Spreading and Despreading

In WCDMA the information signal is spread over the whole frequency band which is 5 MHz both in uplink and downlink. Due to spreading transmissions are, for instance, robust against interference and jamming as well as they are secure. For the actual spreading operation WCDMA utilizes so called *Direct Sequence Spread Spectrum (DSSS)* technique where user bits are coded with unique binary sequence, i.e., with spreading/channelization code. The bits of code are called chips and length of codes determines the end bit rate. Shorter code equals to higher bit rate but at the same time it requires better *Signal to Interference and Noise Ratio (SINR)* to be detectable at the receiver.

The spreading operation is done so that user data is multiplied chip by chip with the spreading code. At the receiving end the signal is despread by multiplying unscrambled data with the same spreading code. The process is illustrated in Figure 3. The despread signal is then integrated, i.e., summed together over the length of spreading code. If the values of integrated signal vary around spreading factor and its opposite number the receiver knows that the signal is meant for it. Otherwise, if the integrated values are lingering around zero the transmission is from interfering signal. Moreover, the power of the own signal increases on average by the used spreading factor relative to the interference present in the CDMA systems. This effect is known as *Processing Gain (PG)* and is a fundamental aspect of all CDMA systems. PG gives CDMA systems the robustness against self-interference that is necessary in order to reuse the available 5 MHz carrier frequencies over geographically close distances [45].

2.4 Multipath Radio Channels

Transmitted radio signal can be reflected, duplicated, attenuated, etc. during its way to the receiver from the transmitter as demonstrated in Figure 4. As a result,

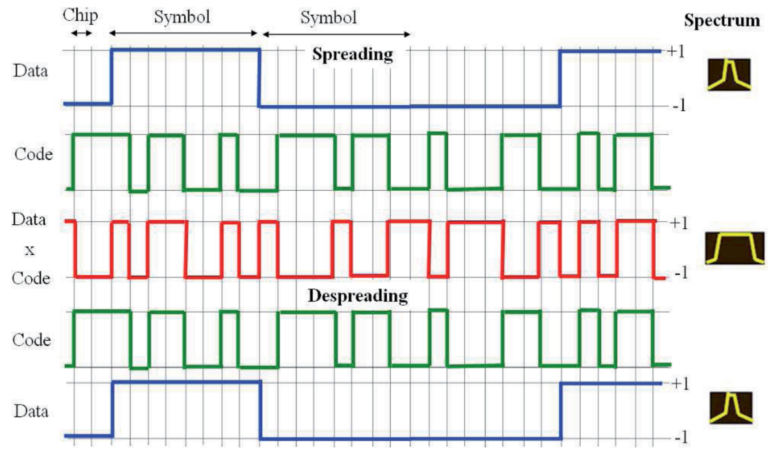

FIGURE 3 Example of Signal Spreading and Despreading, [45]

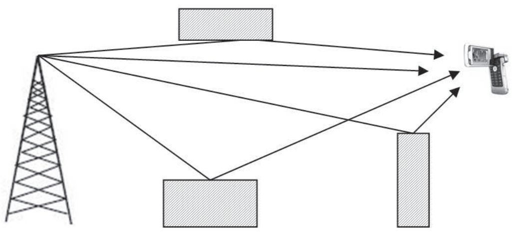

FIGURE 4 Example of Radio Signal Propagation

the receiver actually receives multiple copies of the original signal which differ from each other by, e.g., amplitudes and phases. These copies are referred as so called multipath components of the original signal. By using the information of these different multipath components the receiver can enhance the quality of received signal and mitigate, e.g., the effect of fast fading. Taking advantage from multipath components is referred as diversity gain. Other forms of diversity and how they can be taken into account are shortly covered in section 2.6 and covered with more details in [47]. However, first forms of signal attenuation, i.e., distance attenuation, slow fading and fast fading are discussed shortly in the following section. More detailed study regarding the subject can be seen in [28].

2.4.1 Attenuation

Attenuation can be categorized into three components: distance attenuation, shadowing, i.e., slow fading and fast fading. These components are gone through in the following subsections.

2.4.1.1 Distance Attenuation

Distance attenuation, i.e., pathloss means the loss of a signal's energy as it travels over the air from the transmitter to the receiver. The Okumura-Hata pathloss model is one of the most widely used for the calculations of coverage. Okumura-Hata model is based on the empirical measurements made by Y. Okumura in Tokyo [70], which were then fitted into mathematical model by M. Hata [43]. Initially Okumura-Hata model was applicable only for lower bandwidths but later it was extended with COST 231 model [36], which made it applicable for bandwidths covering also the area of $1500 \leq f(MHz) \leq 2000$. This combined model is called COST-Hata model and even though originally those applicable frequencies were restricted to below 2000 MHz, it is widely applied for UMTS frequencies exceeding that. For macro cellular simulation purposes 3GPP has adopted modified Okumura-Hata model which is described in detail in [21]. When considering a carrier frequency of 2000 MHz and a base station antenna height of 15 meters, the modified model can be described as:

$$P_L(d) = 128.1 + 37.6 \times \log_{10}(d), \tag{1}$$

where d is the distance expressed as kilometers. $P_L(d)$ shall in no circumstances be less than free space loss.

2.4.1.2 Slow Fading

Obstacles such as hills and buildings can come between the UE and Node-B and create so called shadowing to the received signal. Shadowing is caused by larger movements of a mobile within the propagation environment and usually it is changing relatively slowly, hence it is often referred as slow fading. The amount of slow fading and more generally signal attenuation depends greatly on the environment, for instance in the country side there can be much better propagation environment than in manhattan type of scenario where large number of buildings are obstructing the signal. This type of fading is generally modeled through a process with a log normal distribution and a correlation distance when simulation tools are considered [21] [75].

2.4.1.3 Fast Fading

As mentioned before, when signal is, for example, reflected from a surface of a building it can create a multipath component of the transmitted signal. When these multipath signals/components arrive at different times to the receiver they can cause either a constructive or destructive addition to the arriving plane waves as shown in Figure 5. This kind of phenomenon is referred as small scale or fast fading [75].

Multipath components can cause constructive addition to the arriving plane waves if their amplitudes and phases match and destructive addition if their phases do not match. Result of destructive addition is often signal with a very

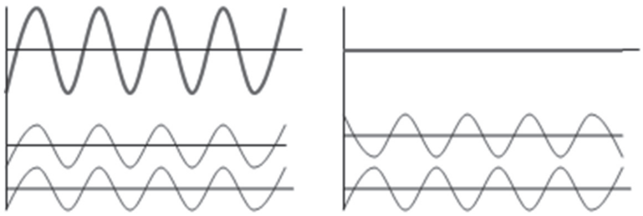

FIGURE 5 Example of Constructive (left hand side) and Destructive (right hand side) Addition

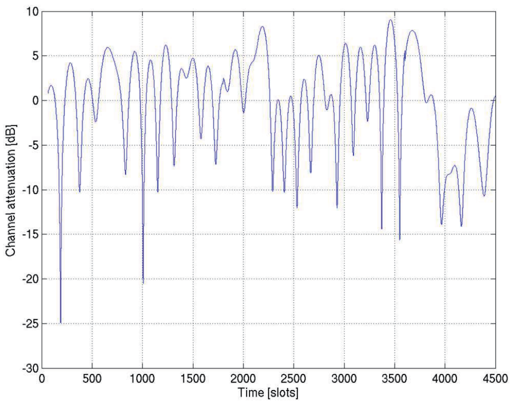

FIGURE 6 Example of Fast Fading Process

small or zero energy. The rapidity of amplitude variation of received signal is influenced by the speed of the receiver/transmitter and therefore variations might be very intensive as an example of fast fading in Figure 6.

2.5 Receiver

Originally and even to some extend nowadays in WCDMA systems the most commonly used receiver is so called Rake receiver. Though, as implied, more advanced receivers such as *Linear Minimum Mean Squared Error (LMMSE)* [69] are introduced. Rake receiver is specially designed to compensate the effects of fading ([71], [72], [78], [79], [45]). Compensating is done by using several 'sub-receivers' which are called fingers. Each of those fingers can receive individual multipath components. Every multipath component arriving at the receiver more than one chip time (0.26 μs) apart can be distinguished by the Rake receiver. Each component is then decoded independently and after that combined in order to make the most use of the different multipath components and thus reduce the effect of fading.

Combining method used in the Rake receiver is so called *Maximum Ratio Combining (MRC)*. The Principle of MRC is that the fingers of Rake receiver re-

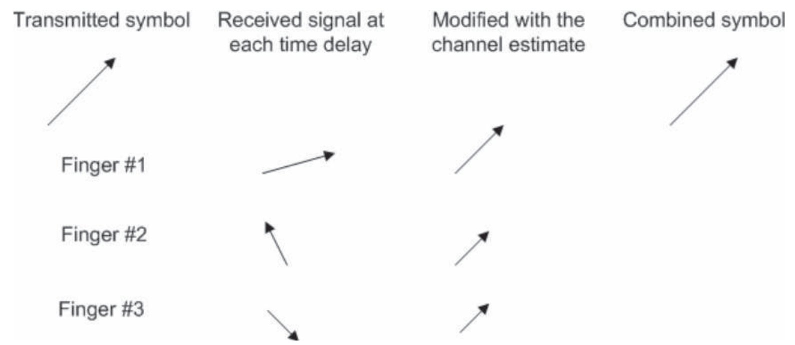

FIGURE 7 Principle of Maximum Ratio Combining Within the Rake Receiver [45]

ceive their individual multipath components which are then modified with channel estimate as illustrated in Figure 7. Channel estimate is acquired from known pilot measurements individually for each finger. Finally, those components are combined by simply summing them together.

2.6 Macro Diversity

Multipath components can be utilized also in such a way that the receiver collects, instead of only one source, signals from different cells/sectors and combines them in a predefined manner. Sectors from which the signals are used for combining, are chosen according to UE's measurements and the group of sectors used for combining in a certain moment of time is called a *Combining Set (CS)*. This kind of method is referred as macro diversity combining. Macro diversity can be considered as the most likely option for performance enhancement for services like MBMS as the current trends indicate that MBMS is going to be available through large parts of the network. In the following two different methods, introduced in 3GPP Release 6 [9], for macro diversity combining are briefly presented.

2.6.1 Selective Combining

Receiver with selective combining, Figure 8, receives and simultaneously decodes packets from different sectors which are in its CS. After the reception and decoding the received packet is handled at the *Radio Link Control (RLC)* layer where *Cyclic Redundancy Check (CRC)* is performed. Based on the CRC, the first packet which is received correctly is chosen. If any of the sources cannot provide error-free packet then the packet is lost.

2.6.2 Soft Combining

The other scheme used to mitigate the erroneous transmissions, illustrated in Figure 9, is called as soft combining. Unlike selective combining, soft combining is

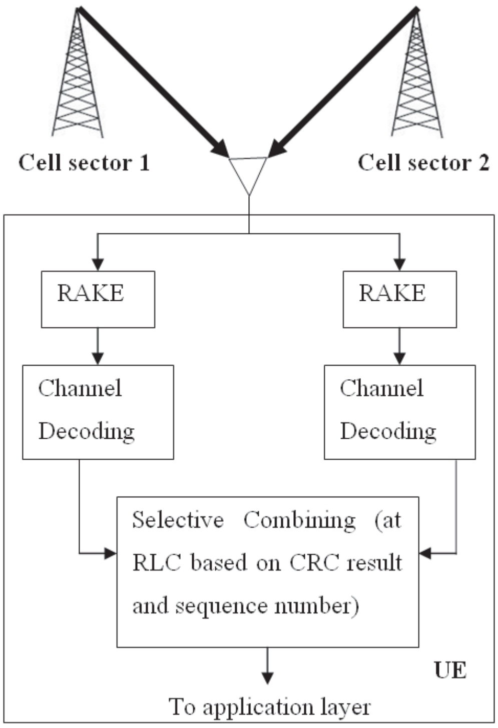

FIGURE 8 Selective Combining

performed on the physical layer. Receiver that uses soft combining collects all the packets from different sectors, which are included in the CS. The packets are aligned in time and finally soft values are combined. After the combining is done, the receiver decodes the (combined) packet and checks whether the packet was received correctly or not.

2.7 Receive Diversity

Receive diversity [48] means that the receiver has two or more receive antennas that are used to collect the multipath components. Even though there are a limited number of usable multipath components due to components starting to resemble each other, multiple receive antennas can receive larger number of usable components than receiver with one receive antenna. However, an adequate antenna spacing is required. If adequate antenna spacing between multiple antennas is achieved then the received multipath components are sufficiently independent of one another and hence the quality of the received signal can be enhanced with receive diversity.

In this thesis receive diversity is considered as one of the options to enhance MBMS performance since in addition to the diversity gain the received combined

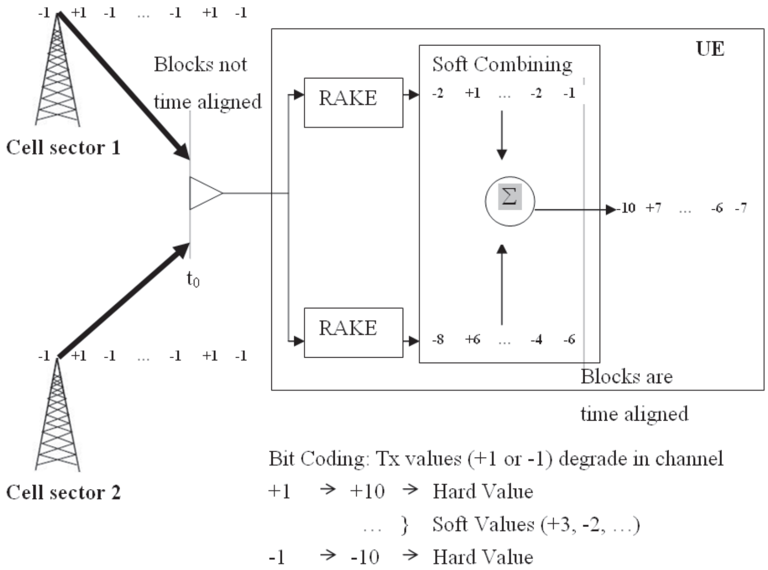

FIGURE 9 Soft Combining

signal power is theoretically doubled with two receive antennas when compared to single antenna reception. Apart from this study, impact of receive diversity is studied extensively, e.g., in [69]. In addition to 'traditional' receive diversity also a concept closely related to the receive diversity called Rx-switching, i.e., switching the second receive antenna off in good channel situations for power saving purposes is studied.

2.8 Power Control

Power control (PC) is one of the key features included in WCDMA [45]. The purpose of PC is to ensure, for instance, that each user receives and transmits just enough energy to prevent blocking of distant users, i.e., near-far-effect, exceeding reasonable interference levels and to mitigate effects of (fast) fading.

Near-far-effect is caused in situations where UEs are in different radio channel conditions caused by, e.g., interference, distance attenuation (pathloss) or shadowing. UEs in poor condition need higher transmission power to guarantee similar signal interference ratio as 'good UEs'. The problem arises when UE in poorer conditions, such as 'UE3' in Figure 10, increase their transmission power so that other UEs (in better conditions) experience higher interference and would eventually increase their transmission power as well. Thus, creating a loop that continues as long as UE(s) with lowest resources are being highly interfered and possibly even blocked by UEs with higher amount of resources at their disposal due to better channel conditions. Thus, advocating usage of PC in WCDMA.

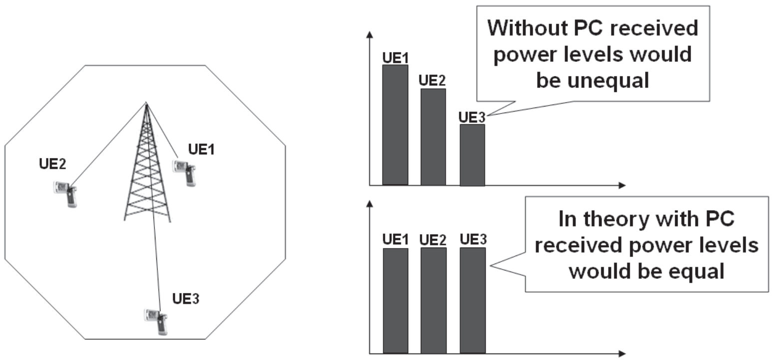

FIGURE 10 Power Control in WCDMA

Power control can be divided into *Open Loop (OL)* power control, which is also referred as slow power control and to *Closed Loop (CL)* PC, which is referred as fast power control. OL power control is used to compensate, e.g., free-space loss in the beginning of the call when the power levels are not yet controllable with pilot symbols. When connection has been established CL power control steps in. CL is applied 1500 times per second, hence the name fast power control.

Moreover, as described in [45] closed loop power control can also be divided into two parts: *Innerloop Power Control (ILPC)* and *Outerloop Power Control (OLPC)*. In ILPC the received signal levels are compared against the target value. If the value is higher than the target then the power is lowered with a certain step and other way around if the value is below the target. OLPC is then designed to adjust the target value for ILPC. OLPC can be used to control, e.g., the quality of service but generally the update resolution is lower than 1500 Hz which is used for ILPC.

2.9 Handovers in WCDMA

Generally speaking, handover is an action related to ongoing call where one radio channel to switched to another in order to guarantee the defined quality of service and continuity of an established call. Handover categories, types and produces are covered in detail in [45] and covered briefly in the following subsections.

2.9.1 Handover Categories

WCDMA supports three kinds of handovers. Those categories are:

- Intra-frequency handover,
- Inter-frequency handover and
- Inter-system handover.

Intra-frequency handover means a handover procedure within the same frequency and system. Inter-frequency HO is a handover between different frequencies but within the same system. And finally, inter-system HO is handover to another system, e.g., from WCDMA to GSM.

2.9.2 Handover Types

Different handover categories support only certain type of handovers. Handover types can be divided into soft, softer and hard handover. Soft handover is a handover between different base stations, where UE is connected simultaneously to multiple base stations and thus the transition between them should be seamless. Softer handover is a handover within the coverage area of one base station but between different sectors. Actual functionality of softer handover is similar to soft handover and thus the transition should also be seamless. Finally, hard handover is such that the source is released first and then new one is added, thus short interruption time in the service is expected.

From the different types mentioned before intra-frequency handover is in a way the most versatile as it supports soft, softer and hard handover. In contrast, inter-frequency and inter-system handovers support only hard handovers.

2.9.3 Soft and Softer Handover Procedure

As described before, soft(er) handover means maintaining connection to multiple base stations before the old connection is released. This is done to guarantee as seamless and lossless handover as possible. This procedure is characterized by number of parameters, namely:

- Add window and timer,
- drop window and timer, and
- replace window and timer.

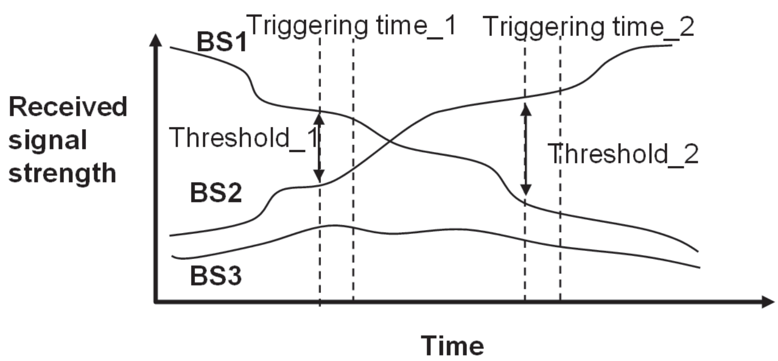

FIGURE 11 Soft and Softer Handover Procedure

Add window indicates a value of how much poorer a new signal can be compared to the best one in the current active set in order to be added into the set. Candidate needs meet the criteria at least for the duration of add timer. However, adding link to combining set can be done only if maximum number of links is not full yet (defined with parameter). Add procedure is illustrated in example of soft(er) HO procedure in Figure 11 as 'Threshold_1' and 'Triggering time_1'.

The opposite for adding is dropping the link from the active set. Link is to be dropped if the 'drop window' parameter, illustrated as 'Threshold_2' in the figure, is met. Similarly to adding also signal which is to be dropped needs to fulfill the drop condition after the corresponding drop timer is expired.

Finally, 'replace window' represents a value for how much better a new signal has to be compared to the poorest one in the current active set in order to replace its place. Replace event takes place only if active set is full as otherwise 'add' event would be applied.

2.10 Channels in WCDMA

In WCDMA there exists two types of transport channels: *Dedicated Channels (DCHs)* and common channels. The main difference between them is that the resources are shared between users when common channels are used whereas a DCH resource is reserved for a single user only that is continuous and independent from the DCHs of other UEs. The main transport channels used for packet data transmissions in WCDMA are called as DCH and *Forward Access Channel (FACH)* [45].

In WCDMA systems there exists only one dedicated channel which is known as DCH. DCH is mapped into physical data and control channels, *Dedicated Physical Control Channel (DPDCH)* and *Dedicated Physical Control Channel (DPCCH)*, respectively. DPDCH is used to carry user data and DPCCH all higher layer control information, such as handover and power control commands. Fast data rate change on a frame-by-frame basis is supported in the uplink. However, in the downlink data rate variation is taken care of either with a rate-matching operation or with *Discontinuous Transmission (DTX)* instead of varying spreading factor frame-by-frame basis. If downlink rate matching is used then data bits are either repeated to increase the rate or punctured to decrease the rate. With DTX the transmission is off during a certain part of the slot.

FACH is a shared downlink transport channel that can be used to carry packet data in addition to mandatory control information. FACH is used, for instance, to indicate that random access message has been received by BTS. As FACH carries vital control information, FACH has to have low enough bit rate so it can be received by all the UEs in the cell area. However, there can be more than one FACH in a cell enabling the possibility to have higher bit rates for the other FACHs. The FACH does not support fast power control or soft handover.

In addition to FACH there are five common channels in WCDMA: the *Broadcast Channel (BCH)*, *Paging Channel (PCH)*, *Random Access Channel (RACH)*, *Uplink*

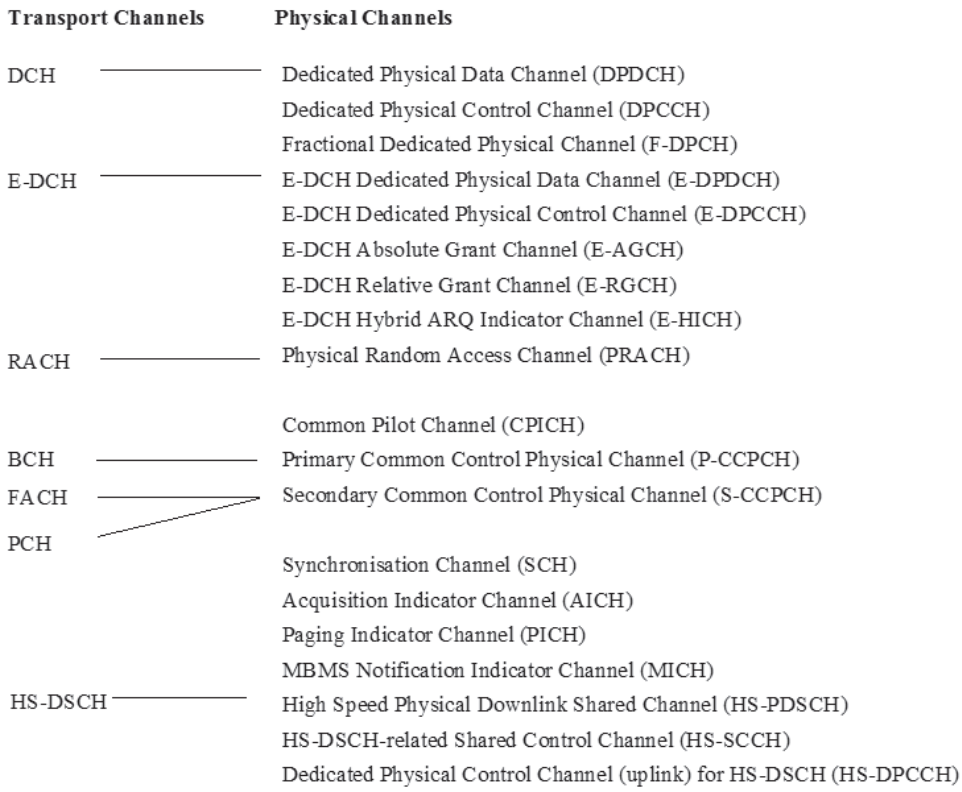

FIGURE 12 Mapping of WCDMA/HSPA Transport Channels to Physical Channels, [6]

Common Packet Channel (CPCH) and *Dedicated Shared Channel (DSCH)*. From common channels the most notable in terms of data transmissions is FACH as DSCH was optional feature that was seldom implemented by the operators and later replaced by HSDPA channel. Thus, 3GPP decided remove DSCH from Rel'5 specifications onwards. Also, CPCH has been taken out of the specifications from Rel'5 onwards as it was not implemented in any of the practical networks [45].

