JYVÄSKYLÄ LICENTIATE THESES IN COMPUTING

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Kari Aho

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ABSTRACT

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Finnish summary

The purpose of this thesis is to address the performance of *Multimedia Broadcast Multicast Service (MBMS)* and *Voice over IP (VoIP)* services in *Wideband Code Division Multiple Access (WCDMA)*/*High Speed Packet Access (HSPA)* networks. MBMS and VoIP services have already gained vast popularity in the fixed networks and thus it is very likely that they will also be key concepts in wireless cellular networks in the near future. However, from practical perspective, wireless network sets very strict environment for this kind of services. Thus, it is crucial to be able to evaluate the need of radio resources, possible enhancements and vast number of practical network parameters so that cellular network and these services operate in optimal manner in different situations. This kind of information can be then used by cellular operators in addition to network and product vendors in dimensioning and fine-tuning cellular networks. Special attention in this thesis will be paid on the mobility aspect of both of those services. Studies include the performance evaluation of MBMS with macro diversity combining with and without receive diversity. From the perspective of macro diversity soft and selective combining are evaluated. In addition, concept called Rx-switching is evaluated in order to provide some battery saving opportunities for MBMS terminals employing multiple receive antennas. On the area of VoIP this thesis addresses the performance both in uplink and downlink in terms of user velocity, handover parameters and delays. This kind of performance evaluation is critical as VoIP is very sensitive to any additional delays and/or packet losses that handover performance could potentially cause. This thesis evaluates the performance with help of fully dynamic system simulator where user mobility, fading, propagation and radio resource management functionalities are explicitly taken into account.

Keywords: Mobility, Handover, Handover delay, Receive diversity, Transmit diversity, Macro diversity, Soft combining, Selective combining

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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
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
C/I	Carrier to Interference Ratio
CAPEX	Capital Expenditure
CBS	Cell Broadcast Service
CDMA	Code Division Multiple Access
CPICH	Common Pilot Channel
CSCF	Call Session Control Function
CN	Core Network
Codec	Coder Decoder
CPCH	Uplink Common Packet Channel
CS	Circuit Switched
CS	Combining Set
CRC	Cyclic Redundancy Check
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
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
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
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
ITU	[International Telecommunication Union
L1	Layer 1
LA	Link Adaptation
LTE	Long Term Evolution
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
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
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'99	Release 99
RFC	Request for Comments
RLC	Radio Link Control
RNC	Radio Network Controller
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
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
WCDMA	Wideband Code Division Multiple Access
WLAN	Wireless Local Area Network

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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.

The author of this thesis was the main author of articles **[PI]** and **[PII]** for which he took the main responsibility of the simulation work and participated in the implementation work to add performance indicator statistics to the research tool. In article **[PIII]** the author conducted majority of the simulations, contributed to the result analysis and was the main author of the article. The author of this thesis was responsible of the implementation and modeling work, carrying simulations and writing process in article **[PIV]**. Finally, in article **[PV]** the author participated in simulation scenario planning, analysis and writing of the article.

1 INTRODUCTION

Wireless telecommunications industry has undergone many even revolutionary steps within timespan 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 offering voice services were taken into use. First generation networks, based on analogue technology, were replaced by digital *Global System for Mobile Communications (GSM)* 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)*. Along with SMS technology and capability to provide other small bit rate data services GSM has reached huge level of success by reaching close to 2.7 billion subscribers world wide by the end of 2007 [58].

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)* [26] technique were taken into use. With the highest commercially implemented data rate of 384 kbps [26] 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 [27] to 3G networks were introduced. HSPA i.e. 3.5G networks can provide theoretical peak data rates up to 14.4 Mbps. These higher bit rates companied by the new range of services have already attracted over 440 million subscribers world wide by the end of fourth quarter 2008 [58].

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)* [7]. 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 is 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. 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 *Capital Expenditure (CAPEX)* and *Operating Expense (OPEX)* related to it. Additionally, by having services like *Voice over Internet Protocol (VoIP)* i.e. voice carried over PS elements it would enable much more simpler implementation of rich call services as the voice and data would be then carried via same network elements.

MBMS and VoIP are undeniably the key concepts in wireless cellular networks in the future and thus their performance and possible performance enhancements require close analysis before actual network implementations. This thesis aims to answer to the question of MBMS and VoIP performance in 3G/3.5G networks. In the next section, the research problem of this thesis will elaborated further. Studies related to research problem are presented and discussed in 1.2. Following sections after related studies deal with basics of WCDMA, HSPA, MBMS, VoIP and finally achieved results accompanied with the presentation of the research tool.

1.1 Research problem

When considering user's mobility in the network, it is crucial to optimize the *Radio Resource Management (RRM)* functions in the network to prevent major performance losses. From the RRM functions, the impact of *Handovers (HO)* are especially emphasized in this thesis. Moreover, this thesis focuses on evaluating how the handover delays, *Active Set (AS)* sizes and other handover related parameters take effect on the overall system performance in terms of MBMS and VoIP concepts.

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. However, in order to evaluate the concept from the perspective of the first 3G devices that do not support HSPA evolutions, this thesis focuses on 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. Macro diversity is the most likely option for performance enhancement as the current trends indicate that MBMS is going to be available through large parts of the network.

Receive diversity, on the 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 [45]. However, the size, cost and power resources of UE somewhat 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.