3 HIGH SPEED PACKET ACCESS

High Speed Packet Access (HSPA) introduces improved packet data capabilities for both uplink and downlink WCDMA. Downlink evolution is called as *High Speed Downlink Packet Access (HSDPA)* and it was introduced in Release 5. Release 6 introduced uplink counterpart of the evolution called as *High Speed Uplink Packet Access (HSUPA)*.

HSPA was originally conceived to achieve higher capacities and coverage for non-real time traffic with high transmission rate requirements. These requirements were met by the means of innovative techniques such as fast *Layer 1 (L1)* retransmissions referred as *Hybrid Automatic Repeat Request (HARQ)*, *Link Adaptation (LA)*, Node-B controlled scheduling and shortened *Transmission Time Interval (TTI)*. However, as the previous capacity and mobility studies regarding the HSPA evolution have proved (see, Section 1.2), also delay critical small bit rate services, such as VoIP, can benefit from HSPA evolution.

The following sections deal with features and properties of HSPA which are the most essential in terms of this thesis. More details of HSPA can be acquired from, e.g., [8], [4] [46] and [35].

3.1 High Speed Channels

As described above, the introduction of high speed channels was two done in two phases. HSDPA introduced in Rel'5 brought some changes to downlink packet data operations and HSUPA in Rel'6 brought changes for the uplink. These changes extend also to the channel structures and thus the purpose of this section is to give brief overview of the new channels.

In Rel'5 *High Speed DSCH (HS-DSCH)* for user data and *High Speed Common Control Channel (HS-SCCH)* for control information were introduced. HS-SCCH is sent two slots before HS-DSCH to inform the scheduled UE of the transport format of the incoming transmission on HS-DSCH. Timing offset is illustrated in Figure 13.

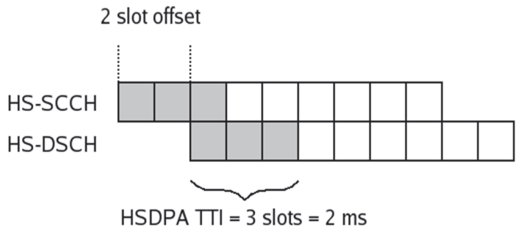

FIGURE 13 Offset with HSDPA Channels

Regardless of the fact that HSDPA introduced a new high speed channels the Rel'99 based DCH is the key part of the system. Rel'5 HSDPA is always operated with the DCH. DCH with HSDPA is used for following actions:

- If the service is only for packet data, then at least the *Signaling Radio Bearer (SRB)* is carried on the DCH. However, with Release 6 signaling can also be carried without the DCH.

- In case the service is circuit-switched then the service always runs on the DCH.

- In Release 5, uplink user data always go on the DCH.

In Release 6 new uplink transport channel, *Enhanced Dedicated Channel (E-DCH)*, was introduced. Unlike HS-DSCH with HSDPA, E-DCH is not a shared channel, but a dedicated channel. Similarly to DCH, E-DCH is also mapped to physical control and data channels. The user data is carried on the *Enhanced Dedicated Physical Data Channel (E-DPDCH)* while new control information is on the *Enhanced Dedicated Physical Control Channel (E-DPCCH)*.

Much like HSDPA, HSUPA is not a standalone feature but requires Rel'99 DCH. From the Rel'99 DCH, the dedicated physical control channel is unchanged and the need for the DPDCH depends on possible uplink services mapped to the DCH. *Dedicated Physical Control Channel (DPCCH)* is used, e.g., for fast PC.

In addition to transport channel new channels for scheduling control were also introduced with HSUPA: *E-DCH absolute grant channel (E-AGCH)*, *E-DCH relative grant channel (E-RGCH)* and *E-DCH HARQ indicator channel (E-HICH)*. E-AGCH is used to inform about absolute scheduling value, i.e., an absolute power level that the UE should adopt. E-AGCH can in principle allow transition between minimum and maximum data rates as well as any smaller data rate change in between the two extremes [45]. Absolute grant channel is sent only from serving cell. E-RGCH is a relative scheduling channel which transmits step up/down commands. Non-serving cell can transmit only hold/down commands in order to reduce and keep the interference levels on reasonable levels. New channel for retransmission control, E-HICH, carries the information in the downlink direction on whether a particular base station has received the uplink packet correctly or not. HSUPA channels are illustrated in Figure 14.

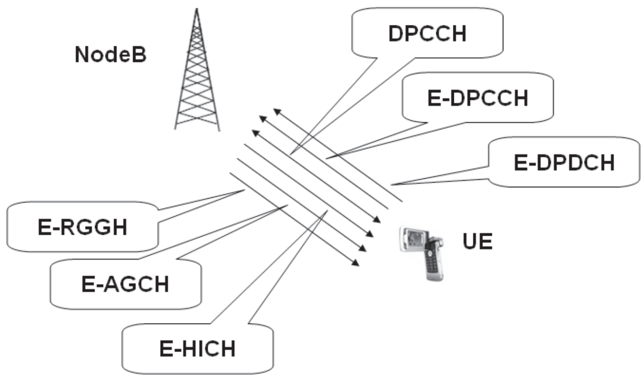

FIGURE 14 HSUPA Channels

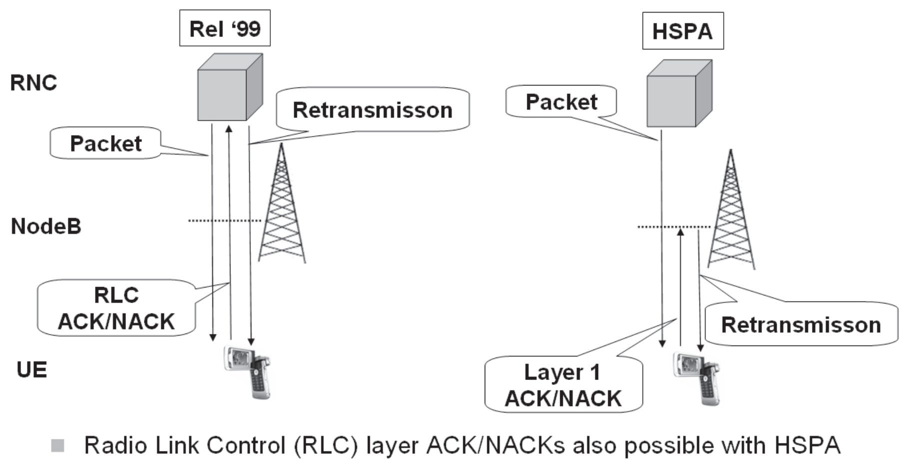

FIGURE 15 Rel'99 Retransmissions Versus Fast Retransmissions in HSPA

3.2 Fast Retransmissions

HSPA introduces changes to the basic WCDMA architecture in terms of transferring functionality from RNC to the Node-B. Fast retransmissions are among those new features placed in the Node-B level. In Rel'99, retransmissions can be employed only from RNC but after the introduction of HSPA retransmissions can be initiated/requested also from Node-B, see Figure 15. Scheduling is another important aspect related to architectural and transmission related changes but it will be addressed later in this thesis.

The benefit from fast retransmissions comes from layer 1 signaling, depicted in Figure 16, which indicates the need of retransmission much faster leading to noticeably lower round trip times than with Rel'99 retransmission procedure. Layer 1 retransmissions (HARQ) are operated in practice so that at the receiving end the decoder does not get rid of the received symbols if the transmission

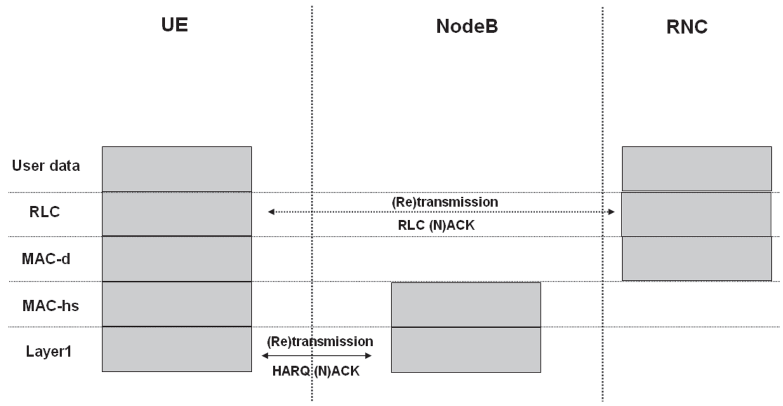

FIGURE 16 HSPA Retransmissions on Protocol Level

(i.e., CRC check) fails but combines them with retransmissions. Thus, achieving higher probability to get the transmissions through. There are two ways that retransmissions can operate: with identical retransmissions which are commonly referred as soft/chase combining or with non-identical retransmissions which is known also as incremental redundancy combining [46].

3.3 Scheduling

Scheduling, i.e., when the users are allowed to transmit or to be transmitted to, is definitely one of the most important aspects in terms of keeping, e.g., the delay in reasonable limits and handling the efficient utilization of radio resources. In the following brief description will be given on how scheduling is handled in the downlink (HSDPA) and uplink (HSUPA). More detailed description of scheduling, related algorithms and performance analysis can be found, e.g., from [46], [45] and [69].

In general the resources needed for each transmission are characterized by the transmission rate, the radio conditions, the intra-cell and inter-cell interference and the thermal noise. In HSPA HARQ retransmissions are required to be done with the same rate as the first transmission. This implies that the data rate should be chosen so that the required resources are sufficient to allow the possibility for retransmissions with higher resource requirements in case channel conditions have worsened.

3.3.1 HSDPA Scheduling

Scheduling in the downlink is controlled by the Node-B. Actual functionality of the scheduling is dependent of the algorithm in use but in general (in addition to available data to be scheduled) Node-B has two competing interest for allocating the resources: user fairness and radio resource utilization. In the following few different scheduling algorithms are presented that handle fairness and resource utilization in different manners.

3.3.1.1 Round Robin

Round Robin (RR) [46] is one of the most simplest scheduling algorithms and it is often used as a benchmarking reference for more advanced scheduling algorithms. With RR users are served in sequential order, i.e., all users are handled without priority. Positive sides with RR are that it is very easy to implement and each user gets served equally. On another hand, channel conditions are not taken into account and thus valuable radio resources might be wasted and thus network throughput might suffer.

3.3.1.2 Max C/I

When compared to RR *Max C/I* [46] is the other extreme of the scheduling alternatives if channel conditions and the scheduling fairness between users are taken as indicators. Whereas RR schedules users in sequential order Max C/I schedules users purely according their channel conditions. The user that has the best channel conditions in given time gets scheduled. This maximizes the throughput from the network perspective but the user fairness can suffer significantly.

3.3.1.3 Proportional Fair

Proportional Fair (PF) [46] is a compromise-based scheduling algorithm based upon maintaining a balance between two competing interests: maximizing network throughput by serving users in good channel conditions but at the same time allowing all users at least a minimal level of service. In practice this is done by assigning each users a scheduling priority that is inversely proportional to its anticipated resource consumption. High resource consumption leads to low priority and vice versa. However, priority is shifted with the factor which takes into account the occasions the user has been scheduled previously. In general the priority metric for users is calculated in a following manner [45], [46]:

$$p = d/r. \tag{2}$$

In Eq. 2 the instantaneous data rate, d, is obtained by consulting the link adaptation algorithm and average throughput of the user, r, is defined as follows:

$$r = \begin{cases} (1-a) \times r_{old} + a \times d, & \text{if the user is served} \\ (1-a) \times r_{old}, & \text{otherwise.} \end{cases} \tag{3}$$

Here a is so called forgetting factor. Hence, a^{-1} equals to the equivalent averaging period in a number of TTIs for the exponential smoothing filter.

3.3.1.4 VoIP Optimized PF

Normal proportional fair scheduler has some disadvantages when delay critical traffic is considered as it does not take delay of the queued packets into account as such. Improved scheduling and resource allocation algorithm for VoIP has been studied closely in [82] and [66]. Those papers present enhancement to pure PF algorithm by introducing so called *Scheduling Candidate Set (SCS)* from which the actual scheduling algorithm, for instance PF, selects users from. SCS is updated in each TTI so that it includes users who have certain number of VoIP packets buffered in the Node-B, whose *Head-of-Line (HoL)* packet delay is large enough or who have pending retransmissions in their HARQ manager.

3.3.2 HSUPA Scheduling

Scheduling for uplink is necessary to keep the interference levels reasonable [1] and thus improving the probability to get transmissions through. Similarly to HSDPA, also HSUPA scheduling has been moved from RNC to Node-B making the scheduling decisions happening with minimum latency. Regarding HSUPA two different scheduling schemes are defined: scheduled transmissions and *Non-Scheduled Transmissions (NST)* [46].

3.3.2.1 Scheduled Transmissions

Scheduled transmissions are, as the name implies, controlled by Node-B. Node-B and UEs will signal of the data rate requirements and how much resources UEs has available (referred also as power headroom) and then the Node-B assigns maximum transmit rate for each user so that the interference level experienced at the BS will not exceed certain reasonable level. However, the downside of scheduled transmissions with this approach is that when, for instance, VoIP is considered satisfactory minimum bit rate might not be achieved and in addition each request requires time and resource consuming signaling.

3.3.2.2 Non-scheduled Transmissions

NSTs are controlled by RNC which defines a minimum data rate at which the UE can transmit without any previous request. This reduces signaling overhead and consequently processing delays. To meet the tight delay requirements set for VoIP application, RNC controlled NTS of E-DCH is the most suitable choice for VoIP traffic as it enables the UE to transmit the VoIP packets as soon as they arrive to the physical layer.

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[1] measured generally through *Rise over Thermal (RoT)*, i.e., noise rise which indicates the ratio between the total power received from all of the UEs at the base station and the thermal noise [80]

3.4 Multiplexing

As referred earlier, HSPA was designed originally to provide higher data rates for the end-users. When introducing relatively low bit rate services, such as VoIP, capacity could be wasted by serving only one user at a time. To allow scheduling of multiple users per TTI mulplexing schemes for both HSDPA and HSUPA have been developed [46]. In the following those will be presented briefly.

3.4.1 HSDPA Multiplexing

In HSDPA multiplexing users to a single TTI transmission is handled through code multiplexing. For instance, assuming 15 available codes and three users to be scheduled, each could be served with five codes during the same TTI. However, each simultaneously scheduled HSDPA user requires HS-SCCH channel. Thus, total amount of required power for HS-SCCH increases when users are code multiplexed and moreover it also limits the number of code-multiplexed users.

3.4.2 HSUPA Multiplexing

Multiplexing of users in HSUPA systems can be accomplished through three different ways:

- *Code Division Multiplexing (CDM)*,
- *Time Division Multiplexing (TDM)*, or
- *CDM/TDM*.

With pure CDM all users are first provided with minimum bit rate which is followed by reserving resources for retransmissions. After that higher allocations can be given according to the scheduling priority metric (and request) and *Rise over Thermal (RoT)* level that the scheduler is aiming at as Figure 17 illustrates.

TDM allows only one user to be scheduled in each TTI resulting in increased queuing times but also reduced interference levels. Retransmissions are prioritized over new transmissions due to synchronous HSUPA HARQ operation. Figure 18 shows TDM procedure and from which components RoT is consisting of over the predicted interference.

Combination of the two above, CDM/TDM, does not guarantee minimum bit rate for all users but does not limit the maximum number of scheduled users to only one either. The amount of multiplexed users is up to the RoT level that the scheduler is aiming at. Each user, up until RoT target is reached, is allocated with bit rate according to scheduling priority metric and request as it is illustrated in Figure 19.

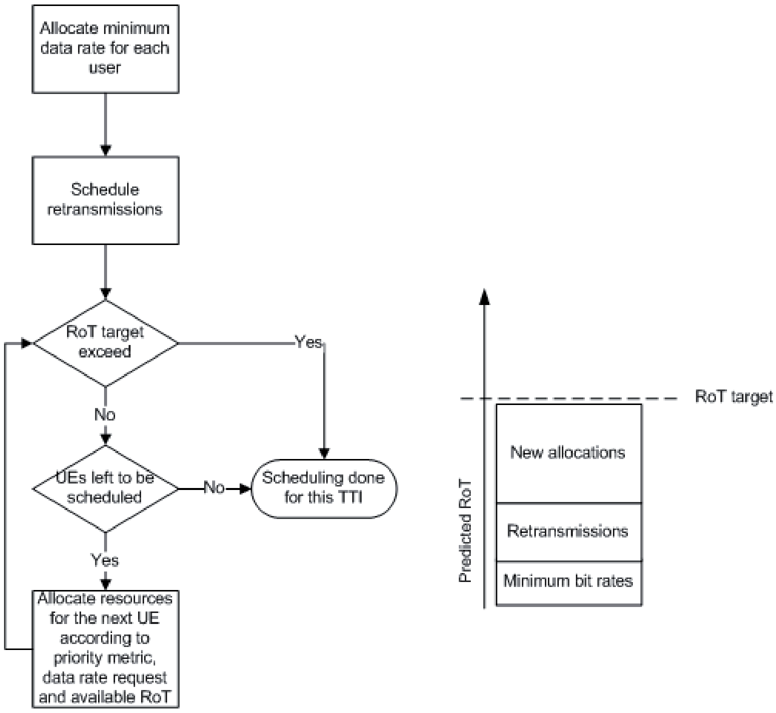

FIGURE 17 HSUPA Code Division Multiplexing Procedure and an Example of Predicted RoT (without interference) After Scheduling

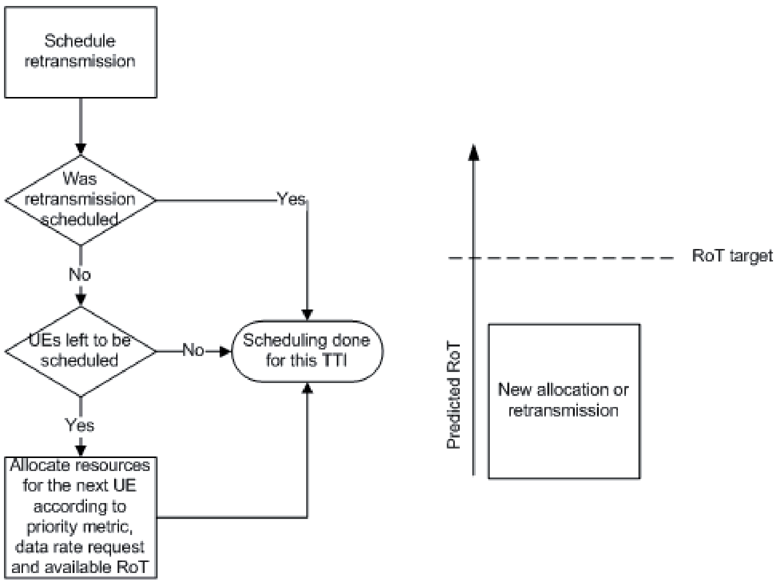

FIGURE 18 HSUPA Time Division Multiplexing Procedure and an Example of Predicted RoT (without interference) After Scheduling

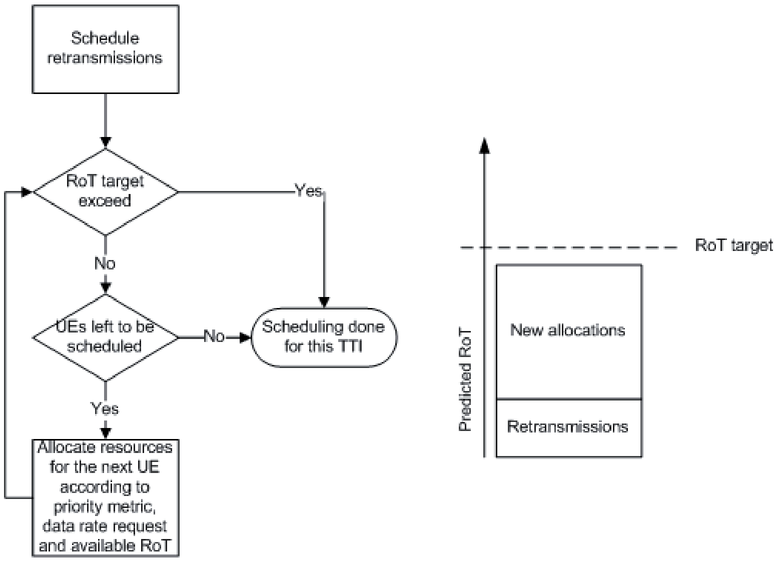

FIGURE 19 HSUPA CDM/TDM Procedure and an Example of Predicted RoT (without interference) After Scheduling

3.5 Mobility Management in HSPA

Handovers are one of the most crucial parts of the successful user experience as they can cause, for instance, additional delays or even disruption times to the transmissions. In terms of HSPA the handover procedure is somewhat altered when compared to Rel'99 WCDMA [46]. With HSDPA, the user can be connected to one serving HSDPA Node-B at the time leading to hard handover when the handover is required. On another hand, when HSUPA is considered the soft handover procedure is still supported similarly to Rel'99 WCDMA.

With HSDPA, even though, there is only one serving HS-DSCH cell, the associated DCH itself can be in soft handover and maintain the active set as in Rel'99 as Figure 20 illustrates. The handover procedure with HS-DSCH is initiated when a link in (DCH) active set becomes higher in strength and stays stronger for certain period of time, referred as time-to-trigger. If the condition is met then the measurement report is sent from UE to Node-B, which forwards it to the RNC. If the requirements for, e.g., admission control are met the RNC can then give the consent (via Node-B) for the UE to make the handover. In the case of intra-Node-B handover, the HARQ processes (transmissions) and Node-B buffers can be maintained and thus there is only minimal interruption in data flow. However, with inter Node-B handover, i.e., between Node-Bs, the Node-B packet buffers are flushed including all unfinished HARQ processes which are belonging to the UE that is handed off.

As mentioned before, with HSUPA the soft(er) handover procedure described in Section 2.9.3 is still supported. The HARQ operation in soft handover situation is done so that if any Node-B part of the active set sends an ACK, then the information given to the *Medium Access Control (MAC)* layer is that an ACK has been received and the MAC layer will consider the transmission successful [46]. At the UTRAN side this means that RNC, which is responsible for packet reordering, will receive at least one successful transmission. This procedure is also illustrated in Figure 21.

3.6 Continuous Packet Connectivity

Continuous packet connectivity is a 3GPP standardization item ([3], [17], [12]) covering challenges such as having high amount of simultaneous users in the system, users staying connected long periods of time, battery consumption and bursty type of traffic. CPC, introduced in *Release 7 (Rel'7)*, handles challenges through three main features:

- UL discontinuous transmission (UL DTX),
- DL discontinuous transmission (DL DRX), and
- Restricted HS-SCCH.

From the perspective of this thesis the most essential ones are UL DTX and DL DRX and thus in the following focus will on those. Generally speaking, DTX and DRX eliminate the need for continuous transmission and reception when data is not exchanged. Whereas in earlier releases control channel (DPCCH) was required to be transmitted continuously whether there is data or not. Similarly in DL HS-SCCH was required to be monitored continuously.

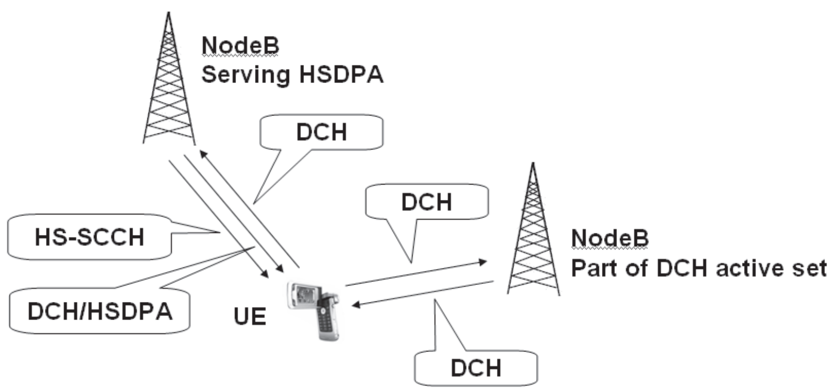

FIGURE 20 HSDPA Hard Handover

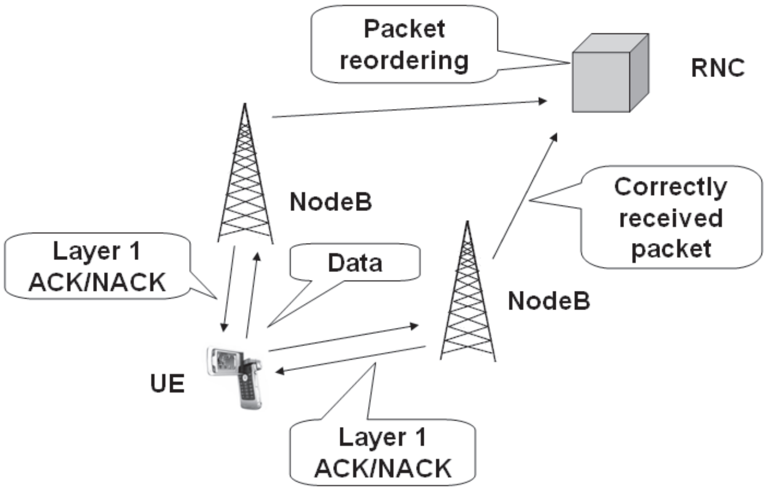

FIGURE 21 HSUPA Soft Handover

3.6.1 Discontinuous Transmission

As mentioned above, before 3GPP Rel'7, DPCCH was defined to be transmitted continuously regardless whether there is actual user data to be transmitted or not. Thus, highly loading the cell and draining the UE battery with continuous transmissions. An ideal solution for mitigating the impact of those would be to keep the UE silent during the periods that it is not transmitting any data and activate the control channels just for the transmissions periods as Figure 22 shows. However, that could compromise, for instance, the fast power control which would be then updated only during the times when the data is exchanged.

Release 7 addresses this problem with discontinuous DPCCH transmissions. With UL DTX the transmission of DPCCH is controlled by data transmissions and various different timers which are illustrated in Figure 23 and described shortly as follows:

- *Data transmission:* DPCCH is transmitted continuously throughout the data transmissions.

- *DPCCH preamble:* before the actual data transmission or DPCCH burst is initiated either short or long burst is transmitted to allow the fast power control to correct the UE's transmit power levels. The long DPCCH burst is sent if UE is in DTX cycle 2 and data transmission is initialized, otherwise short is sent.

- *DPCCH postamble:* after the data transmission or DPCCH burst has ended, DPCCH is still transmitted as long as the postamble length defines.

- *Inactivity timer:* transitions from DTX cycle 1 to 2 are controlled by inactivity timer. Once the timer expires, UE moves from cycle 1 to cycle 2. The timer is (re-)started after each data transmission.

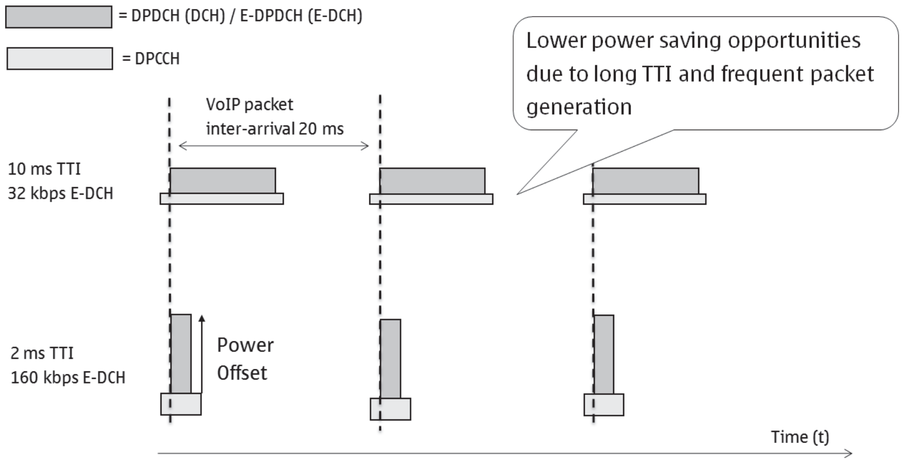

FIGURE 22 Example of VoIP UL Traffic with Different TTIs and ideal UL DTX

- *DTX cycle 1 and 2:* control the periodic DPCCH bursts that are sent regardless is there actual data transmission or not. UE uses cycle 1 when there has been data transmission during the past inactivity timer length.
- *DPCCH burst 1 and 2:* define the length of DPCCH bursts initialized by cycle 1 or cycle 2 timers, respectively.

3.6.2 Discontinuous Reception

In the course of providing satisfactory user experience the UE battery life can very well become limiting factor on top of the actual network performance. Thus, this is addressed through DRX which is aimed to provide opportunities for battery savings at the user terminals. In case of DRX, UE and UTRAN may limit the number of subframes where the UE needs to monitor the HS-SCCH so that:

- DL scheduling is still possible.
- UE is able to shut-off the receiver circuitry over some periods of time to yield a non 100 % receiver duty cycle.
- Minimum monitoring/measurement possibilities to keep up with changes in UE's active set due to mobility.

As a result UE monitors a limited subset of HS-SCCHs in the time domain, e.g., one subframe every two, or every four subframes. This monitoring interval is referred as *DRX cycle timer*. In addition to cycle timer DRX operation is controlled through *inactivity timer*. Inactivity timer is started if UE successfully decodes HS-SCCH indicating data transmission. Inactivity timer represents the number of consecutive subframes that UE will monitor for incoming transmissions before falling back into inactivity. If HS-SCCH is not received UE will fall back to inactivity and has an option to switch off its receiver circuitry for battery saving

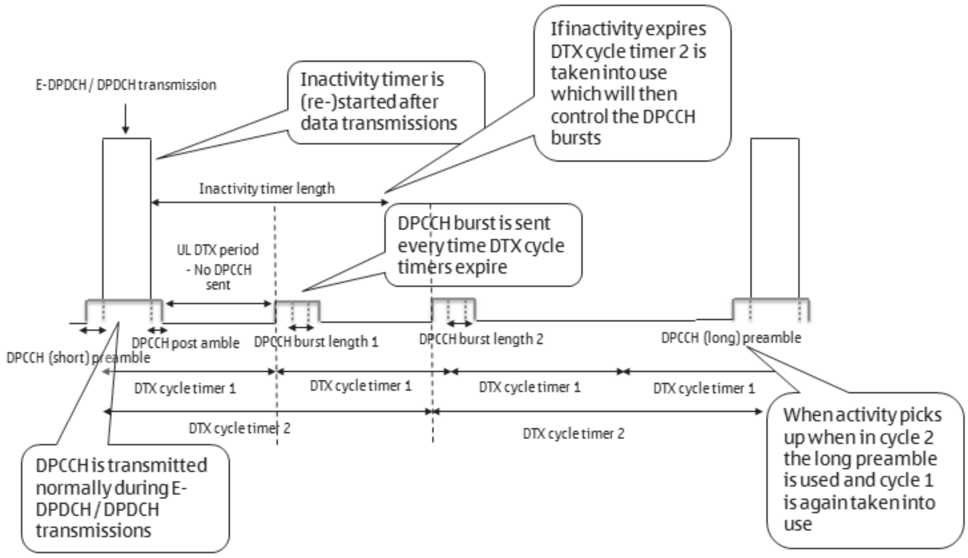

FIGURE 23 Uplink Discontinuous Transmission

purposes. Moreover, DRX cycles should be adjusted with different offsets for different terminals to guarantee fair and adequate scheduling opportunities in the time domain. Example of DRX operation is illustrated in Figure 24.

3.7 Interference Coordination

In general the total system coverage is limited by the uplink mainly due to two factors: limited power resources of the user terminals and interference levels. Interference in the uplink consists of three components: intra-cell (i.e., own cell), inter-cell interference (i.e., other cell) and thermal noise. Intra-cell interference arises because the signals between UEs are not completely orthogonal when arriving at the base station. If the intra-cell interference is successfully mitigated, then the next largest source of interference is inter-cell interference. Mitigation of inter-cell interference has been studied extensively as part of the development of the LTE standard, for an overview, see, e.g., [67], [29], [85], [76].

In this thesis applicability of time domain *Interference Coordination (IC)* for HSUPA system is studied. Time domain IC sets some restrictions and requirements for the system. Firstly, time domain IC for HSUPA requires that the base stations are synchronized so that it enables the possibility that in certain cells only the cell edge users transmit while in the neighboring cells only the center users are allowed to transmit. Secondly, criteria for defining cell edge / center users needs to be introduced. Thirdly, scheduling procedure needs to be modified accordingly.

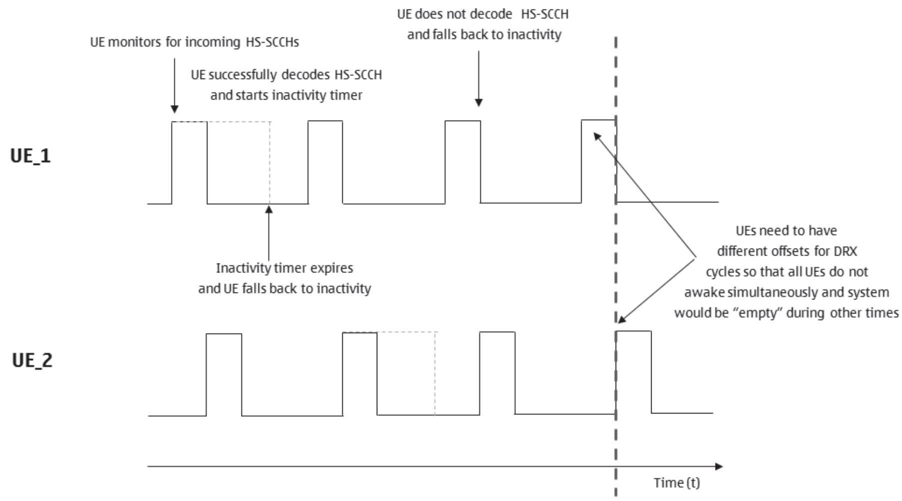

FIGURE 24 Example of DRX Operation

3.7.1 IC Procedure

The actual process of the time domain IC consist of restricting the power of *Enhanced Dedicated Physical Data Channel (E-DPDCH)* for cell edge users when center period is in turn. Cells in center period allow the normal scheduling/multiplexing operation only for center UEs. When cells are in edge period only cell edge or all (depending on the configuration) UEs may be scheduled without restricting E-DPDCH. In this thesis it is assumed that cell edge periods are restricted to 1/3 of the TTIs and cell center periods for the remaining 2/3 of the TTIs. IC operation with that kind of configuration is illustrated in Figure 25.

This kind of coordination could be achieved by activating/deactivating HARQ processes, or alternatively by the means of RNC restricting the maximum data rate applicable to UEs in certain HARQ processes in certain time periods. It should be noted that activating/deactivating HARQ processes is possible in the Release 6 specifications and thus applicable to legacy UEs, whereas specifying a different minimum data rate in some HARQ processes would require a specification change and thus would require new terminals.

3.7.2 UE Classification Criteria

The proportion of edge and center users is up to the configuration. In this thesis a user is classified as cell edge user if the E_c/N_o difference between the user's serving cell and any other cell is smaller than a certain value defined by a parameter. Thus, the exact proportion of edge and center users will vary between cells. The percentage for edge/center splits quoted in this thesis refer to the average number across all of the cells.

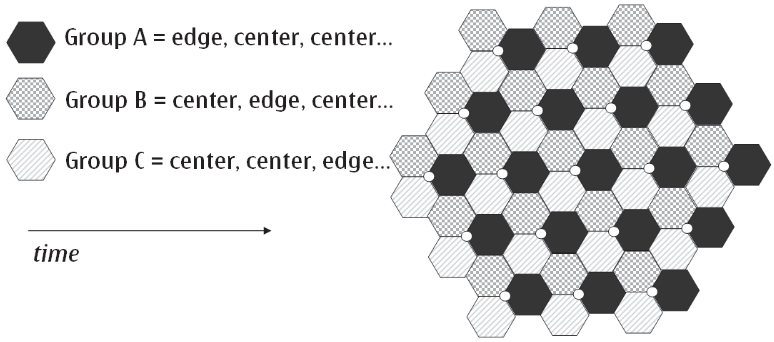

FIGURE 25 Illustration of Interference Coordination

3.7.3 IC Scheduling

HSUPA scheduling/multiplexing presented in Sections 3.3.2 and 3.4.2 requires some changes when interference coordination is applied. Recall that there are two periods of operation: cell center period and cell edge period.

With CDM and CDM/TDM scheduling the following actions are taken:

- During both periods all the users are prioritized according to the PF metric but the priority of users who are not in turn is modified so that they will have lower priority than the users whose turn it is.
- During cell center periods only cell center users are scheduled according to priority metric. Cell edge users will be allocated only with configured minimum transmission rate.
- During cell edge periods the cell edge users are scheduled first according to priority metric. But after that there is a possibility to schedule also cell center users if that mode is configured. Otherwise cell center users will be scheduled only with minimum transmission rate.

With TDM scheduling only the scheduling candidate set needs to be modified so that appropriate users are available. Recall, only one user is scheduled at each transmission time interval.

3.8 Dual Carrier

Before *Release 9 (Rel'9)*, HSUPA was defined to utilize single 5 MHz bandwidth. However, as operators may have multiple 5 MHz spectrum bands available and as *Dual Carrier (DC)* HSDPA was already specified in *Release 8 (Rel'8)*, logical continuum was to study the possibility of DC-HSUPA, see Figure 26. Study in 3GPP lead to standardizing DC-HSUPA for Rel'9, see, e.g., [4], [18], [16], [13], [19] and [24]. In this thesis the motivation towards DC-HSUPA is studied. Results

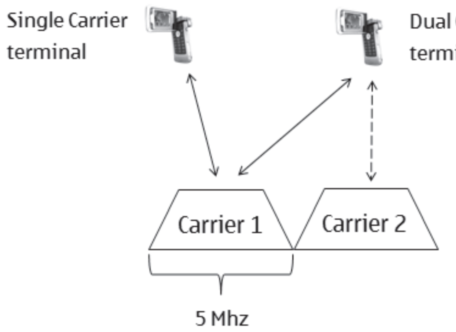

FIGURE 26 Example of Dual Carrier Operation

of this thesis have been also utilized to support the DC-HSUPA standardization work.

3.8.1 Motivation

Enabling dual carrier operation for HSUPA provides many benefits over single carrier for both the network and the UEs. Some of the most significant benefits and motivations behind DC are covered below:

- Some UEs may send with the highest *Enhanced Transport Format Combination Indicator (E-TFCI)* but still have spare power. (1)
- Some UEs may send with high code rate (>0.7) or with 16-QAM. (2)
- RoT limit - scheduler cannot allocate that much resources as UE would be able to use. (3)
- Enables load balancing between carriers and maximizing the use of resources. (4)

In the cases (1) and (3), with dual carrier UEs may use the spare power on the other carrier to increase their total transmission rate. In case (2) high code rate or 16-QAM transmission could be divided over two carriers to get higher link efficiency for the transmission. Or with adequate power resources UEs could double their peak data rate in theory. In the case (4) network and UEs can benefit from less users transmitting at the same time on E-DPDCH and thus generating less interference.

3.8.2 Requirements

DC-HSUPA operation has some restrictions / requirements that have been defined in 3GPP. In the following some of those are listed briefly:

- DC-HSUPA requires DC-HSDPA for operating.
- The carriers belong to the same Node-B and on adjacent carriers.

- UE has a single *Power Amplifier (PA)* and the total transmit power is shared between carriers.
- Control channels are transmitted on both channels.

In addition those requirements, naturally, additional spectrum and terminal support is also required to implement DC in practice.

3.8.3 Scheduling

Scheduling with dual carrier follows the same principles as Sections 3.3.2 and 3.4.2 present. However, due to availability of two carriers in order to maximize the resource utilization slight modifications to scheduling could be needed. In the following three potential methods to handle DC-HSUPA scheduling are presented.

3.8.3.1 DC Scheduling 1

Scheduling of UEs is always done according to their priority metric on *Carrier 1 (C1)*. In other words C1 is scheduled first and then *Carrier 2 (C2)* in the same order. This can leave C2 slightly inefficient as rate requests are done for C1 and when C2 is scheduled the best UEs and request are not necessarily scheduled first. However, this kind of scheduling should provide frequency diversity gain for the terminals transmitting on both carriers.

3.8.3.2 DC Scheduling 2

Second scheduling option is similar to option 1 when C1 is question. However, when C2 is in question, the UEs that were not scheduled on C1 are scheduled first on C2. This kind of scheduling resembles scheduling of two separate single carriers with the exception that UEs can switch more easily between carriers.

3.8.3.3 DC Scheduling 3

Third scheduling option presented here is such that rate requests (done by the UE) and scheduling (by Node-B) takes into account which carrier is better (i.e., has lower DPCCH level). UEs are sorted to a list according to their better carrier and UEs are scheduled in that order with each UE having its better carrier scheduled first. This kind of scheduling should improve the utilization of both carriers and provide frequency diversity gain as well.

4 MULTIMEDIA BROADCAST MULTICAST SERVICE

Multimedia Broadcast Multicast Service allows different forms of multimedia content to be delivered efficiently by using either broadcast or multicast mode. The term broadcast refers to the ability to deliver content to all users who have enabled a specific broadcast service and find them in a broadcast area. Multicast, on the other hand, refers to services that are delivered solely to users who have joined a particular multicast group. Multicast group can be, for example, a number of users that are interested in a certain kind of content, such as sports. In contrast to the existing solutions for one-to-many data delivery in mobile networks MBMS makes more efficient use of network resources and capacity. In the following sections MBMS is presented from the technical specifications point of view in order to provide the reader with basic understanding of MBMS.

4.1 Background

Before MBMS broadcast and multicast transmissions in 3G networks were dealt with by using either the *Cell Broadcast Service (CBS)* which is defined in [2] and [23] or the *IP Multicast Service (IP-MS)* which is described in [5] and [7]. However, those methods contain some problems. With CBS it is not possible to send anything other than message-based services with low bit rates. Situation with IP-MS is not that much better as there is no capability to use shared radio or core network resources for traffic intended for multiple recipients. Without shared resources, the overall use of radio resources easily exceeds the wanted/reasonable level. In order to compensate these defects there has been research on the area of delivering information to a large number of users. This research has brought up a new group transmission method called: Multimedia Broadcast Multicast Services (MBMS), [14].

The aim of MBMS transmissions is to send identical information from single source to vast amount of recipients with high data rates by using minimum amount of radio resources. MBMS efficiency is based on the possibility to use ei-

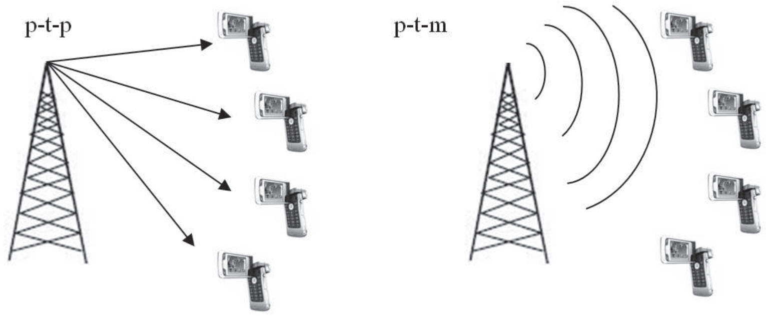

FIGURE 27 Different Ways to Deliver Data with MBMS

ther shared or dedicated resources, i.e., *point-to-multipoint (p-t-m)* or *point-to-point (p-t-p)* transmission modes for the content delivery as shown in Figure 27. Point-to-point transmissions are used to transfer both MBMS specific control/user plane information as well as dedicated control/user plane information between network and single UE. Point-to-multipoint transmissions are used to transfer MBMS specific control/user plane information between the network and several UEs. In general p-t-p connections are appropriate to be used when only a few users in a cell want to use MBMS services and p-to-m connections are suitable when a large number of users want to view the identical content. From the network point of view there exists so called counting procedures which allow network to figure out whether services should be configured as p-t-p or p-t-m in a particular area. Thus, UE might have to deal with dynamic changes between dedicated and common resources when crossing the cell edge.

As MBMS can use bandwidth more efficiently and is also capable of delivering multimedia content with different modes (multicast/broadcast), also the scope of the possible services is much larger than with CBS or IP-MS. Services like mobile-TV and delivering real-time sport results can be considered examples of such services.

4.2 Services

Structure of MBMS service provision is two-fold: one for multicast mode and the other for broadcast mode. According to 3GPP's specifications [11], MBMS multicast services contain following stages: subscription, service announcement, joining, session start, MBMS notification, data transfer, session stop and leaving as depicted in Figure 28. MBMS broadcast mode from service provision point of view is actually special case of multicast mode service provision, i.e., it does not contain stages initiated by the users (subscription, joining, leaving) as illustrated in Figure 29.