Taking previously mentioned aspects into account this thesis aims to answer to the following questions with the respect of MBMS performance and performance enhancements:

- How much MBMS can benefit from macro diversity combining?
- How do the handover parameters effect to the macro diversity combining schemes?
- What kind of performance enhancement can be achieved with receive and transmit diversity?
- When network is dimensioned to terminals with *one receive antenna (1RX)* and we have terminals with *two receive antennas (2Rx)*, how good power savings opportunities in the UE can we accomplish by switching off additional receiver branches when radio conditions allow?

From the perspective of VoIP, the *Release 99 (Rel'99)* by *Third Generation Partnership Project (3GPP)* introduced WCDMA which made it possible for the first time to run VoIP over cellular networks with reasonable quality but however 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)* [27]. 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)* and shortened *Transmission Time Interval (TTI)* [27]. However, as the previous studies regarding the HSPA evolutions have proved, also delay critical small bit rate services such as VoIP can benefit from those new features. This thesis aims to deepen the knowledge of VoIP performance over HSDPA and HSUPA in different mobility related scenarios. This kind of study is essential as voice services are very sensitive to any additional delays and/or packet losses that user mobility can cause.

VoIP over HSPA studies in this thesis aim to answer to the following questions:

- What kind of impact user velocity has on the VoIP performance?
- How much VoIP performance is degraded due to delays in the handover procedure?
- How well can VoIP users be satisfied when active set size is limited?
- How reliable serving HSDPA cell change is?
- Do *Silence Indicator Frames (SID)* i.e. comfort noise during silence intervals have significant impact to the overall performance?

To answer these questions regarding MBMS and VoIP concepts, fully dynamic system simulations are carried over. There are at least two factors advocating the usage of system simulations: on the one hand, the complexity of the dynamic 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, fully dynamic system level simulations where user mobility, RRM functionalities and their inter-actions are taken explicitly into account are essential to the performance evaluations.

1.2 Related studies

3GPP is currently standardizing the MBMS framework and thus studies related to the MBMS performance and its enhancements are very crucial. Also as mentioned before, cellular networks are adopting VoIP and moving toward all-IP networks which is proved by the fact that e.g. *Long Term Evolution (LTE)* systems, which are to follow 3G/3.5G networks, support only PS voice [2]. Thus, studies regarding MBMS and VoIP support the standardization work and in addition aid network vendors and operators to set many practical parameters so that the network performance would be optimal in different situations. This section aims to point out some of the most important studies related to this thesis and elaborate how this thesis deepens that knowledge.

1.2.1 Multimedia Broadcast Multicast Service

Extensive capacity analysis for MBMS in 3G networks was carried out in [23], 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 [19], 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 [43] 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 [14] and later extended to cover also macro diversity concepts in [51]. 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 [17] 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 [17] 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 as it is very limited resource. 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 to the key parameters which have to be set in practice. Those include e.g. threshold values for adding, dropping and replacing a certain 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. [56] and [44]. In [56] scheduling algorithm for VoIP traffic is presented which is extended in [44] with dynamic transmission (code and power) resource allocation method. According to results presented in [44] with both of those scheduling related enhancements VoIP performance can be improved in terms of capacity even 115 % when compared to traditional RR scheduler.

Apart from scheduling related issues, impact of mobility and thus serving *High Speed Downlink Shared Channel (HS-DSCH)* change procedure is very crucial regarding services, such as VoIP, which are highly sensitive to excess delay and packet losses. Mobility aspect in terms of VoIP over HSDPA are addressed e.g. in [55]. 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 impact the overall VoIP performance.

As coverage is typically limited by the uplink, studies regarding VoIP over HSUPA are also important to evaluate the overall performance of VoIP. The capacity of how many VoIP users can be served per cell 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 [16]. 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 effect 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 in reduced efficiency and maximum transmission power. These power issues have impact on the VoIP capacity and therefore it is crucial to study their impact. This kind of study is presented in [49]. 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 TPC error rate.

This thesis aims to deepen the knowledge of VoIP over HSPA performance by conducting detailed studies where e.g. mobility of users and interactions of the radio resource management functionalities are explicitly taken into account. By doing this we are 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 on the impact of (hard) HSDPA and (soft) HSUPA handovers, how sensitive VoIP service is to handover/active set update delay, velocity and thus find out the conditions where VoIP users are able to smoothly switch from cell to cell while maintaining good quality of service.

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 [26] and in 3GPP specifications [1]. Uplink and downlink packet evolutions, HSUPA and HSDPA respectively, to WCDMA are introduced in detail in [26], [27] and [20]. 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 Fig. 1, consists of such logical elements as *Radio Network Controller (RNC)* and *Node B* [26].

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.

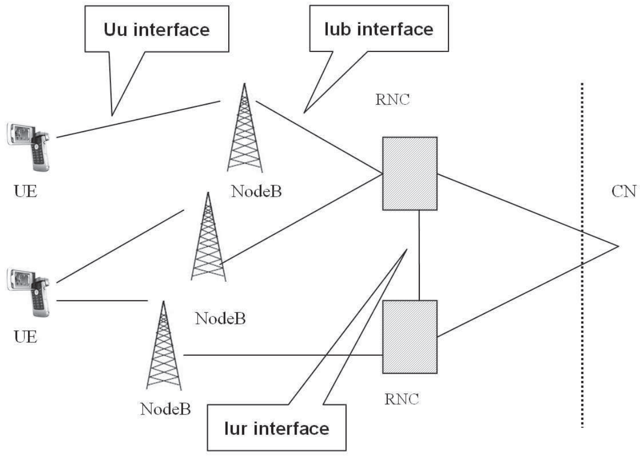

FIGURE 1 UMTS Architecture

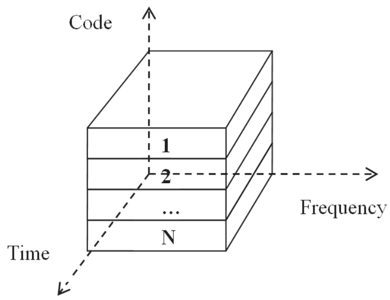

FIGURE 2 CDMA Principle

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 Fig. 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. 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

	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 Fig. 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, it can be concluded that the amplitude 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 [26].

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 Fig. 4. As a result,

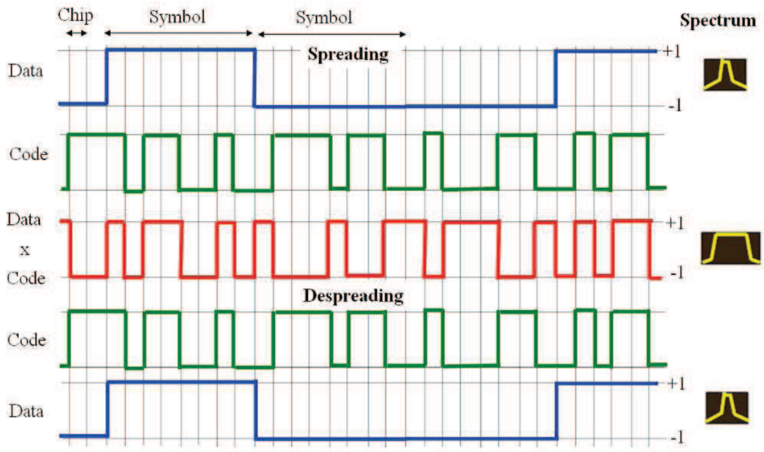

FIGURE 3 Example of signal spreading and despreading, [26]

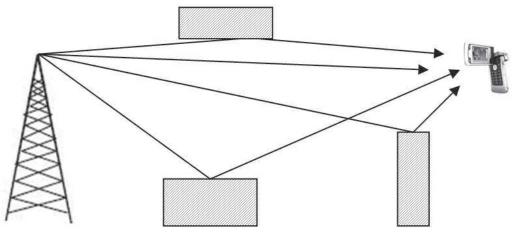

FIGURE 4 Example of radio signal propagation

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 [28]. 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 [15].

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.

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 [46], which were then fitted into mathematical model by M. Hata [24]. Initially Okumura-Hata model was applicable only for lower bandwidths but later it was extended with COST 231 model [21], 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 [13]. 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.

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 [13] [50].

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 Fig. 5. This kind of phenomenon is referred as small scale or fast fading [50].

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

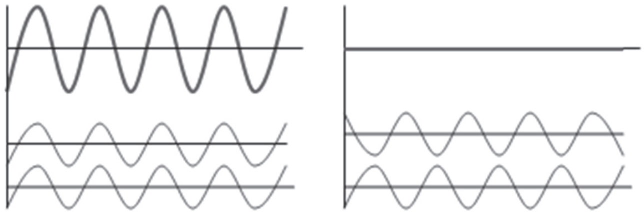

FIGURE 5 Example of constructive (left hand side) and destructive (right hand side) addition

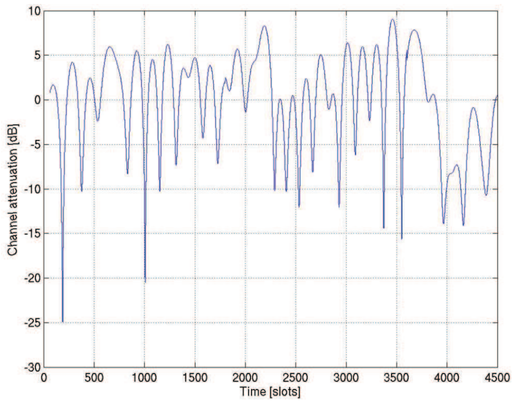

FIGURE 6 Example of fast fading process

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 Fig. 6.

2.5 Receiver

In WCDMA systems the most commonly used receiver is so called Rake receiver. Rake receiver is specially designed to compensate the effects of fading ([47], [48], [52], [53], [26]). 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 receive their individual multipath components which are then modified with channel estimate as illustrated in Fig. 7. Channel estimate is acquired from known

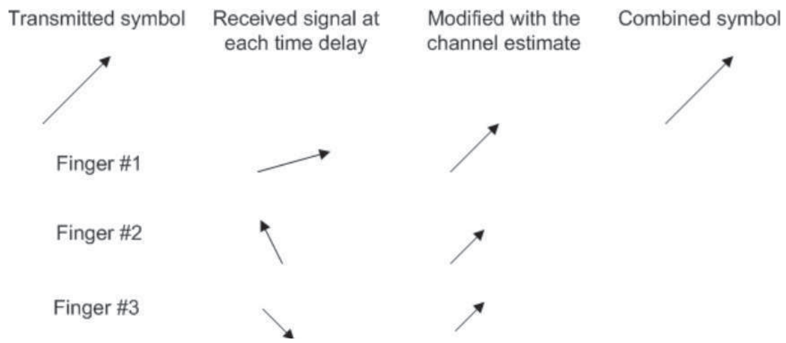

FIGURE 7 Principle of Maximum Ratio Combining within the Rake receiver [26]

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 [10], for macro diversity combining are briefly presented.