Figure 30 presents the use of different stages of MBMS services in a timeline example. As it can be seen from the top left corner of that figure, user can sub-

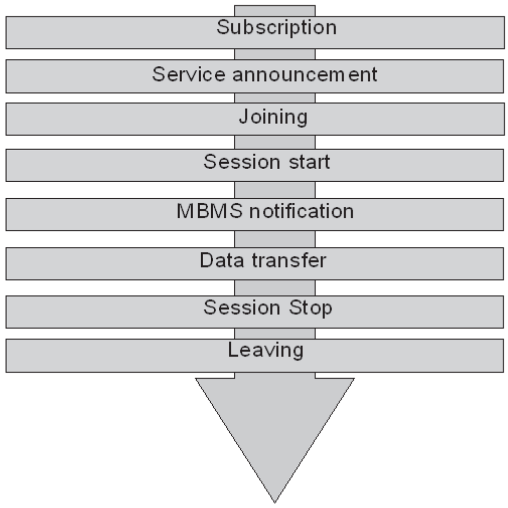

FIGURE 28 Basic Structure of MBMS Multicast Service Provision [11]

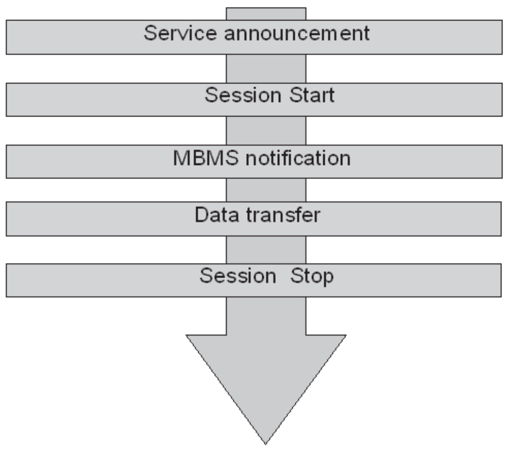

FIGURE 29 Basic Structure of MBMS Broadcast Service Provision [11]

scribe service at any time. By subscribing a certain service(s) user establishes the relationship to the service provider by making agreement of receiving service(s) offered by the operator/service provider. After the subscription has been made, according to [11], service provider starts the service announcement which is used to distribute information about the service to the users. Information about the service contains the parameters required for service activation (e.g., IP multicast address(es)) and possibly other service related parameters such as service start time. Moreover, as Figure 30 shows, after service announcement has been made by the service provider, users can join to the service group at any point they like which means that the MBMS service stages can overlap.

After subscribing, service announcement and joining phases session start is the point when RNC receives a request from the CN to start the session. Session start is independent of the actions made by users and therefore users can join in to the multicast group even during of a MBMS session. After session start request has been received MBMS notification procedure is required to be used to inform member of multicast group about forthcoming (and potentially about ongoing) MBMS multicast data transfer. This procedure must be transmitted so that every UE in MBMS service area can receive it as it contains vital information such as MBMS radio bearer setup information. Notification procedure should also allow terminals to minimize their power consumption, meaning that UEs with an activated MBMS service should not listen to it constantly, but on regular intervals.

After the data has been transferred and the RNC receives information from the CN that there will be no more data to send for some period of time session, stop phase is carried out. During session stop phase radio access bearers are released.

Even though session has been stopped users are not obligated to leave the group because another session might start after a small period of time. However, if service provider makes a service stop announcement then users must leave the MBMS multicast group. More detailed information about MBMS services can be found in [11] and [9].

4.3 Quality of Service

The purpose of this section is to discuss a few of the quality of service related aspects when MBMS is considered. Some of the discussed matters are set by 3GPP as requirements for MBMS, [15].

As mentioned earlier in this thesis, MBMS data transfer is specified to be unidirectional traffic and to be more precise downlink only. This is specified in [15] along with the fact that QoS attributes shall be the same for MBMS multicast and broadcast modes. One of the most important QoS aspects is caused by the lack of uplink traffic as it discloses such things as feedback and individual retransmissions. Thus, the reception of MBMS content is not guaranteed. However,

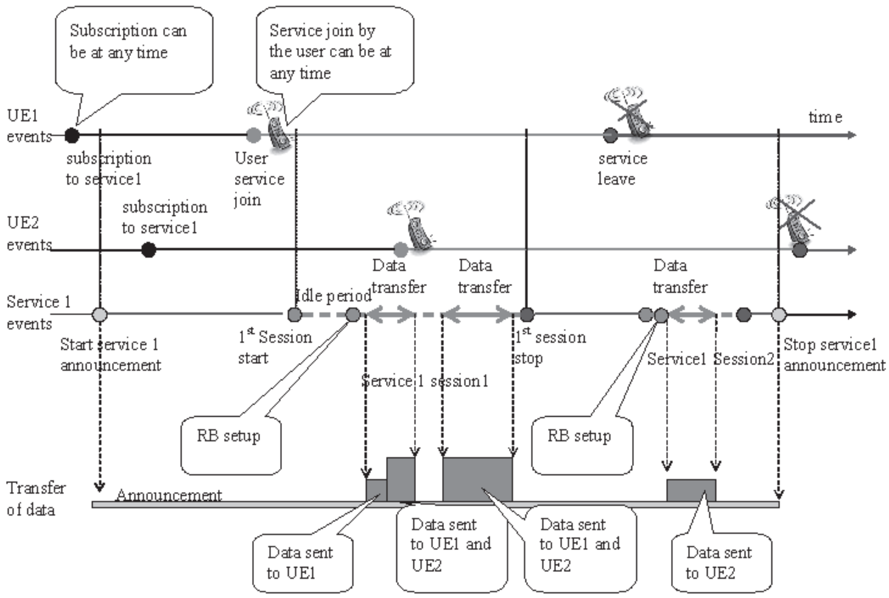

FIGURE 30 Timeline Example of MBMS Multicast Service Provision, [11]

reliability of MBMS transmissions is improved, e.g., by using periodic repetitions of MBMS content from the service provider so that the number of satisfied users can be maintained at a reasonable level. Repetitions are not precluded by the lack of uplink traffic because the service provider can transmit them without feedback from the UE. Periodical repetitions are done on RLC level with identical RLC sequence numbers and *Protocol Data Unit (PDU)* content. As data loss is required to be minimal also during cell change, there has been made effort to achieve this, e.g., by using soft and selective combining. Moreover, also antenna diversity techniques can be considered as an option to improve the reliability. Thus, this thesis aims to investigate the applicability of those improvements.

4.4 Architecture

In MBMS architecture existing packet switched domain entities such as *Gateway GPRS Support Node (GGSN)*, *Serving GPRS Support Node (SGSN)*, UTRAN and UE are included/enhanced to be able to provide the MBMS bearer services. 3GPP's reference model for MBMS architecture is described in [11] and illustrated in Figure 31. MBMS related entities and their functionalities are discussed briefly in the following.

Starting from the left side of the Figure 31, the first entity involved in MBMS operation is UE which is operated by the end user. User will use UE to activate and deactivate MBMS services and no other explicit user requests are required. Depending on UE's capabilities, it is possible to support receiving of MBMS and

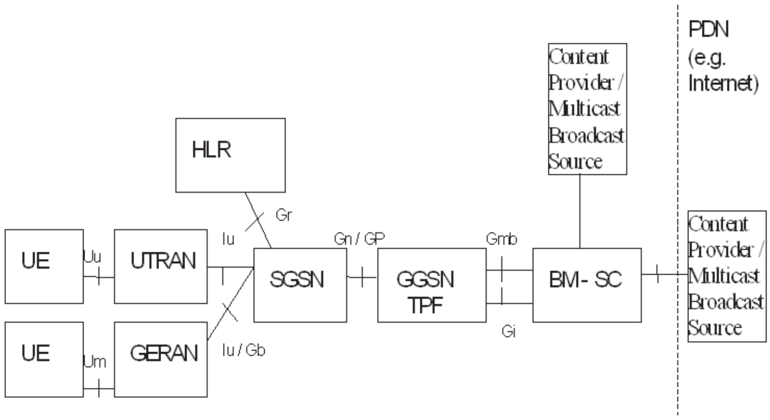

FIGURE 31 MBMS Architecture; Reference Model, [11]

non-MBMS services simultaneously as well as multiple MBMS services. Also, some MBMS related security issues are needed to be dealt with in UE side.

The next entity after the UE is the radio access network, i.e., *GSM/EDGE radio access network (GERAN)* or UTRAN, see Section 2.1 for further details on UTRAN. From the MBMS point of view, GERAN/UTRAN is responsible for delivering the MBMS related data efficiently to designed service area. Efficient delivery consist, e.g., of data reception from the core network, initializing and terminating MBMS services, choosing appropriate radio bearer and channels.

Core network to which UTRAN is connected via lu interface consist of SGSN, *Home Location Register (HLR)*, GGSN and *Broadcast-Multicast Service Centre (BM-SC)*. In addition to entirely new MBMS-specific BM-SC modifications are needed to SGSN and GGSN.

The SGSN's role within the MBMS architecture is to perform MBMS bearer service control functions and to provide MBMS transmissions to UTRAN/GERAN. The SGSN is also responsible for handling mobility functions in MBMS by supporting intra-SGSN and inter-SGSN procedures. Generating charging data per multicast MBMS bearer service for each user is also part of SGSN's tasks.

The GGSN's role in MBMS is to serve as an entry point for IP multicast traffic. According to [11], upon notification from the BM-SC the GGSN must be able to request the establishment of a bearer plane for a broadcast or multicast MBMS transmission. Furthermore, upon BM-SC notification the GGSN must be able to tear down the established bearer plane.

Finally, the element between content provider and GGSN is the broadcast-multicast service centre. Generally BM-SC is responsible for providing functions for MBMS user service provisioning and delivery. In [11] it is described that BM-SC may provide the following actions:

- Serve as an entry point for content providers MBMS transmissions,
- authorize and initiate MBMS bearer services, and

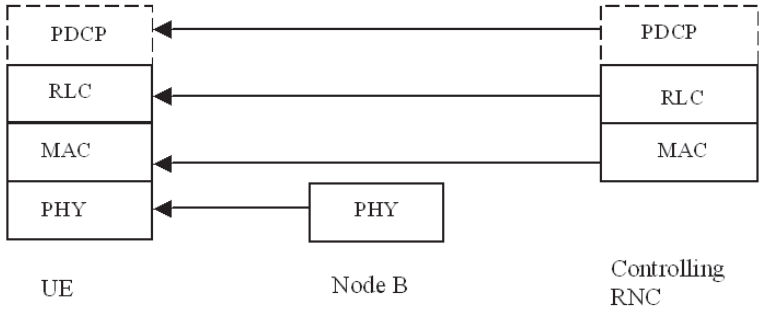

FIGURE 32 Protocol Stack for MBMS Traffic Channel, [9]

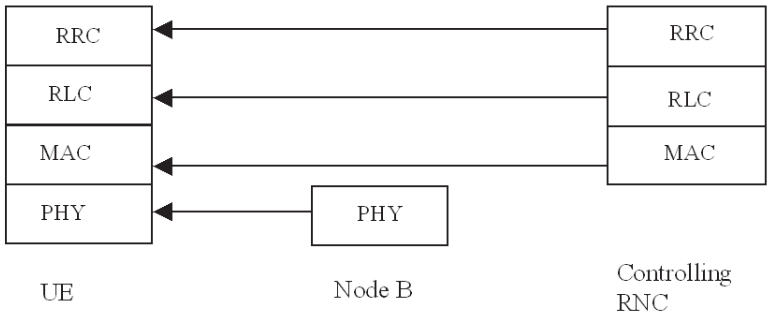

FIGURE 33 Protocol Stack for MBMS Control Channel, [9]

- schedule and deliver MBMS transmissions.

4.5 Protocols

The purpose of this section is to cover MBMS protocol stack very briefly. More information of MBMS protocol stack can be found from [9] and additional details of basic WCDMA protocol stack layers (RLC, MAC and *Physical Layer (PHY)*) from [45].

Figure 32 illustrates the protocol stack that is used for data transmissions and Figure 33 illustrates protocol stack for control information. As those figures show, only the top layers differ in these two protocols stacks. Starting from the common layers, the bottom layer is the physical layer (PHY). PHY handles the transmission of signals and the activation and deactivation of connections. Next two layers on top of that are MAC and RLC layers that form the data link layer of the UMTS radio network. Those layers handle access to shared media so that the transmission path should always be available during transmission. *Packet Data Convergence Protocol (PDCP)* layer that is the top layer for data transmission protocol stack is responsible for header compression/decompression for the MBMS data traffic. The top layer showed in Figure 33 is used for radio recourse control with MBMS transmissions.

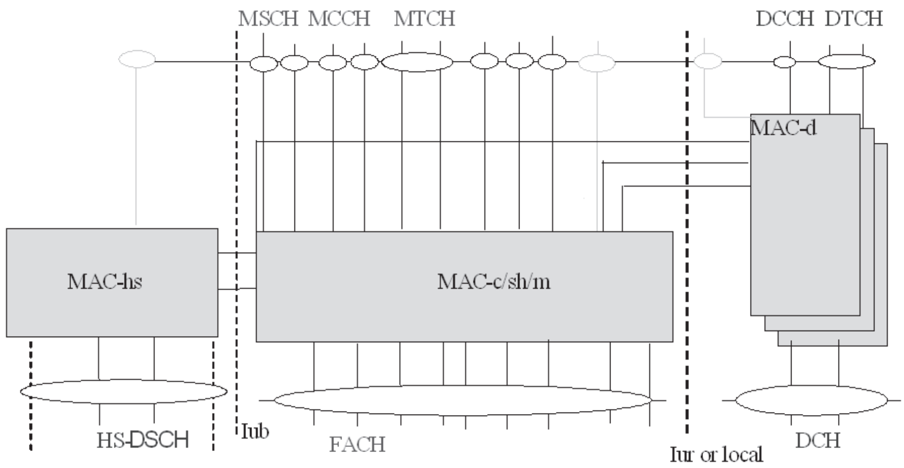

FIGURE 34 Simplified UTRAN MAC Architecture with Logical MBMS Channels, [9]

4.6 Channels

Logical MBMS channels can be categorized according to different transmission modes (p-t-p and p-t-m). In this section channel structures for both of those modes will be presented and they are discussed with more details in [9].

MBMS p-t-p transmissions use the same logical, transport and physical channels that are used in normal p-t-p connections in Rel'99 and onwards. Logical channels used for p-t-p transmissions are *Dedicated Control Channel (DCCH)* and *Dedicated Traffic Channel (DTCH)* which are mapped on DCH. P-t-p transmission mode over HSDPA is also supported. The transport channel for carrying the user data under HSDPA is HS-DSCH. Logical and transport channels used for MBMS p-t-p transmissions are illustrated in Figure 34.

MBMS transmissions which are intended to use p-t-m transmissions have somewhat different channel structure than in p-t-p connections. This new channel structure introduces three new logical MBMS channels which are mapped into forward access channel. According to [9] logical channels reserved for MBMS p-t-m transmissions consist of *MBMS Control Channel (MCCH)*, *MBMS Scheduling Channel (MSCH)* and *MBMS Traffic Channel (MTCH)*. These channels are also illustrated in Figure 34.

MCCH is used for a p-t-m downlink transmission of control plane information between network and UEs while MSCH is used for transmitting MBMS scheduling information. Finally, MTCH is intended for a p-t-m downlink transmission of user plane information between network and UEs.

Transport channel used to carry those three logical channels is FACH. The FACHs where MBMS p-t-m logical channels are mapped are explicitly signaled. FACH is a natural choice for MBMS purposes as it can be used for carrying control information and secondly be used for transmission of packet data. There can also be more than one FACH in a cell, which makes it possible to have variety of bit rates for MBMS transmissions. Physical channel used to carry FACH is called *Secondary Common Control Physical Channel (S-CCPCH)*.

The main differences between Rel'99 dedicated channels (p-t-p transmissions) and common channels (p-t-m transmissions) are that the FACH (common channel) does not use fast power control and it does not support soft handover. Lack of fast power control means that FACH is transmitted with constant power and therefore the average power can be higher than it would be, e.g., with a single DCH.

4.7 Mobility

This section is aimed to provide information of issues related to mobility from the MBMS point of view. As this thesis focuses on p-t-m connections the mobility of p-t-p connections are not discussed separately in this section, but information about that can be found in [9].

When UE comes to a point when it needs to reselect the cell due to mobility or because it returns to an 'on-service' state from an 'out-of-service' state, UE needs to acquire the MBMS control channel information of the service that it is interested in, provided that the service is available in the selected cell. The service is available when the session has already been started or the service can be initiated in the cell if the UE requests it. According to the 3GPP's TS 25.346 [9] if the MBMS service is available in the cell, the UE will perform an action for the service reception in the cell. Thus, the network side is not responsible of the service continuity any other way than keep sending the MBMS data.

Whenever the UE moves between p-t-m cells while continuing to receive a service, UE shall receive MCCH information in a new cell, which includes an MBMS cell group identity. If a UE moves between cells which belong to the same MBMS cell group based on the MCCH information, the UE does not need to re-establish RLC entity and re-initialize PDCP entity for the service received on MBMS traffic channel. If a UE moves between cells belong to different MBMS cell groups based on the MCCH information, the UE shall re-establish RLC entity and re-initialize PDCP entity for the service received on MTCH.

From the radio access point of view UE mobility is a bit more problematic, e.g., because soft handovers cannot be utilized in MBMS. The use of soft handover is out of the question because of different kinds of resources (dedicated/shared) used in MBMS and also because of the lack of uplink traffic. Lack of uplink traffic leads to that the UE cannot inform Node-B of its measurements, which are needed in soft handover. Therefore, initially it was defined that MBMS must apply hard handovers. However, to improve the situation especially for the cell-edge users macro diversity combining has been considered as viable option in terms of MBMS, see Section 2.6.

5 VOICE OVER IP

Voice over Internet Protocol is commonly referred as technology that includes only the transportation of speech over packet switched networks. However, in reality VoIP covers not only speech but many value added services combination of which could be referred as rich calls. Speech or voice is indeed the main service but VoIP employs the possibility for video feed, presence information, instant messaging and others. Even though VoIP brings a large variety of benefits, VoIP also introduces some challenges to tackle with. VoIP does not tolerate high level of packet loss, delays nor jitter (i.e., delay variance). These challenges can be especially problematic in the perspective of wireless cellular networks. The purpose of this section is to shortly introduce VoIP, especially in terms of wireless cellular networks.

5.1 Background

Voice over IP or VoIP protocols were used to carry voice signals over the early internet for the first time in 1973. Those trials were for non-commercial use and the first commercial software for end-users was made available much later in the mid 1990's. Company called Vocaltec released software called 'Internet Phone' in 1995 which did not still allow gateway to analog networks but communication between other 'Internet Phone' users was possible. Other competing companies that were working on similar products introduced their solutions subsequently. These included Nuera, Net2phone, Cisco, Clarent, Vienna Systems, etc. At the late 90's after the possibility to make also digital to analogue calls VoIP was getting more and more popular as people where able to make, e.g., international calls with fraction of the cost of traditional calls.

VoIP-trend from the fixed networks is nowadays starting to emerge also in the wireless cellular networks as the evolution of packet data technologies has made it possible to carry the voice efficiently over packet switched domain instead of traditional circuit switched solutions. In addition to mere technology, flat rate pricing of wireless data connections has a role in pushing more and more users using voice over IP due to lowered costs. VoIP does not benefit only the end users as VoIP has introduced the possibility to move into All-IP networks instead of using separate CS network elements for voice and PS elements for data. This would lead to cost savings for the operators as the CS related part of the core network would not be needed anymore. These cost savings along with the possibility to implement rich call services ([44]) with less complexity have increased interest toward VoIP. The growing interest toward VoIP has made VoIP very likely feature in the cellular networks which is proven by the fact that upcoming LTE networks support only packet based voice.

5.2 Quality of Service

Generally requirement for successful penetration of VoIP is that the voice quality should be comparable to what is available using the *Public Switched Telephone Network (PSTN)*, i.e., (roughly) traditional circuit switched voice. There are numerous factors affecting VoIP quality of service some of which are addressed shortly in this section.

Coder Decoder (Codec) is a procedure to digitalize voice signal using as few bits as possible still maintaining a reasonable level of speech quality. Codec should thus work so that it gives good compression in order to save transmission resources, provide good enough voice quality so that speech is still understandable and recognizable and finally this procedure should introduce as low additional delay as possible so that mouth-to-ear delay does not get excessively long. Standardization bodies like *International Telecommunication Union (ITU)* and 3GPP recommends the use of certain codecs, e.g., for all wide area networking applications ITU recommendation is G.729 codec and for cellular networks 3GPP recommend *Adaptive Multi-Rate - Wideband (AMR-WB)* codec [22].

Echo cancellation is also critical to be compensated in systems where the delay is a factor, VoIP over cellular networks is such a system. Echo is caused by reflections of the signal which can result in hearing one's own voice repeated. In other words, an echo is the audible leak-through of your own voice into your own receive (return) path. If one-way delay exceeds 25-150 ms the echo becomes audible and thus cancellation is required [33].

One of the most critical factor effecting VoIP quality is the delay. Delay can be divided into total transmission delay and to delay jitter. Total transmission delay is the sum of the compression, decompression, processing, buffering/queuing, transmission delay and the network delays. According to ITU e-model [61], the maximum acceptable mouth-to-ear delay for voice is on the order of 250 ms as the reference values in Figure 35 show. In the respect of HSPA networks assuming the delay for core network is approximately 100 ms, the tolerable delay for air interface should be strictly lower than 150 ms. Hence, assuming that both users are UTRAN users, tolerable delay for RLC/MAC buffer-

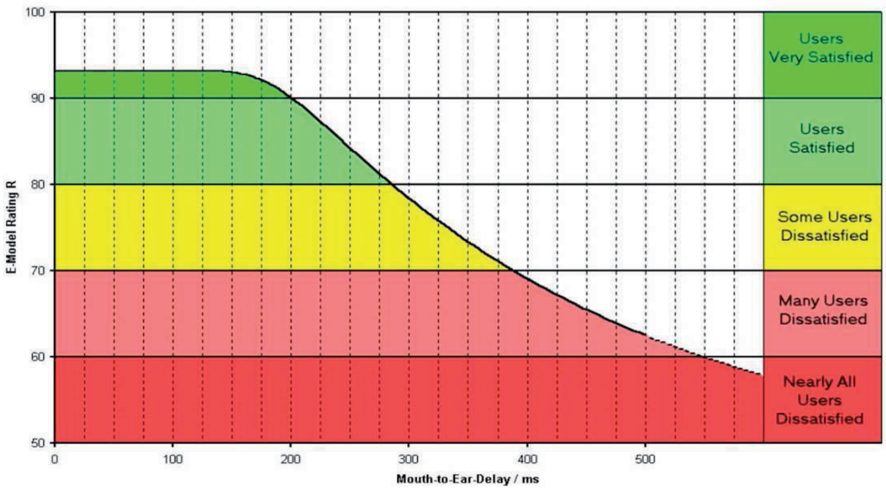

FIGURE 35 E-Model Rating as a Function of Mouth-to-ear Delay [ms], [61]

ing/scheduling and detection is below 80 ms. If total delay exceeds 200-250 ms, the two speakers may result in stepping on each other's messages which naturally does not correspond to the requirement introduced before about VoIP QoS should match the quality of PSTN voice. Total transmission delay (and jitter to some extend) can however be managed/controlled, for instance, by prioritizing VoIP packets.

Delay jitter is the variance in arrival times of the transmitted packets. Thus, to allow variable packet arrival time and still produce a steady outgoing stream of speech the receiving end does not play the speech as soon as the first packet arrives. Instead packets are held in the jitter buffer for a while and then played. When a packet does not arrive in time to fit into the voice stream played for the receiver it has to be discarded resulting in disruptions in voice.

From the perspective of this thesis codecs are taken into account in packet sizes and in interarrival times. The packet corruption and total transmission delay is monitored and the maximum end-to-end delay is assumed as described before. Echo cancellation on another hand is assumed to be ideal and jitter is also assumed to be very minimal. More details on the VoIP modeling aspects are presented later in this thesis.