2.6.1 Selective Combining

Receiver with selective combining, Fig. 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 Fig. 9, is called as soft combining. Unlike selective combining, soft combining is performed on the physical layer. Receiver that uses soft combining collects all

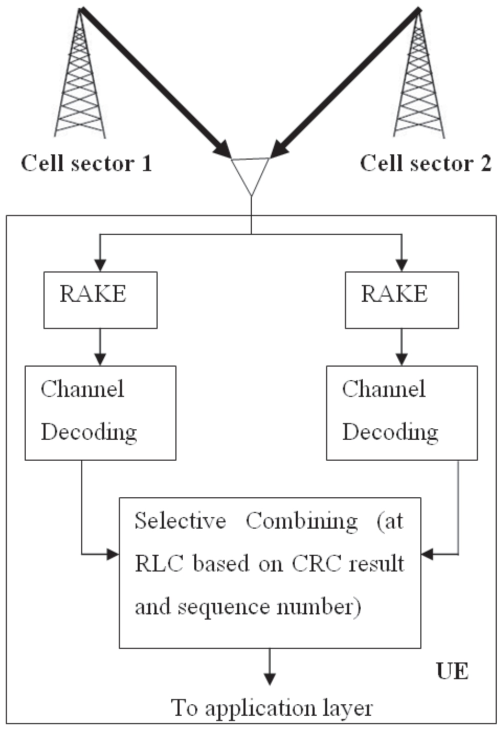

FIGURE 8 Selective Combining

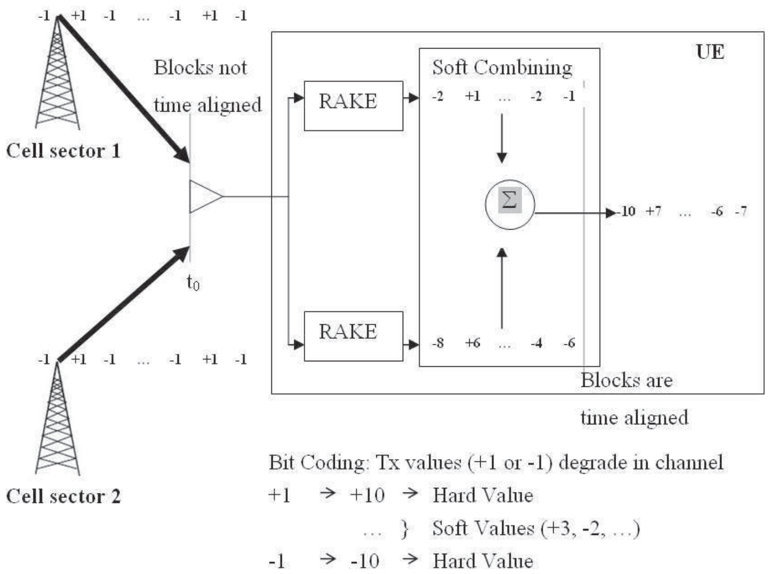

FIGURE 9 Soft Combining

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 [29] means that the receiver has two or more receive antennas that are used to collect the multipath components. Even though there is 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 from one to 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 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 [45]. In addition to 'traditional' receive diversity also a concept closely related to the receive diversity called Rx-switching i.e. turning the another 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 [26]. The purpose of PC is to ensure that each user receives and transmits just enough energy to prevent blocking of distant users (near-far-effect) and exceeding reasonable interference levels.

Fig. 10 illustrates how users further away from the base station might be blocked out without power control. In that example, without PC, 'UE3' might be blocked out due to trying to send with as high power as possible to guarantee higher signal interference ratio. However, 'UE1' and 'UE2' have relatively more power available as they are closer to the BS and more likely in a better radio channel condition. However, in this example they would naturally increase their power as the interference level is increased because of 'UE3'. Eventually this competitive situation would lead to situation where power limited 'UE3' would be highly interfered by 'UE1' and 'UE2' and thus might be blocked out. Therefore power control is an essential part of WCDMA.

The actual power control can be divided into open loop power control, which is also referred as slow power control and to closed loop power control, which is referred as fast power control. Open loop 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 closed loop power control steps in. Closed loop power control is used to eliminate the effect of fast fading and it is applied 1500 times per second, hence the name fast power control.

Moreover, closed loop power control can also be divided into two parts: *Innerloop Power Control (ILPC)* and *Outerloop Power Control (OLPC)*. In ILPC the signal levels are measured and compared to the target value. If the value is higher (resp. lower) than the target then the power is lowered (resp. increased) with a certain step. OLPC, on the other hand, adjusts 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 which switches one radio channel to another in order to guarantee the defined quality of bearer service and continuity of an established call. The purpose of this section is to present what kind of handover categories and types there are in addition to soft(er) handover procedure which is essential part of the WCDMA [26].