Finally, the public Internet and wireless radio access environment produces substantial packet corruption and loss. In the respect of voice, packet retransmissions from the transmitting source are not possible due to delay and thus techniques such as HARQ can compensate for the corrupted or missing packet, see Section 3.2.

5.3 Protocols

Forming, maintaining and releasing VoIP calls requires both signaling and transport protocols. Signaling protocols are used for, e.g., forming and releasing connections, resolving endpoint addresses and capabilities, selecting the media for the session and conveying session state information. The most notable signaling protocol generally associated with VoIP is *Session Initiation Protocol (SIP)* [52], which has been standardized by *Internet Engineering Task Force (IETF)*. Transport protocols on another hand are needed for the actual media transport. Generally, IETF's *Real Time Transport Protocol (RTP)* [54] is utilized as a transport protocol. The purpose of the following subsections is to present SIP and RTP protocols briefly in addition to an overview of (robust) header compression which is imperative for services such as VoIP.

5.3.1 Session Initiation Protocol

Session initiation protocol is a peer-to-peer application-layer protocol that can establish, control, and terminate multimedia sessions with one or more participants. According to *Request For Comments (RFC)* 3261, [52], SIP supports following five facets of establishing and terminating multimedia communications:

- User location, i.e., determination of the end system to be used for communication.
- User availability, i.e., determination of the willingness of the called party to engage in communications.
- User capabilities, i.e., determination of the media and media parameters to be used.
- Session setup, i.e., 'ringing', establishment of session parameters at both called and calling party.
- Session management including transfer and termination of sessions, modifying session parameters, and invoking services.

SIP is independent of the underlying transport protocol and thus, e.g., *Transmission Control Protocol (TCP)* or *User Datagram Protocol (UDP)* can be used. In addition to transport protocol, SIP works in conjunction with other protocols such as *Session Description Protocol (SDP)* and *Media Gateway Control Protocol (MEGACO)* which is used for controlling gateways to the public switched telephone network. SDP, [53], which is carried by the SIP describes the session details such as:

1. Session name and purpose,
2. Time(s) when session is active,

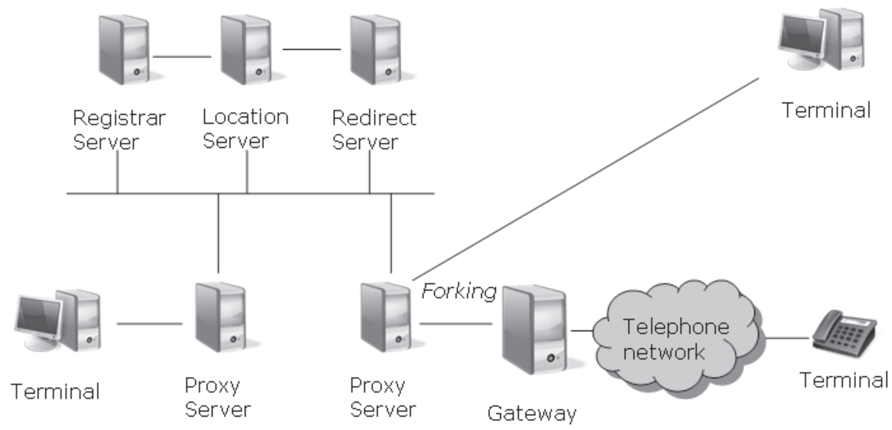

FIGURE 36 SIP Architecture

3. Which media are included in the session, and
4. Information needed to receive those media (addresses, ports, formats, etc.).

Architecture of SIP, illustrated in Figure 36, consists essentially of user agent and network servers. User agent (or end terminal) usually consists of two parts: *User Agent Client (UAC)* and *User Agent Server (UAS)*. UAC is an application that initiates and sends SIP requests and UAS receives and responds to SIP requests on behalf of user; accepts, redirects or refuses calls. User agent is generally formed of both UAS and UAC as the user most likely wants to initialize and accept calls.

Network servers on another hand contain four different types of logical servers: proxy, redirect, registrar and location. Proxy is an intermediary device that acts as a client and server, and which makes requests on behalf of other clients. Redirect server accepts SIP requests but rather than passing these onto the next server it maps the address into zero or more new addresses and returns those addresses to the client who made the request. Registrar server accepts REGISTER requests, and then places that information into the location service for the domain it handles. Location server provides information about a caller's possible locations to redirect and proxy servers.

Servers can operate in two different modes: stateful and stateless. Stateful mode means that server remembers the incoming requests it receives, responses it sends back and the outgoing requests it sends on. Stateless mode means that server forgets all information once it has sent a request/response.

5.3.2 Real-time Transport Protocol

Real-time transport protocol [54] is an IETF protocol used to transport real-time data, including audio and video. RTP can be used for media-on-demand as well as interactive services such as VoIP. RTP actually consists of data and control parts. Data part is referred as RTP and the control part as *Real-time Transport Control Protocol (RTCP)* which allows monitoring of the data delivery and provides

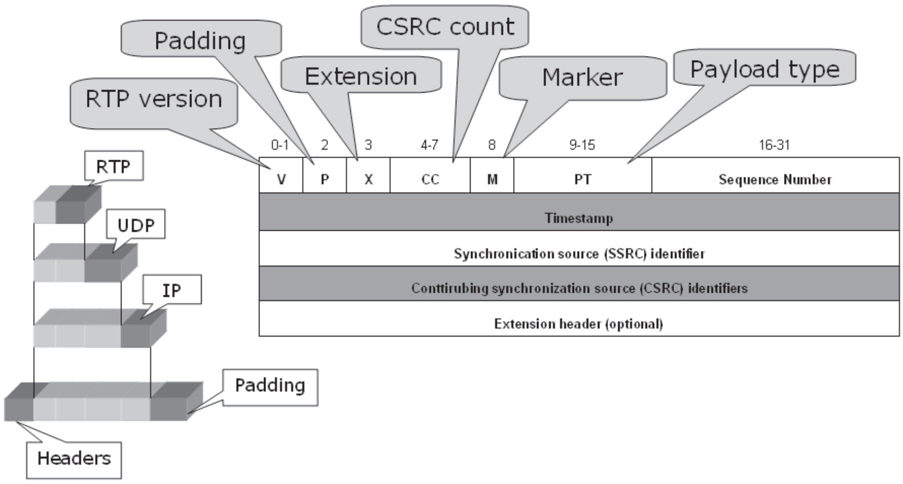

FIGURE 37 RTP Frame Structure

minimal control and identification functionality.

The RTP frame structure illustrated in Figure 37 is defined to inform the receiving end of issues such as:

- Traffic/payload type which can be selected from various available formats that are defined in their own standards, e.g., [55], [56] and [57].
- Sequence number which allows the receiver to reconstruct the sender's packet sequence.
- Time stamp which makes possible to correctly synchronize the media stream, e.g., voice and corresponding video.

What RTP does not provide is mechanisms to ensure timely delivery, guarantee delivery of packets, prevent out-of-order delivery or provide any other quality-of-service guarantees. This is because RTP is actually session layer protocol and thus it relies on lower-layer services to handle previously mentioned properties.

Much like with SIP underlying transport protocol for RTP can be either UDP or TCP as both of them fulfill the RTP requirement that underlying protocol must provide multiplexing of the data and control (RTCP) packets. For delivering audio and/or video for playback, TCP may be appropriate. Moreover, TCP is often run over highly lossy networks to reach acceptable throughput. However, for real-time delivery of audio and video, TCP and other reliable transport protocols are inappropriate. The main reason for that is that delay-sensitive data such as real-time audio and video suffer from retransmissions caused by reliable transport protocols. Retransmissions cause problems as when the sender has discovered the missing packet and retransmitted it at least one round-trip time (likely more) has elapsed. In addition congestion control mechanisms such as ’slow start’ in TCP can cause problems by ’starving’ the receiver with sudden decrease of transmit rate.

As mentioned earlier Real-time control protocol is part of RTP. RTCP, [54], is used periodically to transmit control packets to participants in a session by using the same distribution mechanism as the data packets use. RTCP is used for:

- Feedback of the quality of service which is the primary function of RTCP.
- Transport alias name of sender (CNAME) which are needed if connection is interrupted or there are conflicts in *Synchronization Source (SSRC)* identifications.
- Control bandwidth used by RTCP which is recommended to remain under or equal to 5 % of session bandwidth.
- Give information about participants, for instance convey participant identification to be displayed in the user interface.

5.3.3 Header Compression

RTP headers altogether consume unnecessary bandwidth compared to the small size of voice packet as demonstrated in Eq. 4:

$$RTP + UDP + IPv4 = 12 + 8 + 20 = 40bytes = 320bits \tag{4}$$

This problem is solved with header compression. Headers can be reduced to only 2-4 bytes with *Compressed RTP (CRTP)* technique [58] and *Robust Header Compression (ROCH)* [59], developed with wireless links as the main target, can compress headers down to 1-3 bytes.

General idea of the compression algorithms, such as CRTP, is based upon the observation that many headers remain constant over the life of the connection. Thus, after sending the uncompressed header once, these fields may be elided from the compressed headers that follow. However, for RTP header compression the main gain comes from the observation that although several fields change in every packet, the difference from packet to packet is often constant and therefore the second-order difference is zero. For instance, for most of the packets, only the sequence number and the timestamp will change from packet to packet. If packets are not lost or disordered upstream from the compressor, the sequence number will increment by one for each packet.

However, some characteristics of wireless links such as lossy behavior and relatively long *Round-trip time (RTT)* add new requirements for compression schemes. Compression scheme must be able to handle loss on the link between the compression and decompression points as well as loss before the compression point [59]. Robust header compression introduced new compression mechanisms with the primary objective to achieve the combination of robustness against packet loss and maximal compression efficiency.

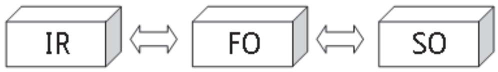

FIGURE 38 ROCH State Transitions

Roughly speaking, the performance of ROCH is based on three different states: *Initialization and Refresh (IR)*, *First Order (FO)* and *Second Order (SO)*. Transitions to and from these different states is illustrated in Figure 38.

According to [59], the purpose of the IR state is to initialize the static parts of the context at the decompressor or to recover after failure. Initialization or recovery is done by sending the complete header information. The compressor stays in the IR state until it is fairly confident that the decompressor has received the static information correctly. The purpose of the FO state is to efficiently communicate irregularities in the packet stream. In FO state, the compressor rarely sends information about all dynamic fields, and the information sent is usually compressed at least partially. Compressor stays in the FO state until it is confident that the decompressor has acquired all the parameters of the new pattern. Some or all packets sent in the FO state carry context updating information. It is very important to detect corruption of such packets to avoid erroneous updates and context inconsistencies. Finally, when SO state is used the compression is optimal. The compressor leaves SO state and goes back to the FO state when the header no longer conforms to the uniform pattern and cannot be independently compressed on the basis of previous context information.

5.4 IP Multimedia Subsystem

IP Multimedia Subsystem is an architectural framework to provide a secure and reliable means for wireless and wireline terminals and applications to reach, negotiate and communicate with each other. IMS was introduced for the first time in 3GGP's Rel'5. In the first IMS release UMTS was recognized as the only access methodology but later it was updated to become access independent solution [10]. Therefore, with the help of IMS one can easily deploy services to make, e.g., VoIP call from cellular phone to laptop with *Wireless Lan (WLAN)* connection, as illustrated in Figure 39. As IMS is highly recommended part of the architecture where VoIP is employed, the purpose of this section is to briefly describe core elements and functionalities of IMS in the respect of VoIP. Further details of IMS can be acquired, e.g., from [10] and [38].

Roughly speaking, IMS is a horizontal control layer that isolates the access network from the service layer and specifies interfaces that services can utilize easily, as depicted in Figure 40. IMS tries to utilize the IETF's 'Internet standards' as far as possible for the cases where an IETF protocol has been selected. SIP is an example of IETF protocol that is supported by IMS.

As Figure 40 illustrates the core of IMS comprises of two main nodes: the

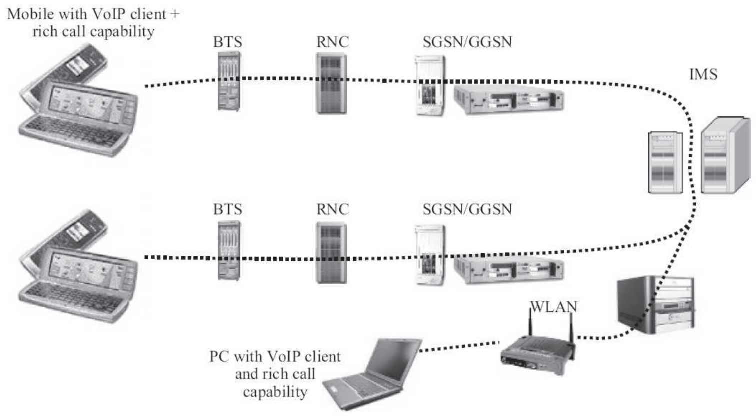

FIGURE 39 Example of IMS System, [46]

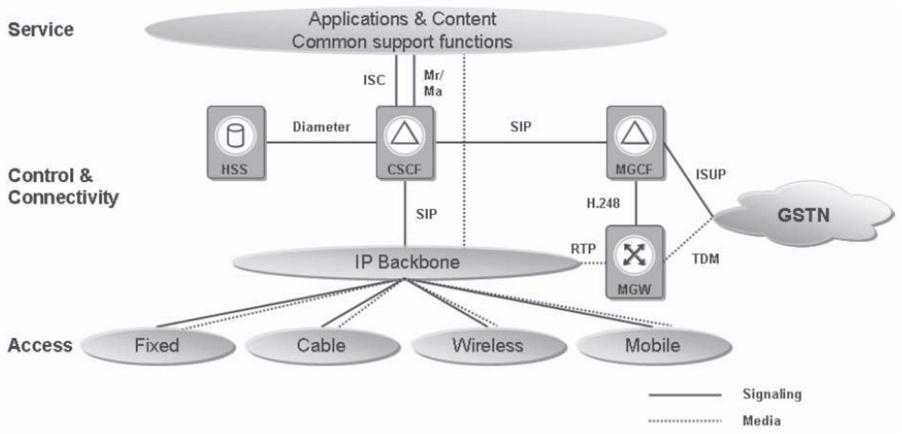

FIGURE 40 IMS Architecture Overview, [38]

Call Session Control Function (CSCF) and the *Home Subscriber Server (HSS)*. In order to be able to provide connections to analogue telephone networks IMS can be connected to *General Switched Telephony Network (GSTN)* or PSTN through interworking functions by *Media Gateway Control Function (MGCF)* and *Media Gateway (MGW)*.

The main function of the CSCF is to manage the SIP sessions and coordinate with other network elements for session control, application/service control and resource allocation. CSCF has also the responsibility for interacting with the HSS which is responsible for identification handling, access authorization, authentication, mobility management in addition to support for session establishment, service provisioning and service authorization. As shown in Figure 41 CSCF can play three different roles: *Serving-, Interrogating and Proxy- Call Session Control Function (S-, I- and P-CSCF)* to handle previously presented tasks.

S-CSCF is the central node of the signaling plane. It provides the mechanism for authenticating the IMS subscriber in the core network and maintains the IMS

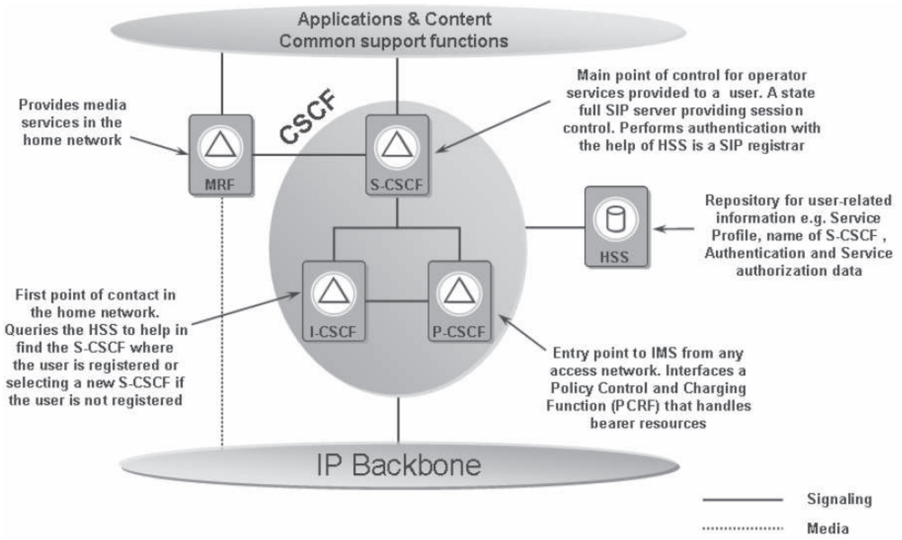

FIGURE 41 Common IMS Core Overview, [38]

subscriber's network registration. The S-CSCF also enforces the operator's policy because it ensures that the media, which are requested by the user through SDP are within the boundaries of the subscriber's profile.

P-CSCF serves as the initial point of contact for the terminal to the network. P-CSCF handles the registration procedure of UE to an appropriate I-CSCF. After the registration P-CSCF sends all subsequent SIP messages received from the UE to the S-CSCF. Other important functions of P-CSCF are for instance establishing an IPsec security association, compressing and decompressing SIP messages to save radio resources and generating charging records.

Main task of the I-CSCF is to select an S-CSCF which is responsible to handle the SIP session within an operator's home domain. All incoming SIP messages from visited or other IMS home domains belonging to the same session are forwarded to the selected S-CSCF.

6 RESEARCH TOOLS

The research tools used to perform studies regarding this thesis are two comprehensive system level radio network simulators. Simulators are designed for studying RRM algorithms to obtain capacity and coverage estimates, to support standardization work and network planning by optimizing and tuning the radio network parameters. Simulation tools enable detailed simulation of users in multiple cells with realistic call generation, propagation and fading (adopted from [21]).

There are at least two factors advocating the usage of system simulations: on the one hand, the complexity of the radio network behavior makes it virtually impossible to calculate accurate analytical results, and on the other hand, physical network trials would be time consuming and expensive. Thus, full system level simulations where RRM functionalities and their inter-actions are taken explicitly into account are essential to the performance evaluations. Moreover as the used simulators are highly configurable with parameters for network traffic, user terminals, network configuration, RRM algorithms and physical layer in addition to vast expandability possibilities those are truly indispensable tools for studying radio network performance and enhancements.

The modeling and reliability analysis of the used system simulators have been presented earlier in many international articles and conferences, see, e.g., [50], [83], [87] and [39] as well as in doctoral theses such as [69], [63] and [49]. The reliability analysis of the tools is addressed in the Appendix 1 at the end of this thesis. The purpose of the following sections is to address modeling issues briefly with special focus on the aspects that this thesis addresses.

6.1 Simulator Types

Implementation of the used system level simulators themselves are a compromise between building an accurate model of a WCDMA/HSPA network and keeping manageable computing efforts. In order to reduce the complexity and computa-

TABLE 2 System Simulator Classification and Main Attributes

	Static	Quasi-Static	Dynamic
Algorithms	Simplified, e.g., no OLPC	Extensively modeled RRM	Fully modeled RRM
Slow Fading	Fixed	Fixed	Varying
Fast Fading	Fixed	Varying	Varying
Time Domain	No	Yes	Yes
Mobility	Static	Static	Yes
Computational Complexity	Low	Middle	High
Use Cases	Coverage predictions, etc.	RRM study excluding explicit mobility, etc.	Mobility study, RRM study, etc.

tional requirements of the simulator, simulators have been divided into link and system level simulators. In link level simulator connection between one UE and base station is modeled in very detailed level (chip or symbol level). Then produced link level data of received *Frame Error Rate (FER)* (or *Bit Error Rate (BER)*) vs. *Energy per Bit to Noise Ratio* (E_b/N_o) performance is used through a so called *Actual Value Interface (AVI)* in the system level simulator as presented in [51]. Link level simulators are out of the scope of this thesis with the exception of AVI usage.

System level simulators can be further classified into three main types: *static* ('snapshot'), *quasi-static*, and *dynamic*. Main attributes for those different system simulator types are listed in Table 2. As that table shows, static simulators provide quite fast but simplified solution for quick evaluations, see, e.g., [68]. For both static and quasi-static approaches statistical confidence is reached through running multiple drops, i.e., independent simulation iterations. In each step terminal locations, slow fading, etc. are randomized but the statistics are gathered and averaged over all drops. In the dynamic simulations the statistical confidence is achieved through longer simulations and through the fact that terminals are actually moving within the scenario.

For the scope of this thesis the quasi-static and dynamic simulator types are used. Dynamic simulations to have more detailed evaluation of mobility scenarios and somewhat faster quasi-static simulations to test the performance of different concepts, such as dual carrier.

6.2 Simulation Flow

Simulation flow includes three fundamental phases: *setup*, *simulation*, and *destruct*. In quasi-static approach also *reset* phase exists, see Figure 42. During the first phase all of the objects including the simulation scenario, RNCs, Node-Bs and UEs are generated according to user defined parameters. In the second

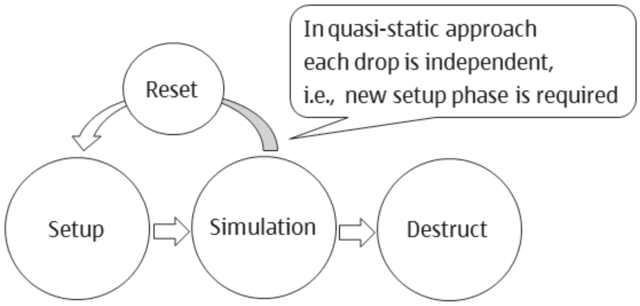

FIGURE 42 Simulation Flow

phase the actual simulation procedure is completed which starts with a warm up period. Warm up period generates the initial load for the network which should reach the level of realistic network. The gathering of statistics starts after the warm up period. Simulation runs in slot resolution, i.e., in every $0,667$ ms all relevant actions and statistic gathering are performed. Relevant actions in every simulation slot contain, e.g., interference calculation between all active connections, measurements for handover (in dynamic simulations) and power control and so on. All elements involved in simulated system (UEs, Node-Bs, RNCs) go through their own actions before control is moved to the next time slot. The second phase will be active until the user defined last time slot for simulation/drop is reached. After the simulation phase, in quasi-static approach the flow continues back to setup phase through reset providing that there are more drops to be run. In reset phase UEs and existing connections are destroyed and necessary statistics saved. After the second phase is completed, final statistics will be saved and then the simulation is finished by destroying all created elements.

6.3 MBMS Modeling

6.3.1 MBMS Sessions and Subscribers

Simulator supports static and dynamic call setup processes, i.e., the number of terminals/calls can either stay static for the duration of the simulation or change dynamically. Dynamic number of terminals/call is defined by the number of subscribers, call arrival rate and average call length.

From the perspective of MBMS sessions, two alternatives exist as well. The whole duration of simulation can be considered as one MBMS session or it can consist of multiple MBMS sessions. If the whole simulation duration is considered as one MBMS session then in the beginning of the simulation static number of terminals are created, which model the users who have subscribed the MBMS service. All the calls last as long as the simulation lasts and thus the MBMS service ends when the simulation ends. In the other alternative, in the beginning of the simulation static number of terminals are created, which again presents the users who have subscribed to MBMS service. Fixed MBMS service duration, i.e., call duration is defined for each terminal. All calls end at the same time, thus the new MBMS service and the new calls start immediately after the previous service has ended. Similar simulation scenarios can be applied to dynamic number of subscribers as well. This thesis assumes static number of terminals and multiple MBMS sessions during one simulation run in order to guarantee satisfactory number of samples to analyze the results.