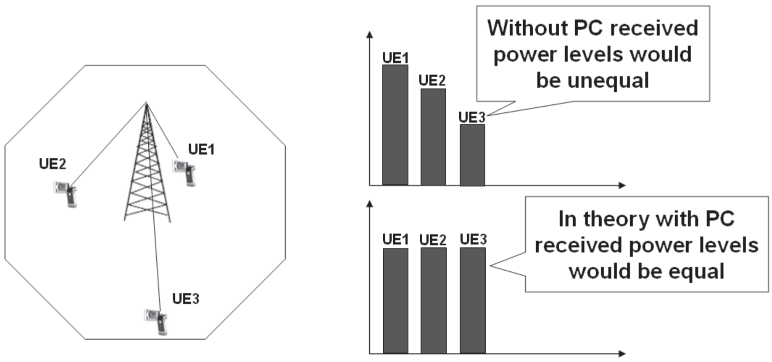

FIGURE 10 Power Control in WCDMA

2.9.1 Handover categories

WCDMA supports three kind 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 the 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 mentioned before soft and softer handover maintain connection to multiple base stations before the old connection is released in order to make the handover

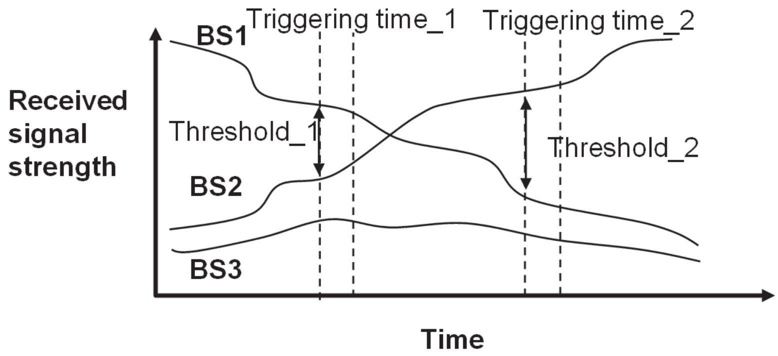

FIGURE 11 Soft and softer handover procedure

as seamless as possible. This procedure is characterized by number of parameters and the purpose of this section is to cover them and their purpose briefly.

Example of soft(er) handover procedure is illustrated in Fig. 11. If we start to follow the x-axis of that figure which represents time in this example, at first UE is connected to only 'BS1' i.e. UE's active (or combining) set contains only 'BS1'. A bit later signal from 'BS2' reaches the 'Threshold_1' which is more commonly known as 'Add window'. 'Add window' represents a value of how much worse a new signal can be compared to the best one in the current active set in order to be added into the set. Adding link to combining set can be done only if maximum number of links is not full yet (defined with parameter). Moreover, a new link is added to the active set only if the difference between the best and the new is still at least as good after the 'Triggering time_1' ('add timer') is expired. Timer is started when the signal first reaches the desired level.

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, parameter called as '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)* [26].

In WCDMA systems there exists only one dedicated channel which is known as DCH. DCH is used to carry user data and all higher layer control information, such as handover commands. DCH is characterized by features such as fast power control and soft handover. 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. Due to the reason that FACH carries vital control information, FACH has to have such a low bit rate that it can be received by all the UEs in the cell area. However, there can be more than one FACH in a cell which makes it possible 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 different common channels in WCDMA: the *Broadcast Channel (BCH)*, *Paging Channel (PCH)*, *Random Access Channel (RACH)*, *Uplink Common Packet Channel (CPCH)* and *Dedicated Shared Channel (DSCH)*. From the common channels most notable in terms of data transmissions is FACH as DSCH was optional feature that was seldom implemented by the operators and later replaced in practice with HSDPA. Thus, 3GPP decided to take DSCH away from Release 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 [26].

2.11 High Speed Packet Access

High Speed Packet Access (HSPA) consists of improved packet data capabilities in both uplink and downlink for WCDMA networks. 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)*, 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 e.g. [56], [44], [55] and [16]), also delay critical small bit rate services, such as VoIP, can benefit from those new features.

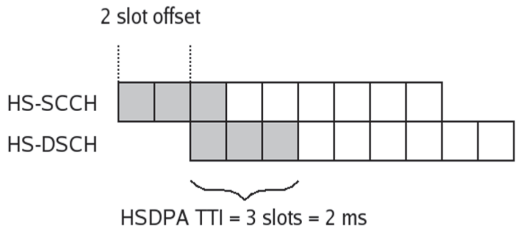

FIGURE 12 Offset with HSDPA channels

The following sections deal with those new features and properties of HSPA which are the most essential in terms of VoIP. The reason why some of the key features of HSPA are neglected in the following sections is that VoIP does not require high peak data rates that are achieved in HSPA e.g. through *Link Adaptation (LA)* and higher order modulation and coding. More details of HSPA can be acquired from e.g. [27] and [20].

2.11.1 High Speed Channels

As described above the introduction of high speed channels was two phaser. HSDPA introduced in Rel'5 brought some changes to downlink packet data operations and Rel'6 along with HSUPA brought changes in the uplink. These changes extend to the channel structures also 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 slot before HS-DSCH to inform the scheduled UE of the transport format of the incoming transmission on HS-DSCH. Timing offset is illustrated in Fig. 12.

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*

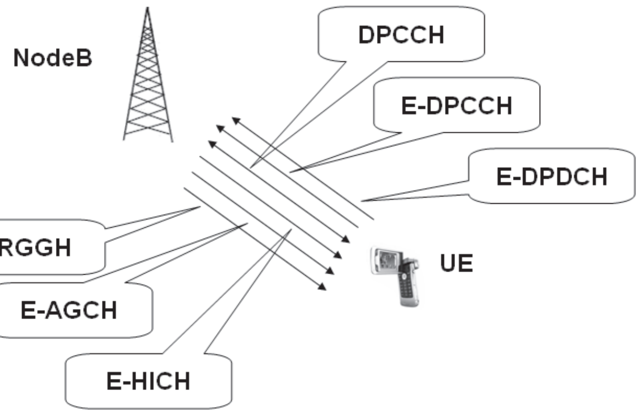

FIGURE 13 HSUPA channels

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. DPCCH is used e.g. for fast power control.

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 [26]. 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 Fig. 13.

2.11.2 Fast Retransmissions

HSPA introduced changes to the basic WCDMA architecture in terms of transferring functionality from RNC to the Node B. Fast retransmissions are one of the new improvements placed in the Node B level whereas in Rel'99 retransmissions were possible to employ only from RNC, see Fig. 14. Fast retransmissions are possible to be initiated/requested from Node B as the scheduling responsibility is one of the key functionalities that has been moved to Node B. Scheduling will be addressed more detailed later in this thesis.

The benefit from fast retransmissions comes from layer 1 signaling, depicted

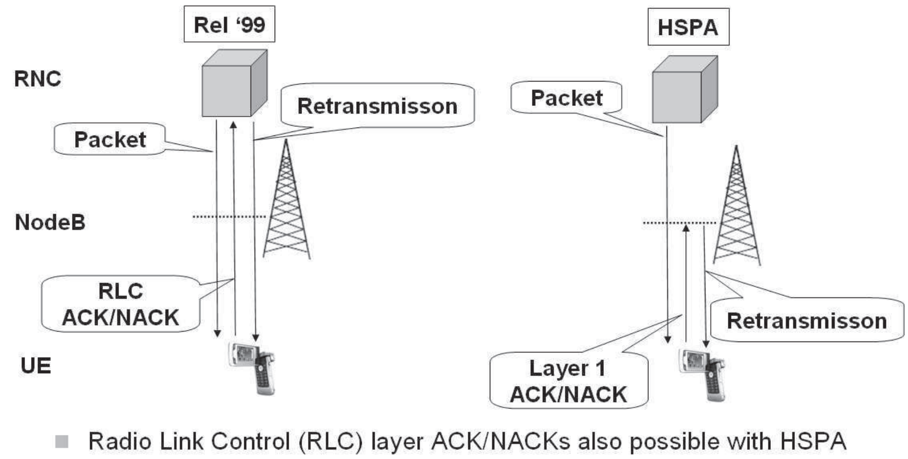

FIGURE 14 Rel'99 retransmissions versus fast retransmissions in HSPA

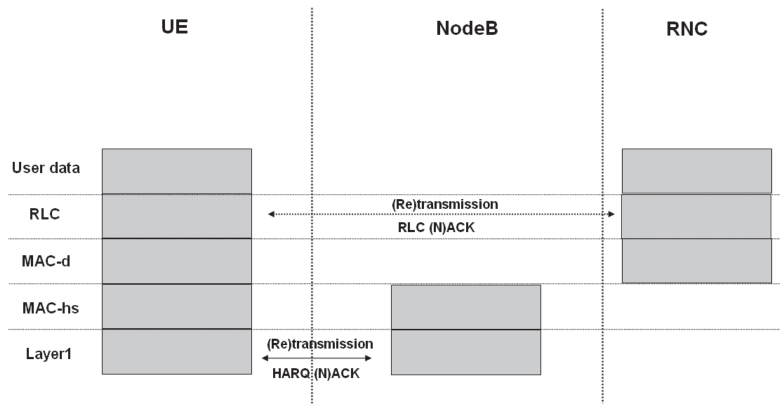

FIGURE 15 HSPA retransmissions on protocol level

in Fig. 15, which indicates the need of retransmission much faster leading to much lower round trip time 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 (i.e. CRC check) fails but combines them with retransmissions. 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 [27].

2.11.3 Code multiplexing in HSDPA

As referred earlier, HSDPA was designed primarily to provide higher data rate delivery to end-users. When introducing relatively low bit rate services, such as VoIP over HSDPA, downlink capacity could be wasted by serving only one

(VoIP) user at a time. For situations like that HSDPA introduces so called code-multiplexing to allow more than one user being scheduled in a single TTI [27]. According to Rel-6 specifications, 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.

2.11.4 Scheduling for HSDPA

In terms of delay critical services, such as VoIP, scheduling, i.e. when users are allowed to transmit or to be transmitted to, is definitely one of the most important aspects in terms of keeping the delay in reasonable limits. Thus, this and the following section cover briefly how scheduling is handled in the downlink and uplink directions. This section will deal with downlink scheduling and firstly couple of the most common scheduling schemes are presented and after that they are followed with potential VoIP enhancements.

Round Robin (RR) 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 the another hand, channel conditions are not taken into account and thus valuable radio resources might be wasted and thus network throughput might suffer.

When compared to RR *Max C/I* in 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.

Proportional Fair (PF) 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:

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

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 [26].

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 [56] and [44]. 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.

2.11.5 Scheduling for HSUPA

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 allowing to make scheduling decisions with minimum latency. Regarding HSUPA two different scheduling schemes are defined: scheduled transmissions and *Non-Scheduled Transmissions (NST)* [27].

Scheduled transmissions are, as the name implies, controlled by Node B. Node B and UEs will signal at each time instant of how much resources UEs would need and then 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.

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.