6.3.2 MBMS Transmission and Reception

For each sector MTCH is created and it has a constant transmit power. MTCH's power is thus taken into account in interference calculations. MTCH is modeled similarly as *Common Pilot Channel (CPICH)*, i.e., only the transmitter, i.e., Node-B side is defined for that channel. Certain power allocation can also be reserved for MCCH. However, in these studies MCCH power is assumed to be part of the other common channels.

Reception of MTCH data can be done from a single or multiple cells/sectors to model soft and selective combining which were presented in Section 2.6. If the reception is done from multiple cells/sectors those cells/sectors are referred as combining set. Combining set update procedure for MBMS p-t-m transmissions is much like the ordinary soft handover, see Section 2.9.3. The difference to soft handover is that the UE does not send any measurement reports to the network side but independently decides, according to filtered pilot measurements, from which sectors it receives and combines signals. Due to the fact that combining set update procedure is very much like soft handover same terminology is used in this thesis when concerning issues related to the combining set update procedure. Parameterized delay is also modeled for combining set updates, i.e., combining set update events are held in the delay queue according to delay parameter before actual combining set update is done.

6.4 VoIP Modeling

Following subsections describe how VoIP specific issues presented in Section 5 are taken into account in the used simulators. The main focus of the modeling is on the radio network interface and related issues.

6.4.1 VoIP Traffic

The VoIP traffic model that is assumed in this thesis is illustrated in Figure 43. The main parameters of the feature are presented in Table 3 and described in the following:

- Duration of each call is random and distributed according to negative ex-

TABLE 3 General VoIP Assumptions

Parameter	Value(s)
VoIP packet size	38 bytes
VoIP packet inter-arrival time	20 ms
SID packet size	14 bytes
SID packet inter-arrival time	160 ms
Average activity period	3.0 s
Average silence period	3.0 s
Probability to start call in silence	0.5
Max. VoIP packet delay	80 ms
Call drop threshold	15 %
Call drop observation window length	10 s
Outage threshold	5 %
Outage observation window length	5 s

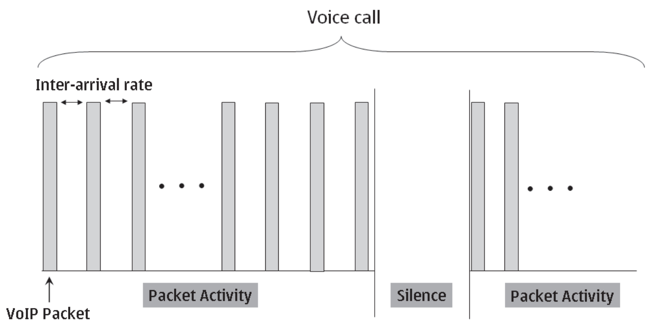

FIGURE 43 Example of VoIP Traffic Model

ponential distribution. The mean value of the distribution is given as a parameter.

- DTX is simulated through alternating, random duration activity and silence periods. The duration of these periods is distributed according to negative exponential distribution. Mean length of the activity/silence periods can be adjusted through the parameters.

- A probability to start the VoIP call in silence can be set through the parameters.

- During activity periods, a new VoIP packet is generated on constant intervals. Size of the packet (including payload and RTP, UDP, IP and other headers) is given in the parameters as well as the time interval between successive packets (e.g., 38 byte packet every 20 ms). ROCH is not modeled explicitly but assumed as ideal and thus taken into account in packet sizes.

For a telephone conversation there are silent intervals between speech frames, referred as silence in Figure 43. During silent intervals SID frames are sent in order to maintain the connection and to convey information needed to generate background (comfort) noise. They are sent in the beginning of the silent period and every 160 ms during the silence.

6.4.2 VoIP Calls

In order to be able to evaluate how many users can be served in a cell before QoS criteria is exceeded and users become unsatisfied, a static number of users per cell are assumed. By simulating different loads, i.e., amounts of users/cell and analyzing the QoS with each user amount the estimate of cell capacity (max. number of VoIP users / cell) can be interpolated. Users are placed uniformly around the simulation area.

6.4.3 Quality of Service Monitoring

Quality of service for VoIP calls is monitored through sliding window period. During that monitoring period, if at least a certain percentage of packets, given as a parameter, are **not** received correctly within a delay criteria, also given as a parameter, a flag to mark that call as unsuccessful is raised. At the end of each call the flag in question is checked and if it has been risen in any point of the call, the call is counted to be in outage. In addition, a VoIP call can be dropped in the dynamic environment if the quality of the transmission drops significantly. Dropping of VoIP calls is based on similar procedure defining the outage, only with different parameters. Finally, the overall outage percentage with the load in question is then calculated on the basis of total number of successful and unsuccessful (incl. dropped) calls. As mentioned earlier the total cell capacity to a level where, e.g., 5 % of all calls are in outage can be then interpolated from outage levels of different loads.

Regarding HSDPA, VoIP call QoS based on outage criteria is monitored in the MAC-hs after possible reordering of packets, see Figure 44 for reference. Packets can arrive out-of-order if, for instance, the initial transmission of the transport block does not succeed and thus it requires HARQ retransmission. The time how long UE waits for packets to arrive referred as T1 timer. The delay caused by T1 timer is thus taken into account in VoIP packet delay.

Regarding HSUPA, VoIP call QoS based on outage criteria is monitored in the MAC-es illustrated in Figure 45. The delay between Node-B and RNC is not explicitly modeled but it is taken into account in the delay budget.

Another way to define uplink VoIP capacity in uplink is through noise rise generated by the users in the system. According to that criteria system can support as many of VoIP users per cell where the average NR remains below certain level. Threshold of 6 dB is generally considered as a reasonable threshold for NR [31], also see Section 3.3.2 for more information.

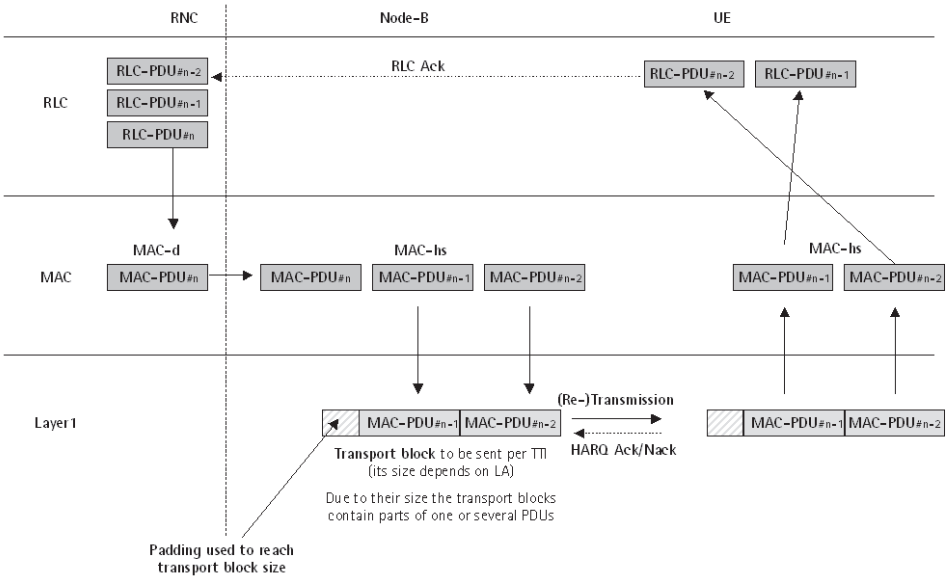

FIGURE 44 Data Flow for HSDPA, simplified from [8]

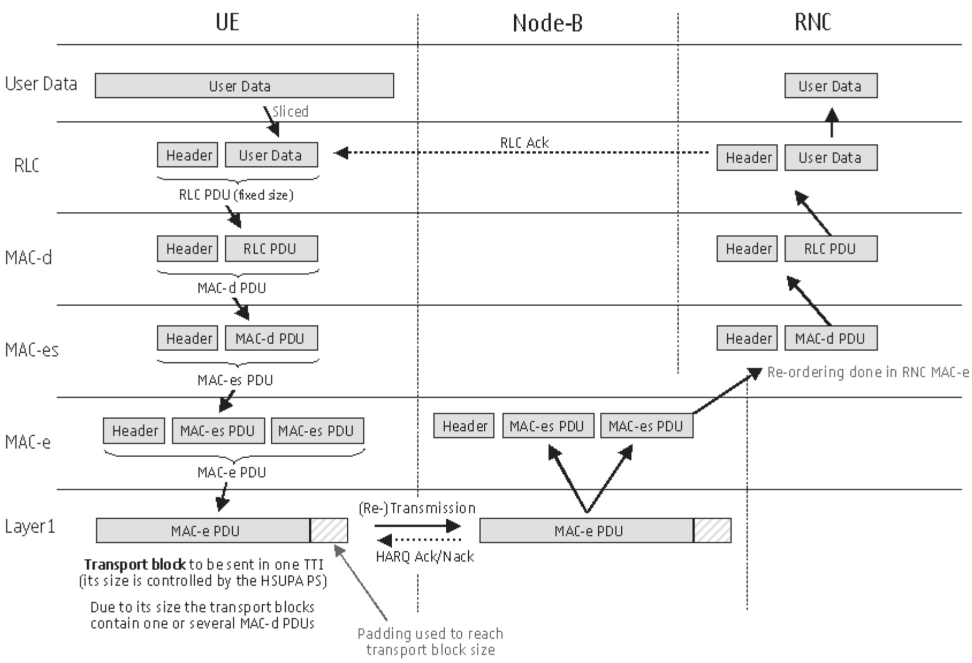

FIGURE 45 Data Flow for HSUPA, simplified from [4]

6.5 Interference Coordination Modeling

Interference coordination for HSUPA is implemented to quasi-static simulator as time domain coordination following the principles presented in Section 3.7. The base stations and scheduling functionalities are synchronized so that in certain cells only the cell edge users transmit while in the neighboring cells cell center period is in turn. For the purpose of the simulations, 9 HARQ processes are assumed as it allows a straightforward division of the HARQ processes into three groups, and application of a re-use pattern of 3 shown in Figure 25.

6.6 Dual Carrier Modeling

Dual carrier is modeled to quasi-static simulator as well in accordance to the concept presented in Section 3.8. Modeling includes, e.g., in addition to E-DCH DPCCH for all carriers to keep the power control and related actions up to date for all carries. Additional DPCCHs bring some overhead (interference) to the system which is thus taken into account. Rate requests, scheduling, retransmissions (always on the same carrier), etc. are also modeled in detail for the simulator.

7 ACHIEVED RESULTS

The objective of this chapter is to first present study scenarios and assumptions used with the research tools introduced in Chapter 6. Scenarios and assumptions are then followed by the result analysis.

7.1 Simulation Scenarios

The studies presented in this thesis are based on various different network layouts. Macro scenarios range from hexagonal cellular environment to a performance analysis in non-regular network layout. In addition to macro scenarios this thesis addresses the performance in manhattan scenario as well.

Majority of the simulative studies are conducted in hexagonal macro cellular scenarios. Most of the macro scenarios are utilizing *Wrap-Around (WA)*. The purpose of the wrap-around is to model the interference correctly also for outer cells. This is achieved by limiting the UE mobility / placement around the actual simulation area but replicating the cell transmissions around the whole simulation area to offer more realistic interference situation throughout the scenario. A UE is able to make a handover to the outer cells but if it moves outside the mobility area, its position is moved to the opposite side of the mobility area.

In this thesis two different WA layouts are applied: one with 21 cells and 7 base stations and another with tier type of a structure. In latter WA scenario, two tiers results into 21 cells and 7 BSs and three tiers into 57 cells and 19 BSs. Figure 46 depicts the cellular structure of the first wrap-around scenario and Figure 47 depicts the actual simulation area of the latter.

VoIP over HSDPA simulations presented in article **[PV]**, were initially based on scenario with 18 cells and 9 base stations without wrap-around. That scenario is illustrated in Figure 48. This thesis broadens that article slightly by providing comparison of those results into re-simulated results in a WA scenario to exclude, e.g., the impact of border-effects.

Figure 49 illustrates the non-regular macro scenario. Unlike with other sce-

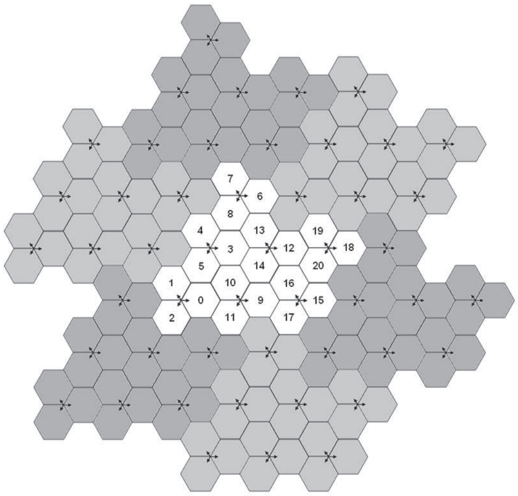

FIGURE 46 Simulation Scenario with 21 Cells and Wrap-around

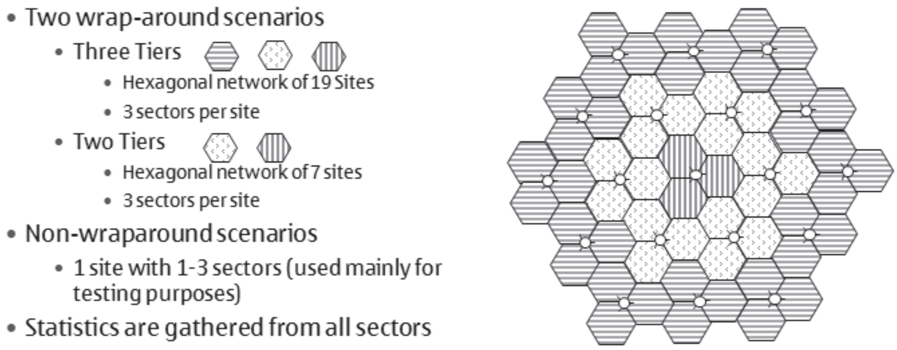

FIGURE 47 Simulation Scenario with 57 Cells

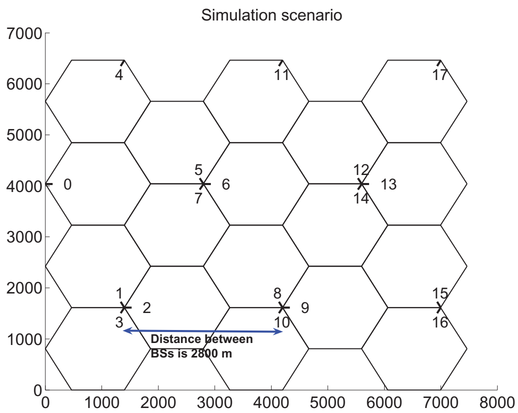

FIGURE 48 Simulation Scenario with 18 Cells and no Wrap-around

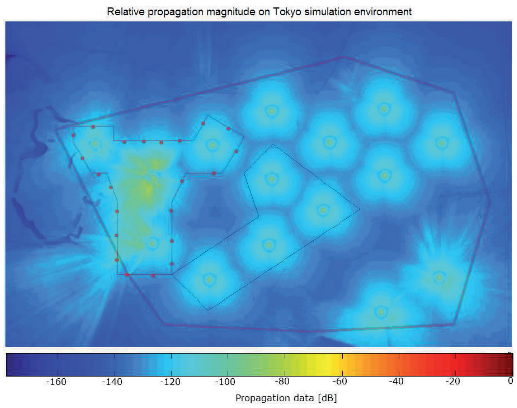

FIGURE 49 Non-regular Simulation Scenario

narios where propagation is defined to simulation parameters and functions for this scenario propagation data is acquired from network planning tool. The data includes, e.g., the effect of distance pathloss, large scale shadowing and antenna pattern. The used data is generated over Tokyo city downtown area to give more realistic network layout. The complete scenario consists of 15 base stations resulting into 45 cells with different propagation conditions. However, from that scenario certain sections are selected to study different aspects of non-regular scenario. The data is imported into system simulators via look-up tables.

Finally, complete manhattan scenario, presented on the right hand side of the Figure 50, consists of 132 buildings with dimensions of 200 (length) x 200 (width) x 100 m (hight). Node-Bs, 72 of them, are distributed in the streets between them. Each street is 25 m wide and UEs move along them. When UEs enter street crossings they either turn or continue straight. UEs do not enter the buildings. Manhattan scenario and related propagation model is based on UMTS 30.03 [21] and it provides so called 'urban canyon' -type of a scenario. In VoIP studies where, for instance, high amount of UEs increase the required amount of computational resources, the manhattan scenario is reduced to smaller 32 Node-Bs version. Nevertheless, with the same fundamental logic. See left hand side of Figure 50 illustrating the smaller manhattan scenario.

7.2 Simulation Assumptions

The main parameters for the simulations are presented in conjunction with the included articles. The used channel models and presented channel models in those parameters are based on tapped-delay line model in coherence with the ITU recommendations [60]. The model is characterized by the number of taps, the time delay relative to the first tap, the average power relative to the strongest tap, and the Doppler spectrum of each tap. The power profiles are modified from

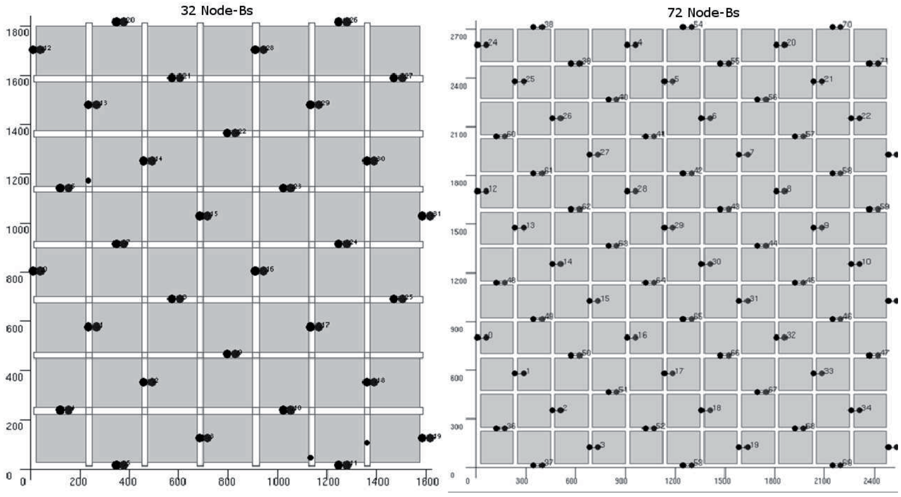

FIGURE 50 Manhattan Simulation Scenarios, [21]

TABLE 4 Vehicular A Channel, [60]

Tap	Relative delay [ns]	Avg. power [dB]
1	0	0.0
2	310	-1.0
3	710	-9.0
4	1090	-10.0
5	1730	-15.0
6	2510	-20.0

the original ITU power delay profiles so that the delay and powers between paths are normalized to at least one chip-time, [21]. For instance, Table 4 shows the tap delays and powers for VehA channel in accordance of ITU recommendations. Those values are modified so that powers are converted into linear scale and linear sum of the powers is calculated. Linear powers are then divided with the sum, i.e., normalized. Finally tap powers are converted back to dBs. Tap delays in chips are chosen so that they are as close as possible to the specified values in nanoseconds. For example, the delay of the tap 1 in VehA channel is one chip as one chip corresponds to 260 ns. Modified VehA channel is shown Table 5.

7.3 Simulation Results

In this section the achieved results of this thesis are presented and analyzed. First diversity techniques and power saving opportunities are covered through MBMS

TABLE 5 Modified Vehicular A Channel

Tap	Relative delay [chips]	Avg. power [dB]
1	0	-3.1
2	1	-4.1
3	3	-12.1
4	4	-13.1
5	7	-18.1
6	10	-23.1

over WCDMA. Those studies are then followed by mobility performance evaluation and additional power saving studies by means of CPC. VoIP over HSPA both in uplink and downlink is used as a reference service in those studies. Finally, HSUPA interference coordination and dual carrier HSUPA is considered before concluding the section with HSDPA and LTE comparison study.

7.3.1 WCDMA Performance

In this section WCDMA performance is covered with MBMS through three main aspects: macro diversity combining, receive and transmit diversity. Receive diversity studies are, in addition, extended by analyzing the possibility to switch off additional receiver branches when channel conditions allow. That kind of operation could provide possibilities for battery savings.

7.3.1.1 Macro Diversity

Macro diversity and how it impacts the MBMS p-t-m performance in WCDMA networks is presented in included articles **[PI]** and **[PIII]**. In addition, article **[PII]** addresses the performance of macro diversity combining together with receive diversity but it will be discussed in later sections.

Studies of macro diversity combining indicate that it can improve the network performance significantly. By adding merely one link into the combining set roughly 5 dB gain in required E_c/I_{or} to reach the coverage with soft combining and 3 dB gain with selective combining (in Vehicular A channel) can be acquired when compared to a situation without combining. Higher gain of soft combining over selective combining is due to the fact that soft combining can benefit also from poor links whereas selective combining benefits only from good links. Gain acquired from macro diversity in general is found to be dependent on the amount of links in the combining set which can be controlled in some sense by adjusting combining thresholds.

7.3.1.2 Receive Diversity and Battery Savings

Performance enhancement studies in WCDMA networks are extended in articles **[PII]** and **[PIII]** to cover the situation where (MBMS) terminals would be deployed with two receive antennas instead of just one. Traditionally multiple antennas have not been employed in smallest handheld devices due to size and cost constraints. However, receive diversity is one of the most efficient diversity techniques, since in addition to the diversity gain the received combined signal power is theoretically doubled with two receive antennas when compared to single antenna reception.

Simulation results show that receive diversity can bring substantial benefit for MBMS transmissions. The gain of 2 Rx receive diversity over 1 Rx is roughly 4 dB in terms of in required E_c/I_{or} depending on the other simulation parameters. Moreover, the results show that even if terminals would be able to utilize receive diversity macro diversity combining still brings quite a significant gain to the performance.

When network is dimensioned for terminals with only one receive antenna (no Rx diversity) and the system contains also 2 Rx terminals the power savings opportunities can in theory be accomplished by switching off additional receiver branches when radio conditions allow. Article **[PIII]** studies possible power saving opportunities in such situations.

The study demonstrates that it is indeed possible to achieve very similar coverage levels in situations where 2 Rx UEs are either allowed to reconfigure between 1 Rx Rake and 2 Rx Rake receiver or to use only 2 Rx Rake receiver. To achieve this goal in terms of coverage levels one only needs to have suitable switching thresholds and C/I filtering windows in use. Apart from network performance, antenna usage statistics indicate that there is a possibility that the 1 Rx Rake configuration is used for a significant amount of time. Thus, there are definitely power saving opportunities relative to UEs using 2 Rx Rake continuously.

7.3.1.3 Transmit Diversity

Finally, transmit diversity is also considered as one way to improve the system capacity and performance. Tx diversity exploits the principle of providing the receiver with multiple faded replicas of the same information-bearing signal. The impact of Tx diversity together with MBMS is studied in article **[PIII]**.

The results of that study show that transmit diversity narrows E_s/N_o distribution and provides slight improvement to lower tail of E_s/N_o distribution with and without soft combining. In terms of coverage this slight improvement leads to around 2 dB gain without combining. The gain is reduced to less than 1.5 dB with soft combining due to the presence of additional source for diversity.