2.11.6 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 [27]. With HSDPA, the user can be connected to one serving HSDPA Node B at the time leading to hard handover when the

[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 [54]

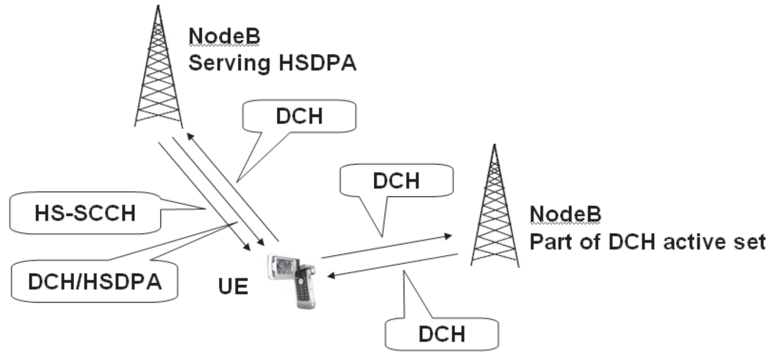

FIGURE 16 HSDPA hard handover

handover is required. On the another 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 Fig. 16 illustrates. The handover procedure with HS-DSCH is initiated when a link in (DCH) active set becomes higher is 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 [27]. 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 Fig. 17.

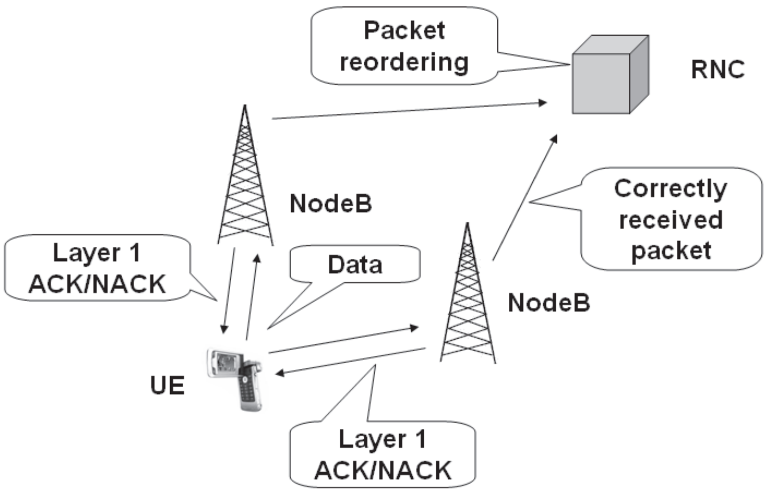

FIGURE 17 HSUPA soft handover

3 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 themselves 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.

3.1 Introduction

Up until recent times broadcast and multicast transmissions in 3G networks have been dealt with by using either the *Cell Broadcast Service (CBS)* which is defined in [3] and [4] or the *IP Multicast Service (IP-MS)* which is described in [5] and [6]. 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), [7].

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-

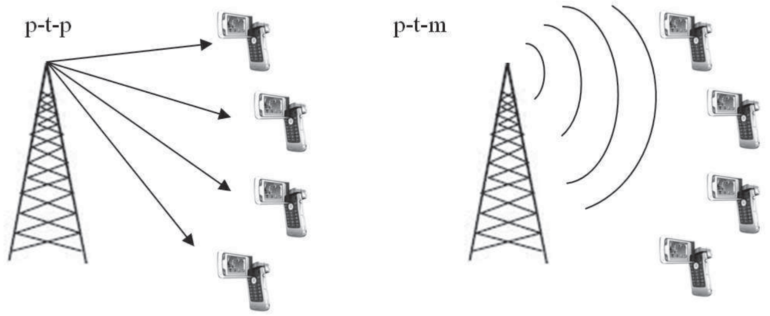

FIGURE 18 Different ways to deliver data with MBMS

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 Fig. 18. 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.

3.2 Services

Structure of MBMS service provision is two-fold: one for multicast mode and the other for broadcast mode. According to 3GPP's specifications [8], MBMS multicast services contain following stages: subscription, service announcement, joining, session start, MBMS notification, data transfer, session stop and leaving as depicted in Fig. 19. 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 Fig. 20.

Fig. 21 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-

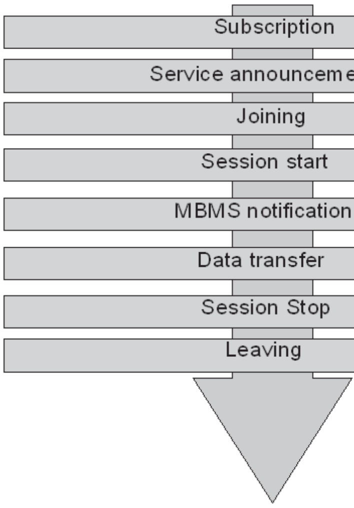

FIGURE 19 Basic structure of MBMS multicast service provision [8]

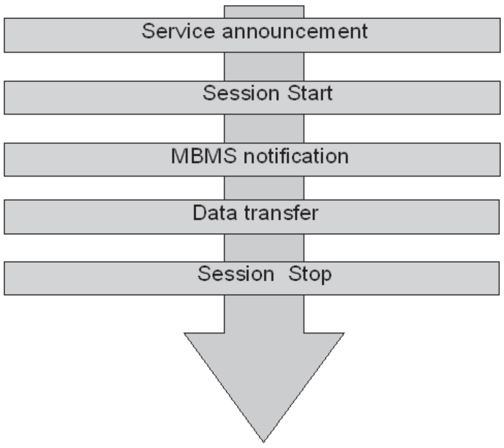

FIGURE 20 Basic structure of MBMS broadcast service provision [8]

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 [8], 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. Morever, as Fig. 21 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 [8] and [10].

3.3 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, [9].