7.3.2 HSDPA Performance

In this section downlink performance over high speed packet access networks is studied in terms of mobility and battery savings with VoIP service. Mobility related aspects are covered in articles **[PV]**, **[PVI]** and **[PV]**. Battery savings are addressed through discontinuous reception in article **[PVII]**. In addition, in article **[PXI]** VoIP over HSDPA performance is addressed in comparison to LTE downlink performance but that article is discussed later in Section 7.3.4.

7.3.2.1 Mobility

The fact that HSDPA is supporting only hard handover can result into excessive packet losses and delays if handovers are not executed in time or contrary are executed too often. The impact of serving HS-DSCH cell change procedure along with different user velocities and cell change execution delays was addressed in article **[PV]**. The probability of signaling radio bearer reception is further investigated in **[PVI]** by means of dynamic simulations in macro and more demanding manhattan scenarios. Finally, **[PVIII]** deepens the mobility studies through restricting the data transmission during the handover procedure and limiting the maximum number of terminals per cell.

The simulation results in article **[PV]** indicate, in general, moderate sensitivity of VoIP capacity to the UE velocity. Very high velocities lead to more noticeable capacity loss. However, fine-tuning handover parameters for different velocities can provide better performance but on the other hand in practice that would require some mechanisms for UE speed detection if it would be done dynamically. In terms of handover delay it starts to have more significant impact only at UE velocity of 50 km/h or more. Reliability of handover signaling, i.e., situation where handover command is actually sent over the air is concluded not to pose problems.

A while after the article **[PV]** was published VoIP over HSDPA studies were re-opened for study, e.g., power consumption related matters. As quite a considerable amount of time had passed from the studies presented in article **[PV]** the simulator had also undergone some updates. The updated version of the simulator includes more detailed modeling of mobility, HSDPA and VoIP to name a few. In addition, support for wrap-around scenario was provided along with the updated version. Due to these changes, performance benchmarking between old and revised results was made.

Comparison between updated results and the results presented in article **[PV]** is illustrated in Figure 51. As the figure shows the absolute capacity numbers are somewhat improved with the updated simulator version. The increase depends heavily on the user velocity which is mainly due to revised handover and mobility operation in the updated simulator version. For instance, with 3 km/h the capacity numbers are still in same order and the difference is only a couple of UEs/cell but for higher velocities the capacity is higher than before. Therefore, also the relative drop caused by user velocity is somewhat lower than

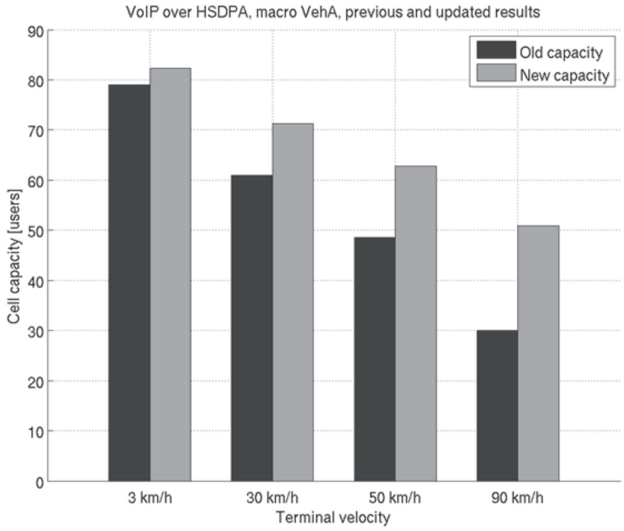

FIGURE 51 VoIP over HSDPA 95 % Outage Level Comparison with Different Velocities

before. However, the overall conclusion of article **[PV]** about UE velocity having moderate sensitivity to VoIP capacity still applies. The impact of delay between the two simulator versions is illustrated in Figure 52. As the figure shows, again the low velocities are practically unchanged but with higher velocities the absolute and also to some extent relative numbers are changed. However, the original conclusion stating that when higher velocities are in question additional delays start to make much more significant impact to the performance sill applies.

In article **[PVI]** the serving HS-DSCH change procedure is evaluated in urban canyon type of manhattan scenario in addition to macro scenario. Special emphasis in article **[PVI]** is on the SRB reception probability. If SRBs indicating cell change are not received correctly within reasonable time it can lead to serious (VoIP) service degradation due to prolonged camping on a poor cell. In the worst case outcome can be even dropping the call, if the cell change command is not received before VoIP call drop criterion is met.

The achieved results confirm the preliminary studies that a relatively low SRB/handover error rate is experienced in a macro cell scenario. However, in a more challenging manhattan scenario the error rate is higher. Study also indicates that failure rates in manhattan scenario can be mitigated by reducing the delays and making the handover procedure faster or by transmitting the serving cell change command from the target cell HS-SCCH rather than from serving HS-DSCH.

Finally, article **[PVIII]** studies how the performance of VoIP is affected if data transmission to a UE is stopped for specific amount of time during the handover procedure. It is expected that during the handover procedure higher packet loss is experienced anyway and thus resources could be freed for other users. The study also addresses situation where the number of UEs per cell is

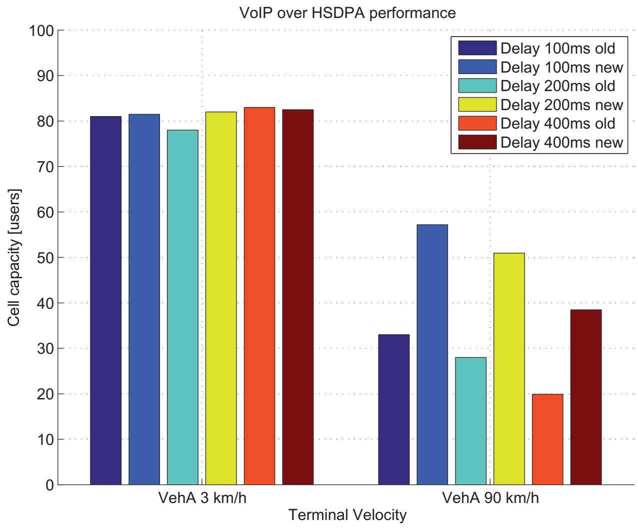

FIGURE 52 VoIP over HSDPA 95 % Outage Level Comparison with Different Delays

limited, e.g., due to hardware constraints or RRM settings.

The article shows that restricting the transmission during handovers (non-Tx periods) does not have an impact when UEs are moving with low velocities. However, in the case of higher UE velocity non-Tx periods can improve the total system capacity slightly as the resources become available for other UEs. Though, restricted periods should not be too long as then the performance of UEs doing HOs and thus the capacity can be compromised. With the respect of limiting the maximum number of UEs per cell it is shown that if the limit is too close to the capacity level (seen without the limiting) the performance can deteriorate significantly. Assuming that proper limit is able to be set and it is not dictated by, e.g., hardware limitations, it can even improve the performance by balancing the load between the cells.

7.3.2.2 Battery Savings

In article **[PVII]** discontinuous reception (DRX) cycles and related timers are studied together with VoIP. As described earlier in this thesis, DRX cycles limit the scheduling freedom of users while increasing the battery saving opportunities in the UE by allowing it to turn its receiver circuitry off for some periods of time. Studies on techniques such as DRX are critical as the battery life of small handheld devices might become a limiting factor in providing satisfactory user experience.

The article indicates that longer DRX cycles can provide quite high battery saving opportunities but at the same time VoIP over HSDPA capacity can be compromised. Capacity deterioration is, however, possible to be mitigated by using an adequately long inactivity timer, i.e., for how long UE keeps its receive antenna on after a successful reception of HS-SCCH. Even though longer inactivity timer limits the power saving opportunities to some extend those are still quite significant.

7.3.3 HSUPA Performance

Uplink is most often considered to limit the overall coverage of services due to the fact that uplink is characterized by power limited terminals and high interference levels. Moreover, with certain service types, such as speech, the high number of simultaneous connections can put high strain on the radio network resources. Similarly to HSDPA, also mobility brings along its own challenges even though SHO is supported in HSUPA. In the following subsection these challenges are addressed. Firstly, mobility related aspects are addressed in dynamic system level study presented in article **[PIV]**. That is followed by article **[PXIII]** covering studies on coverage extension and battery saving possibilities in the uplink. Articles **[PXII]** and **[PIX]** on the other hand cover interference coordination and dual carrier uplink. Studies for articles **[PXIII]**, **[PXII]** and **[PIX]** are conducted with a quasi-static system level tool whereas studies for article **[PIV]** are done with a fully dynamic system level tool.

7.3.3.1 Mobility

The study presented in article **[PIV]** covers various mobility related issues from the perspective of 10 ms TTI HSUPA with VoIP service. Various handover delays, active set sizes and user velocities are covered. Additionally, the impact of SID frames is studied.

Achieved results indicate that the system performance can be highly affected by user's mobility. Handover delay shows with low user velocities the performance loss within 10 % providing that the delay stays within reasonable limits. With higher velocities the sensitivity to the delay increases. With respect of the SHO performance it is shown in the article that the difference between a single link and a two-way soft handover performance is significant, following the same trends as macro diversity studies shown earlier in this thesis indicate. However, the VoIP capacity is only slightly affected if comparison between two-way and three-way SHO is made.

When comparing the achieved results to the HSDPA ones it seems that HSUPA is not limiting the capacity with the studied assumptions and scenarios. SHO proved to secure the HSUPA (VoIP) performance to the extend that mobility or power limitation of terminals did not pose severe problems. The faster changing channel will still pose some issues that should be considered, e.g., in network planning.

7.3.3.2 Coverage and Battery Savings

As described earlier HSUPA defines two TTI lengths: 2 and 10 ms. The latter is mandatory and two milliseconds is optional for all terminals. Those two TTI lengths have different properties which basically define the purpose to which they should be used. While 2 ms provides longer periods of silence it requires higher instantaneous Tx power. With 10 ms lower Tx power is required but the transmission period is longer thus reducing the amount of possible retransmissions and battery saving opportunities with UL DTX. Article **[PXIII]** addresses these issues by means of quasi-static simulations from the perspective of VoIP over HSUPA capacity and achievable battery saving opportunities.

The article **[PXIII]** indicates firstly that utilizing UL DTX is highly beneficial when compared to continuous transmission of DPCCH. Continuous transmission, especially by large number of UEs, generates high additional interference and thus reduces the achievable capacity numbers when compared to UL DTX. Moreover, results show that utilizing a mixture of both transmission time intervals can lead to avoiding potential coverage problems whilst providing enhanced battery saving opportunities. Enhanced battery savings are achieved due to part of the UEs being able to utilize the shorter 2 ms TTI which has higher UL DTX usage whereas the other can utilize 10 ms TTI providing a better coverage. The most significant improvement of utilizing both TTI lengths over just single TTI is seen especially in cells with large *Inter-site Distance (ISD)*. This is caused by the fact that 2 ms consumes more power resources from already power limited (cell edge) terminals in a scenario with large ISD.

7.3.3.3 Interference Coordination in HSUPA Networks

Section 3.7 presented the motivation and principles of interference coordination in HSUPA networks. Roughly speaking IC alternates throughout the network the priorities when users can and cannot transmit to achieve reduced interference levels and higher performance. Article **[PXII]** evaluates the IC performance against 'traditional' network performance in high load situation. Moreover, a large variety of combinations including, e.g., different schedulers, cell center / edge user amounts and RoT targets are investigated through a quasi-static approach.

The results presented in article **[PXII]** indicate that IC can in fact bring gain in certain scenarios. In general the highest gains are visible when network is heavily loaded. Though, the gain comes at the expense of fairness leading to users that already have higher throughput benefiting from the situation. However, that kind of reduction in fairness and increase in throughput could also be achieved through adjusting (FF) scheduling parameters and not as such justify the network wide IC. With low loads IC can even lead to performance loss by limiting the available transmission periods. Moreover, it is shown that the major part of the gain comes from splitting users to different groups and scheduling these groups without any network wide interference coordination. However, this study did not take into account UL DTX which potentially could have reduced the interference levels even more resulting into a higher performance together with IC. Additionally, VoIP could be one potential service to study together with IC and UL DTX to achieve higher capacity through potentially lower interference conditions.

7.3.3.4 Motivation for Dual Carrier HSUPA

As described earlier in this thesis, dual carrier capability for HSUPA was introduced to 3GPP Release 9. The results presented in article **[PIX]** were used to support that standardization work. Article presents quasi-static simulation results from situations where DC can enhance HSUPA performance in comparison to single carrier systems.

The results of the article **[PIX]** first indicates through single carrier evaluations that, especially in small cells, terminals have spare power available for dual carrier operation. These observations are verified with actual dual carrier simulations which indicate that the burst throughput can be roughly doubled with low loads. Results with DPCCH gating, i.e., UL DTX indicate that DC is able to achieve good performance also with higher loads which on another hand means that the control channel overhead is getting higher in DC systems than in two single carrier systems. In the larger cells only the users in good situation can fully benefit from using dual carrier, as expected.

7.3.4 HSDPA Versus LTE Performance

Finally in this thesis, a glance to the next generation cellular networks is given through benchmarking HSDPA and LTE downlink performance. Benchmarking studies are conducted with VoIP and *Constant Bit Rate (CBR)* type of services to give in depth overview on matter. Moreover, realistic non-regular network layout presented in Section 7.1 is imported to the dynamic system simulators to achieve even more realistic performance comparison. VoIP benchmarking study is presented in article **[PXI]** and CBR in article **[PX]**.

In article **[PXI]** VoIP performance over Rel'5 HSDPA and Rel'8 LTE is studied in regular and non-regular scenario layouts selected from the complete scenario. The results indicate that VoIP over LTE has significantly higher performance in terms of capacity than VoIP over HSDPA regardless of the user velocity or the cellular layout. Moreover, it is shown that both systems have similar performance in regular scenario layout as they do in macro environment. This further confirms that macro cellular evaluations presented also in this thesis are well in line with scenarios having more realistic propagation conditions.

Simulation result from non-regular scenario, however, indicate that there can be problems due to some of the strongest cells becoming overloaded. Though, several possible solutions for this are presented, such as fine-tuning the cell-specific handover parameters, advanced self optimizing network algorithms or even (simple) admission control algorithms. The result confirm those presented ideas through a very simple admission control evaluation that is able to remedy the problems quite easily.

Even though, the LTE showed quite high gains over HSDPA one needs to keep in mind that it was Rel'5 HSDPA that was benchmarked against LTE in that article. With later releases several enhancements are presented also to HSDPA. As the results in this thesis have indicated, for instance, receive diversity capability and fast serving cell change (SRB received from target cell) for HSDPA could narrow the gap between the techniques.

Article **[PX]** presents the results for benchmarking of the LTE/HSDPA systems with CBR type of service. That article further deepens the analysis compared to VoIP benchmarking by taking into account higher order modulation and MIMO systems also for HSDPA. In other words Rel'8 HSDPA and Rel'8 LTE downlink are benchmarked.

The results show that LTE outperforms HSDPA in terms of spectral efficiency and user throughput. Difference between the systems is emphasized once the load of the system increases. With highest simulated UE amount, 8 UEs per cell and LTE MIMO, about 90 % of UEs are fully satisfied, while only about 45 % of the UEs are fully satisfied with HSDPA MIMO. Users are considered to be satisfied if the offered bit rate of 512 is achieved. That load results into roughly 30 % gain in spectral efficiency even though with LTE the network is running half empty due to uneven UE distribution in non-regular network with variable cell sizes.

8 CONCLUSION

The purpose of this thesis was to address variety of performance related aspects in terms of 3G/3.5G networks. MBMS and VoIP were used in most of the studies to benchmark different features as they are undeniably key concepts in wireless cellular networks and thus securing their performance can be considered as critical. From practical perspective, wireless network sets very strict environment for different services through different delays, packet loss, jitter, corruption and so forth. This thesis addresses those and related challenges through evaluating different diversity options, mobility performance aspects and robustness, battery saving opportunities, interference coordination and dual carrier uplink. Studies were conducted with the help of time driven system simulators where, e.g., fading, propagation and RRM functionalities are explicitly taken into account.

This thesis shows that there is significant benefits from macro diversity combining and the magnitude of the gain is dependent on the adjustable threshold parameters adjusting the amount of combinable sources. Macro diversity, i.e., combining transmissions from multiple sources is beneficial both in the downlink and uplink. In terms of downlink, macro diversity is likely option for services that are expected to be available through large parts of the network, for instance, services like MBMS. In the uplink one form of macro diversity, soft handover, is part of the system specifications and thus it can and should be utilized to avoid potential performance demonstrated in this thesis. In addition to macro diversity, also receive diversity is found to be very efficient method to improve the performance.

Besides evaluating different diversity techniques, this thesis evaluates in detail the impact of mobility and cell change procedure. For instance, if SRBs indicating cell change are not received correctly within reasonable time it can lead to service degradation due to prolonged camping on a poor cell. In the worst case the outcome can be even dropping the call. The results show that in the uplink mobility and handovers do not pose as significant impact due to the existence of soft handover. However, in the downlink the handover procedure can be compromised in more challenging urban canyon scenario or with high UE velocities. Though, failure rates can be mitigated by reducing the delays and making the handover procedure faster or alternatively transmitting the serving cell change command from the target cell HS-SCCH rather than from serving HS-DSCH. All in all, potential problems in terms of mobility should be evaluated and addressed in detail, e.g., in the network planning to guarantee optimal performance.

Interference coordination is also addressed in this thesis as potential performance enhancement. However, the results of the studied scenarios show that even though IC can bring gain in certain scenarios, the majority of the gain comes at the expense of fairness leading to users that already have higher throughput benefiting from the situation. Moreover, it is shown that gain can be achieved without any network wide interference coordination and thus as such the IC is not that attractive option. Though, by taking into account also the UL DTX and some further considerations IC could be reconsidered in feature studies.

Dual carrier was standardized for HSDPA in Rel'8. This thesis evaluates the motivation and applicability of having the dual carrier also in the high speed uplink. The results show that in especially in small cells, terminals have spare power available for dual carrier operation. The actual dual carrier simulations verify that the burst throughput can be roughly doubled with low loads. When UL DTX is taken into account as well DC-HSUPA is able to benefit also with higher loads.

Apart from addressing the performance from radio network point of view this thesis evaluates the possible battery saving opportunities through continuous packet connectivity features and switching off additional antennas. The results show that in the downlink discontinuous reception cycles can provide noticeable battery saving opportunities and by using optimized timer settings capacity is not compromised. From the uplink perspective discontinuous transmissions are equally highly beneficial providing battery saving opportunities and at the same time increasing the uplink capacity due to decreased interference. Achieved battery saving gains are higher with 2 ms TTI than with 10 ms TTI but the coverage can be compromised in some situations when utilizing only 2 ms TTI. Thus, the mixture of both available TTIs in the uplink can extend the coverage whilst providing optimized battery saving opportunities. In addition, battery saving opportunities could be achieved in devices having multiple receive antennas by switching the other antenna off when channel conditions allow it. Utilizing the switching off the additional antennas is probable actual networks as those are likely to be dimensioned to terminals with one receive antenna (1 Rx) due to the fact that 2 Rx terminals are expected to emerge slowly.

Finally, this thesis provides a glance to the feature by comparing HSDPA and LTE downlink. Comparison is done through VoIP and constant bit rate type of services which utilize different features of the systems. The results show that LTE outperforms HSDPA in terms of VoIP capacity, spectral efficiency and user throughput. Difference between the systems is emphasized once the load of the system increases. However, similarly to LTE HSPA is being constantly developed and new features are being standardized to upcoming Releases. Thus, prolonging the lifespan of HSPA systems and narrowing the gap to LTE performance.

YHTEENVETO (FINNISH SUMMARY)

Ensimmäiset kolmannen sukupolven (3G) matkaviestinverkot otettiin käyttöön vuonna 2002 ja ne pohjautuvat pääasiassa WCDMA-tekniikkaan (engl. Wideband Code Division Multiple Access). Siinä missä toisen sukupolven (2G) verkot tarjosivat pääasiassa mahdollisuuden puhe- ja tekstiviestiliikenteelle, 3G-verkot mahdollistivat ensimmäistä kertaa suurempia tiedonsiirtonopeuksia vaativien pakettipohjaisten palveluiden tarjoamisen matkaviestinverkoissa. Mahdollisten palveluiden kirjo laajeni entisestään, kun kolmannen sukupolven verkkoihin standardisoitiin laajennukset, jotka mahdollistavat entistä suuremmat tiedonsiirtonopeudet sekä pienemmät vasteajat. Nämä laajennukset standardisoitiin kahdessa osassa, ensin alalinkin ns. HSDPA (engl. High Speed Downlink Packet Access) evoluutio ja hieman myöhemmin ylälinkin vastaava evoluutio ns. HSUPA (engl. High Speed Uplink Packet Access). Jos viitataan yhteisesti molempiin evoluutioihin, käytetään nimeä HSPA (engl. High Speed Packet Access) tai 3.5G. Odotetusti WCDMA- ja HSPA-verkkojen suosio on kasvanut lähes räjähdysmäisesti, sillä vuoden 2004 lopussa WCDMA-tilaajia oli vain 17 miljoonaa, mutta nyt tilaajia on jo yli 600 miljoonaa maailmanlaajuisesti. Jos mukaan lasketaan myös toisen sukupolven GSM-tekniikka, tilaajien määrä hipoo jo 4.5 miljardia. Tulevaisuudessa määrän uskotaan kasvavan entisestään, sillä kasvua vauhdittavat alati laajeneva kirjo innovatiivisia palveluita sekä kehittyneempiä laitteita.

Tässä väitöskirjatyössä tutkitaan 3G/3.5G-verkkojen suorituskykyä, mahdollisia parannuksia sekä ongelmakohtia. Useimmissa tapauksissa eri tutkimusosa-alueet käydään läpi pakettiliikenteeseen perustuvan puheliikenteen (Voice over IP) tai ryhmälähetyksiä käsittelevän MBMS-tekniikan (engl. Multimedia Broadcast Multicast Service) suorituskyvyn kautta, sillä langattomien verkkojen puolella nämä tekniikat tulevat eittämättä olemaan olennaisimpien joukossa. Kyseiset palvelut tarjoavat myös uusia näkökulmia verrattuna kiinteisiin verkkoihin: esimerkiksi VoIP-puheluiden ja sen mahdollistamien lisäpalveluiden tuominen osaksi matkaviestinverkkoja tarjoaa operaattoreille mahdollisuuden uudenlaisiin tulovirtoihin sekä mittaviin kustannussäästöihin siinä valossa, ettei tällöin enää tarvittaisi erillistä piirikytkentäistä verkkoa puheelle, kuten nykyään. Kuitenkin samalla langaton matkaviestinverkko asettaa näille ja muille palveluille haasteelliset toimintaolosuhteet mm. eri viiveiden, virheiden sekä radioresurssien hallinnan myötä. Tämä väitöskirjatyö pyrkii vastaamaan näihin haasteisiin tutkimalla eri diversiteettivaihtoehtoja, liikkuvuuden vikasietoisuutta sekä vaikutusta suorituskykykyyn, tehonsäästömahdollisuuksia, häiriön koordinointia sekä taajuuskaistojen lisäämistä ylälinkkiin. Tutkimusmenetelmänä käytetään ohjelmistopohjaisia radioverkkosimulaattoreita, joilla pyritään mallintamaan reaalimaailman matkaviestinjärjestelmää hyvin tarkasti. Simulaattoreissa käytetyt mallit mm. etenemiselle, häipymälle sekä radioresurssien hallinta-algoritmeille ovat yleisesti hyväksyttyjä ja tunnettuja.

Väitöskirjatyön tutkimukset makro-, lähetys- ja vastaanotindiversiteetin osalta käsitellään MBMS-tekniikan kautta. Kyseiset tekniikat ovat keskeisenä osana tutkimusta liikkuvuuteen liittyvien parametrien optimoinnin lisäksi. Makrodiversiteetti tarkoittaa yksinkertaistettuna sitä, että eri tukiasemista tulevat lähetykset yhdistetään päätelaitteessa, jotta signaalin laatua voidaan merkittävästi parantaa. Lähetys- ja vastaanotindiversiteetit taas tarkoittavat tilannetta, jossa lähetys- ja/tai vastaanottopäähän lisätään antenneja parantamaan suorituskykyä. MBMS:n osalta keskitytään pääasiassa ensimmäisten 3G-laitteiden näkökulmaan, ja täten HSPA-evoluutioiden tuomia vaikutuksia ei erikseen käsitellä.

Liikkuvuuden hallintaa ja toimintaa yhteisvastuunvaihtojen (engl. handover) yhteydessä käsitellään tässä väitöskirjatyössä erityisesti VoIP-tekniikan kautta. Erityistä huomiota vaativaa on VoIP-liikenteen käyttäytyminen siirryttäessä tukiaseman/-asemien katealueelta toiselle eli yhteisvastuuvaihtojen yhteydessä, sillä VoIP on hyvin kriittinen kaikille ylimääräisille viiveille sen reaaliaikaisuuden vuoksi. Liikkuvuuden osalta käsitellään käyttäjien nopeuden, yhteisvastuunvaihtoviideiden, yhteisvastuunvaihtokomentojen luotettavuuden ja yhteisvastuunvaihtotapojen vaikutusta.

Radioverkon toiminnan ja palvelunlaadun takaamisen lisäksi yhdeksi merkittäväksi osa-alueeksi hyvän käyttäjäkokemuksien osalta voi muodostua myös tehon- ja täten päätelaitteen akunkulutus. Väitöskirjatyössä otetaan kantaa tähän asiaan tutkimalla, millaiset tehonsäästömahdollisuudet vastaanoton ja lähetyksien optimointi HSPA-verkoissa toisi tullessaan. Vastaanottopuolella tutkitaan, kuinka päätelaite voisi pitää vastaanottopiiristöään päällä vain murto-osan ajasta sopimalla verkon kanssa tietyt syklit, milloin sille voi lähettää dataa ja milloin ei. Ylälinkin puolella taas tutkitaan, kuinka kontrolli-lähetyksiä voidaan rajoittaa niin, että saavutetaan vastaanvanlaiset tehonsäästömahdollisuudet. Molempien tekniikoiden osalta on erityisen tärkeää huomioida, että tehonsäästöt eivät saa vaarantaa palvelun laatua saatika verkon toimintaa.

Häiriönkoordinointi ylälinkissä on yksi teoreettinen mahdollisuus parantaa solun laidalla olevien käyttäjien suorituskykyä. Monesti nämä käyttäjät ovat tehorajoitteisia ja ylimääräinen häiriö vain huonontaa niiden tilannetta. Väistökirjatyössä tutkitaan, kuinka aikajakoinen koko verkon laajuinen lähetyksien koordinointi voisi parantaa tätä tilannetta. Lähetykset pyritään koordinoimaan siten, että solun laidalla olevien käyttäjien lähettäessä dataa niitä ympäröivissä soluissa vain keskellä solua olevat käyttäjät saisivat lähettää.

Toinen mahdollisuus osaltaan vähentää syntyvää häiriötä, mutta samalla lisätä hyvässä tilanteessa olevien käyttäjien maksimitiedonsiirtonopeuksia, on lisätä käytössä olevia taajuuskaistoja. HSPA:n alalinkissä tämä on jo tehty aiemmin, mutta väitöskirjatyössä tutkitaan mahdollisuutta soveltaa ideaa myös ylälinkkiin. Alalinkistä poiketen ylälinkissä on huomioitava päätelaitteiden huomattavasti rajoittuneempi teho, tehonsäätö sekä kontrollikanavien tuoma häiriö.

Lopuksi väitöskirjatyössä verrataan kolmannen sukupolven verkkojen toimintaan seuraavan sukupolven LTE-verkkoihin (engl. Long Term Evolution). Vertailu suoritetaan skenaariossa, jossa radioverkkosuunnitelutyökalulla on luotu oikean kaupungin päälle realistinen radiotien etenemismalli/-data. Vertailussa otetaan kantaa niin pakettipohjaisen puheliikenteen kuin ladattavan datan osalta millaiseen suorituskykyyn eri verkot pystyvät.

APPENDIX 1 RELIABILITY ANALYSIS

System simulators that were used as research tools in this thesis were presented briefly in Section 6 and, for instance, in [49], [63] and [69] one of the tools is verified and presented in very detailed level. This appendix is aimed to give the statistical confidence on simulation results presented in this thesis. Confidence analysis for the dynamic simulator will be conducted through VoIP and MBMS examples. Full buffer test case will be used to demonstrate the reliability of the used quasi-static simulator.

Statistical analysis is based on evaluating the performance with different seeds in selected test cases. Simulation seed is a parameter to a random number generator. Based on the seed a random, e.g., starting position and direction of movement is generated for each UE. In addition to this there are several other random number generators used in the simulations which will change if seed is changed. However, even though all of the simulation results depend on random processes, the results are reproducible with certain level of accuracy which is defined in this appendix.

An interval estimation can be used to define a confidence interval, which means that the sample, ϕ, is within a defined interval with a certain probability, i.e.,

$$P(a \leq \phi \leq b) = 1 - \alpha \tag{5}$$

where the interval $[a, b]$ is a $(1-\alpha) \times 100\%$ confidence interval of ϕ. A probability that the ϕ is not within the interval is α. When the number of samples n is ≤ 30 the standardized normal distribution, $N(0, 1)$, can be used to define confidential interval, which is

$$(\overline{x} - z_{\alpha/2} \times s/\sqrt{n}, \overline{x} + z_{\alpha/2} \times s/\sqrt{n}). \tag{6}$$

In Eq. 6 the $\overline{x}$ is the average value, $z_{\alpha/2}$ is the critical value taken from the standardized normal distribution $N(0, 1)$, s is the standard deviation and n is the number of samples, i.e., in this case the number of simulation runs.

APPENDIX 1.1 Confidence Analysis on Dynamic Simulations

Confidence analysis on dynamic system simulator and the results that it produces is two-fold in this section. Firstly, MBMS results are verified through process described above. MBMS verification is followed by equivalent VoIP analysis.

APPENDIX 1.1.1 MBMS Result Verification

Statistical confidence of MBMS simulations is analyzed by using four test cases with various simulation seeds. Analysis from these cases is focused on percentage of satisfied users of each test case as well as to the 95 % coverage levels which are acquired by interpolating these test cases. Main simulation parameters for these test cases are presented in Table 6.

TABLE 6 Simulation Parameters for MBMS Reliability Analysis

Simulation parameter	Value(s)
Scenario	Wrap-around, 21 hexagonal cells
Cell radius	933 m
UE velocity	3 km/h
Combining scheme	Soft combining
Receiver	Rake 1Rx
Pathloss model	Modified Okumura-Hata
Channel models	Modified ITU Vehicular A
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
MTCH bit rate	128 kbps
TTI length	40 ms
Combining set size	3
Add window	4 dB
Add timer	100 ms
Drop window	6 dB
Drop timer	640 ms
Replace window	2 dB
Replace timer	100 ms
Combining set update delay	0 TTIs
MBMS session length	20 s
Number of sessions	18
Simulation time	360 s
Number of UEs per session	600
Network synchronization	Ideal

Tables 7, 8 and 9 present the results acquired from dynamic MBMS simulation runs with $n = 31$ different random generator seeds. On basis of those simulation results confidence intervals of 90 %, 95 % and 99 % for are presented in Table 10, 11 and 12. As those tables show abnormalities even with 99 % confidence interval are very minor in every case. Thus, it can be concluded that MBMS results presented in this thesis are reliable and reproducible with high level of accuracy.

APPENDIX 1.1.2 VoIP Result Verification

The purpose of this section is to provide statistical confidence analysis for (dynamic) Voice over IP simulations presented this thesis. The scenario for confidence analysis is 21 cell wrap-around macro cell layout presented in Section 7.1. In that scenario VoIP over HSDPA performance is studied with various seeds and user amounts in Vehicalar A channel. Main simulation parameters for this analysis are presented in Table 13 and Table 3. Other than that, the simulation parameters follow the parameters used for HSDPA VoIP articles included in this thesis. In addition to varying the seeds a few test cases were run with different simulation lengths to verify the sensitivity to the simulation length.

The simulation results for this analysis are produced with somewhat updated version of the simulator that was used to conduct studies presented in included article **[PV]**. Refer to the Section 7.3.2 for more details about the updated simulator version.

Analysis from these cases is focused on VoIP capacity and outage levels as those are considered as the key performance indicator in this thesis. As presented in Section 6.4.3 the 95 % coverage/capacity level is acquired by interpolating the outages and the user amounts of the simulated test cases. Outage is calculated based on the QoS criteria which takes into account the packet delay, loss and corruption.

Tables 14 and 15 present the results in terms of outage and capacity from dynamic VoIP simulation runs with $n = 31$ different random generator seeds. As mentioned earlier the confidence intervals can be used to evaluate the statistical reliability of the simulation results. Confidence intervals of 90 %, 95 % and 99 % on the basis of those simulation results are gathered to Tables 17 and 18. As those tables show, similarly to MBMS simulations, abnormalities are very minor, especially when the interpolated capacity numbers are considered. Moreover, the impact of simulation length presented in Table 16 shows only minor differences between interpolated capacity numbers. Thus, dynamic evaluations for VoIP can be considered as reliable.

TABLE 7 Satisfied Users with Different Seed Values, BLER < 10 %

E_c/I_{or} -11 dB	E_c/I_{or} -9 dB	E_c/I_{or} -7 dB	E_c/I_{or} -5 dB
0.9379	0.9989	1.0000	1.0000
0.9387	0.9991	1.0000	1.0000
0.9370	0.9989	1.0000	1.0000
0.9367	0.9992	1.0000	1.0000
0.9379	0.9987	1.0000	1.0000
0.9364	0.9989	1.0000	1.0000
0.9369	0.9988	1.0000	1.0000
0.9381	0.9990	1.0000	1.0000
0.9391	0.9988	1.0000	1.0000
0.9372	0.9989	1.0000	1.0000
0.9370	0.9988	1.0000	1.0000
0.9386	0.9989	1.0000	1.0000
0.9376	0.9988	1.0000	1.0000
0.9365	0.9990	1.0000	1.0000
0.9364	0.9989	1.0000	1.0000
0.9368	0.9990	1.0000	1.0000
0.9376	0.9990	1.0000	1.0000
0.9378	0.9988	1.0000	1.0000
0.9364	0.9989	1.0000	1.0000
0.9379	0.9988	1.0000	1.0000
0.9371	0.9991	1.0000	1.0000
0.9369	0.9989	1.0000	1.0000
0.9376	0.9986	1.0000	1.0000
0.9371	0.9989	1.0000	1.0000
0.9378	0.9988	1.0000	1.0000
0.9380	0.9990	1.0000	1.0000
0.9365	0.9990	1.0000	1.0000
0.9378	0.9988	1.0000	1.0000
0.9372	0.9991	1.0000	1.0000
0.9362	0.9990	1.0000	1.0000
0.9360	0.9990	1.0000	1.0000

TABLE 8 Satisfied Users with Different Seed Values, BLER < 1 %

E_c/I_{or} -11 dB	E_c/I_{or} -9 dB	E_c/I_{or} -7 dB	E_c/I_{or} -5 dB
0.5526	0.8489	0.9797	0.9989
0.5510	0.8507	0.9815	0.9991
0.5516	0.8469	0.9796	0.9987
0.5513	0.8477	0.9783	0.9986
0.5526	0.8481	0.9797	0.9989
0.5519	0.8495	0.9816	0.9986
0.5538	0.8461	0.9797	0.9981
0.5518	0.8486	0.9800	0.9985
0.5507	0.8491	0.9807	0.9987
0.5529	0.8476	0.9794	0.9988
0.5518	0.8495	0.9813	0.9988
0.5538	0.8489	0.9813	0.9989
0.5496	0.8488	0.9806	0.9985
0.5512	0.8490	0.9805	0.9984
0.5519	0.8472	0.9797	0.9985
0.5507	0.8466	0.9817	0.9993
0.5504	0.8478	0.9795	0.9986
0.5523	0.8482	0.9794	0.9988
0.5518	0.8480	0.9797	0.9981
0.5530	0.8526	0.9804	0.9988
0.5501	0.8492	0.9791	0.9977
0.5522	0.8491	0.9799	0.9986
0.5497	0.8466	0.9796	0.9984
0.5503	0.8483	0.9796	0.9983
0.5531	0.8518	0.9804	0.9990
0.5552	0.8496	0.9811	0.9987
0.5529	0.8469	0.9802	0.9986
0.5528	0.8506	0.9807	0.9982
0.5493	0.8481	0.9800	0.9984
0.5525	0.8476	0.9797	0.9984
0.5501	0.8484	0.9806	0.9992

TABLE 9 Required E_c/I_{or} for 95 % Coverage with Different Seeds

BLER < 1 %	BLER < 4 %	BLER < 10
-7.4544	-9.3358	-10.6024
-7.4816	-9.3465	-10.6258
-7.4466	-9.3343	-10.5808
-7.4337	-9.3364	-10.5733
-7.4518	-9.3400	-10.6012
-7.4783	-9.3418	-10.5644
-7.4449	-9.3430	-10.5778
-7.4567	-9.3450	-10.6103
-7.4670	-9.3417	-10.6341
-7.4455	-9.3385	-10.5856
-7.4750	-9.3425	-10.5802
-7.4727	-9.3462	-10.6221
-7.4638	-9.3337	-10.5946
-7.4634	-9.3445	-10.5674
-7.4486	-9.3416	-10.5644
-7.4688	-9.3342	-10.5744
-7.4483	-9.3458	-10.5958
-7.4488	-9.3362	-10.5994
-7.4512	-9.3345	-10.5644
-7.4754	-9.3453	-10.6018
-7.4476	-9.3406	-10.5845
-7.4572	-9.3407	-10.5785
-7.4454	-9.3482	-10.5933
-7.4513	-9.3403	-10.5832
-7.4723	-9.3425	-10.5994
-7.4732	-9.3365	-10.6055
-7.4528	-9.3414	-10.5674
-7.4726	-9.3377	-10.5994
-7.4547	-9.3391	-10.5868
-7.4499	-9.3397	-10.5605
-7.4625	-9.3422	-10.5559

TABLE 10 Statistical Confidence on Satisfied Users, BLER < 10 %

	E_c/I_{or} -11 dB	E_c/I_{or} -9 dB	E_c/I_{or} -7 dB	E_c/I_{or} -5 dB
MEAN	93.73 %	99.89 %	100.00 %	100.00 %
STANDARD DEVIATION	0.0008	0.0001	0.0000	0.0000
CONFIDENCE INTERVAL, 90 %	± 0.02 %	± 0.00 %	± 0.00 %	± 0.00 %
CONFIDENCE INTERVAL, 95 %	± 0.03 %	± 0.00 %	± 0.00 %	± 0.00 %
CONFIDENCE INTERVAL, 99 %	± 0.04 %	± 0.01 %	± 0.00 %	± 0.00 %

TABLE 11 Statistical Confidence on Satisfied Users, BLER < 1 %

	E_c / I_{or} -11 dB	E_c / I_{or} -9 dB	E_c / I_{or} -7 dB	E_c / I_{or} -5 dB
MEAN	55.18 %	84.86 %	98.02 %	99.86 %
STANDARD DEVIATION	0.0014	0.0015	0.0008	0.0003
CONFIDENCE INTERVAL, 90 %	± 0.07 %	± 0.05 %	± 0.02 %	± 0.01 %
CONFIDENCE INTERVAL, 95 %	± 0.09 %	± 0.06 %	± 0.03 %	± 0.01 %
CONFIDENCE INTERVAL, 99 %	± 0.12 %	± 0.08 %	± 0.04 %	± 0.02 %

TABLE 12 Statistical Confidence on 95 % Coverage Level

	BLER < 1 %	BLER < 4 %	BLER < 10 %
MEAN	-7.4586 dB	-9.3405 dB	-10.5882 dB
STANDARD DEVIATION	0.0123	0.0040	0.0197
CONFIDENCE INTERVAL, 90 %	± 0.05 %	± 0.01 %	± 0.06 %
CONFIDENCE INTERVAL, 95 %	± 0.06 %	± 0.02 %	± 0.07 %
CONFIDENCE INTERVAL, 99 %	± 0.08 %	± 0.02 %	± 0.09 %

APPENDIX 1.2 Confidence Analysis on Quasi-static Simulations

In this section the quasi-static simulation results are verified using HSUPA full buffer simulations. Full buffer means that UEs will have their transmission data buffers full at all times. As already shown above in the respect of dynamic simulator the results are reliable and reproducible regardless of the service. Thus, full buffer analysis is applicable also for the VoIP simulations produced for this thesis with quasi-static simulator. Main simulation parameters for quasi-static confidence analysis is presented in Table 19. Notice that full buffer type of traffic provides a lot of samples and thus relatively short simulation time can be utilized with adequate number of drops.

Results from quasi-static full buffer simulation runs with $n = 31$ different random generator seeds are illustrated in Figure 53 and respectful confidence intervals shown in Table 20. As that figure and confidence intervals show there is again only minor variation in the results. Thus, also the results produced through quasi-static approach are well within the lines of statistical confidence.

TABLE 13 Main Simulation Parameters for Statistical Analysis of VoIP Simulations

Simulation parameter	Value(s)
Scenario	Wrap-around, 21 hexagonal cells
Cell radius	933 m
Pathloss model	Modified Okumura-Hata
Channel profile	Vehicular A
Receiver	Rake 1Rx
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
Power resources HSDPA	15 W
Power resources HS-DSCH	10 W
Code resources HS-DSCH	10
Power resources HS-SCCH	2 W
Number of HS-SCCH's	4
Maximum DCH active set size	3
DCH Add, replace, drop window	[1.5 dB, 2 dB, 2.5 dB]
HSDPA HO threshold	2 dB
HSDPA HO time to trigger	100 ms
HSDPA HO delay	200 ms
Max. number of HARQ transmissions	3
Packet scheduling algorithm	VoIP optimized proportional fair
UE velocity	30 km/h
SID	disabled
Average VoIP call length	40 s
Minimum VoIP call length	20 s
Maximum VoIP call length	60 s
Simulation time	100, 200 and 400 s

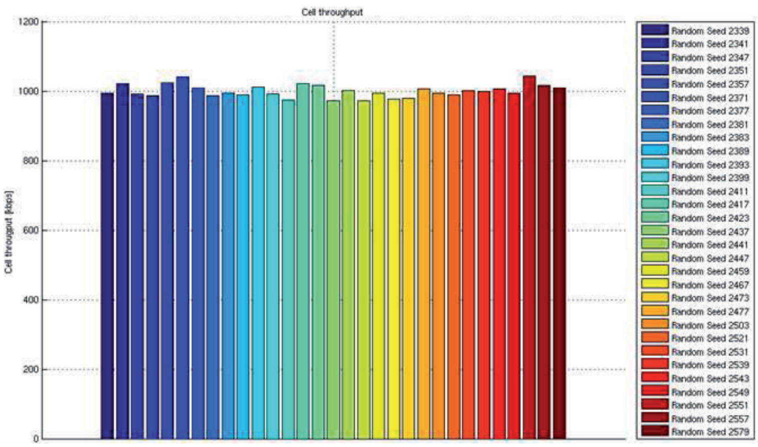

FIGURE 53 Quasi-static Simulator Confidence Analysis Results

TABLE 14 VoIP over HSDPA Outage Level with Different Seeds

1400 UEs/cell, Outage [%]	1600 UEs/cell, Outage [%]
1.5325	9.1996
1.5941	10.7498
1.5649	7.5235
1.8661	9.9737
1.7029	10.2051
1.494	8.8198
2.3881	11.4415
1.1871	9.9487
1.3141	9.6103
1.2423	7.6711
2.1745	8.5621
1.3758	10.8122
1.7796	9.3892
1.1149	8.9881
2.0986	10.1659
1.8387	9.9099
1.701	9.6065
1.6067	9.3758
2.3857	10.738
2.312	8.6826
1.3691	10.9296
1.5632	10.0748
1.9142	8.0996
1.429	9.6956
1.6918	8.0939
1.7229	8.4232
1.6401	9.2506
1.5427	9.2877
1.6937	10.01
1.3706	9.7741
1.7632	8.6935

TABLE 15 VoIP over HSDPA 95 % Outage Level with Different Seeds

Capacity [users/cell]	Capacity [users/cell]
70.9739	70.2095
72.1570	70.3480
70.3599	71.2245
69.4125	70.8113
70.8869	72.2335
70.8794	70.3245
70.6972	71.3663
70.0919	70.3969
70.6410	70.8264
69.6476	70.6852
70.2836	70.5122
71.4179	70.7808
71.5880	71.3248
70.8713	70.9180
70.4530	70.7799
71.1148	

TABLE 16 VoIP over HSDPA 95 % Outage Level with Different Simulation Lengths

Simtime 0.5x	Simtime 1x	Simtime 2x
70.9647	70.9739	69.9607

TABLE 17 Statistical Confidence on VoIP over HSDPA Outage Level

	˜66.67 UEs/cell, Outage [%]	76.19 UEs/cell, Outage [%]
MEAN	1.68	9.47
STANDARD DEVIATION	0.33191	0.97242
CONFIDENCE INTERVAL, 90 %	$\pm$ 5.85 %	$\pm$ 3.03 %
CONFIDENCE INTERVAL, 95 %	$\pm$ 6.97 %	$\pm$ 3.61 %
CONFIDENCE INTERVAL, 99 %	$\pm$ 9.16 %	$\pm$ 4.75 %

TABLE 18 Statistical Confidence on 95 % VoIP over HSDPA Outage Level

	Capacity [UEs/cell]
MEAN	70.78122
STANDARD DEVIATION	0.612021
CONFIDENCE INTERVAL, 90 %	± 0.25 %
CONFIDENCE INTERVAL, 95 %	± 0.30 %
CONFIDENCE INTERVAL, 99 %	± 0.40 %

TABLE 19 Main Simulation Parameters for Confidence Analysis on Quasi-static Simulations

Simulation parameter	Value(s)
Scenario	Wrap-around, 57 hexagonal cells
Inter-Site Distance	1000 m
Pathloss model	Modified Okumura-Hata
Channel profile	Pedestrian A
Node B Receiver	LMMSE 2Rx
Max. UE Tx power	24 dBm
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
Simulation time	12 s
Warm-up time	2 s
Number of drops	4
RoT target	7 dB
TTI lenght	2 ms
Max. number of HARQ transmissions	3
Scheduling algorithm	PF

TABLE 20 Statistical Confidence on Quasi-static Simulations

	Cell throughput [kbps]
MEAN	1000,036
STANDARD DEVIATION	17,95
CONFIDENCE INTERVAL, 90 %	± 0,53 %
CONFIDENCE INTERVAL, 95 %	± 0,63 %
CONFIDENCE INTERVAL, 99 %	± 0,83 %

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