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 [9] 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,

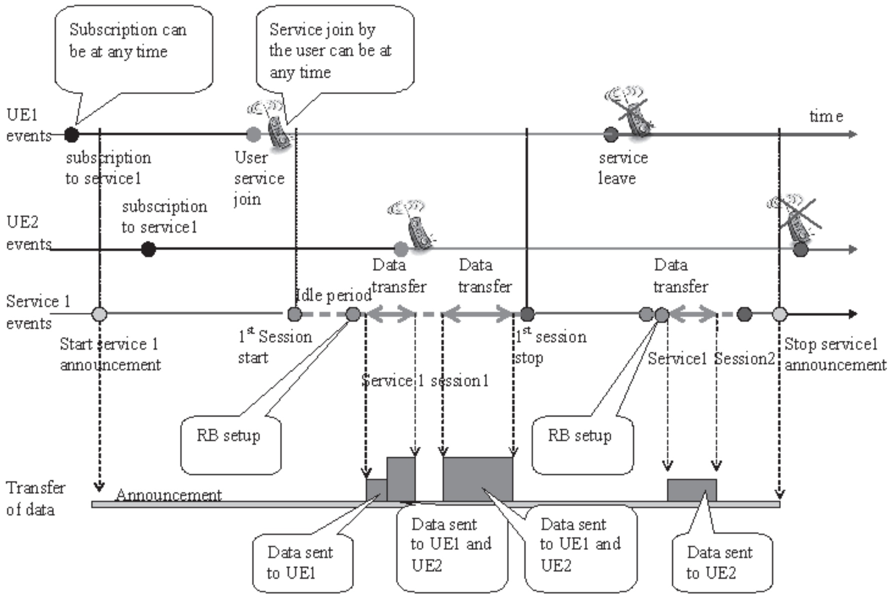

FIGURE 21 Timeline example of MBMS multicast service provision, [8]

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.

3.4 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 [8] and illustrated in Fig. 22. MBMS related entities and their functionalities are discussed briefly in the following.

Starting from the left side of the Fig. 22, 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.

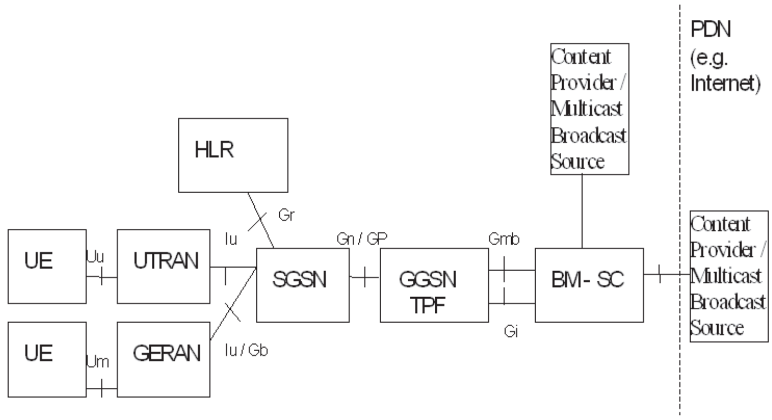

FIGURE 22 MBMS architecture; reference model, [8]

Depending on UE's capabilities, it is possible to support receiving of MBMS and 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 [8], 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 [8] 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,

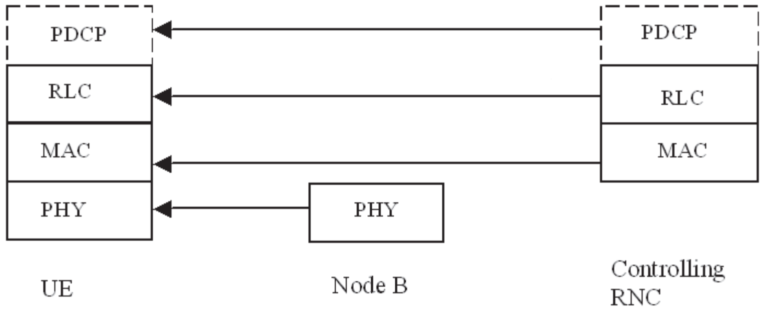

FIGURE 23 Protocol stack for MBMS traffic channel, [10]

- schedule and deliver MBMS transmissions.

3.5 Protocols

The purpose of this section is to cover MBMS protocol stack very briefly. More information of MBMS protocol stack can be found from [10] and additional details of basic WCDMA protocol stack layers (RLC, MAC and *Physical Layer (PHY)*) from [26].

Fig. 23 illustrates the protocol stack that is used for data transmissions and Fig. 24 illustrates protocol stack for control information. As those figures show, only the top layers differ in these two protocol 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 Fig. 24 is used for radio recourse control with MBMS transmissions.

3.6 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 [10].

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

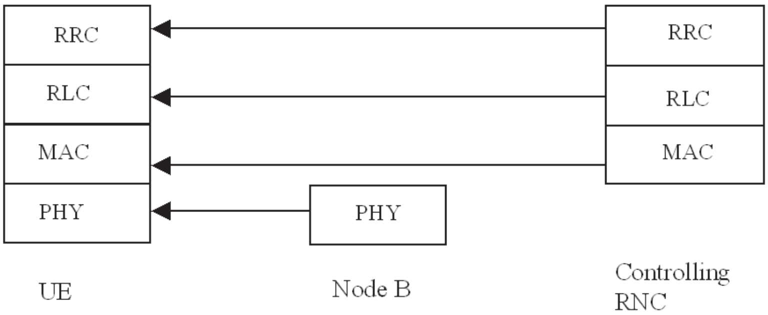

FIGURE 24 Protocol stack for MBMS control channel, [10]

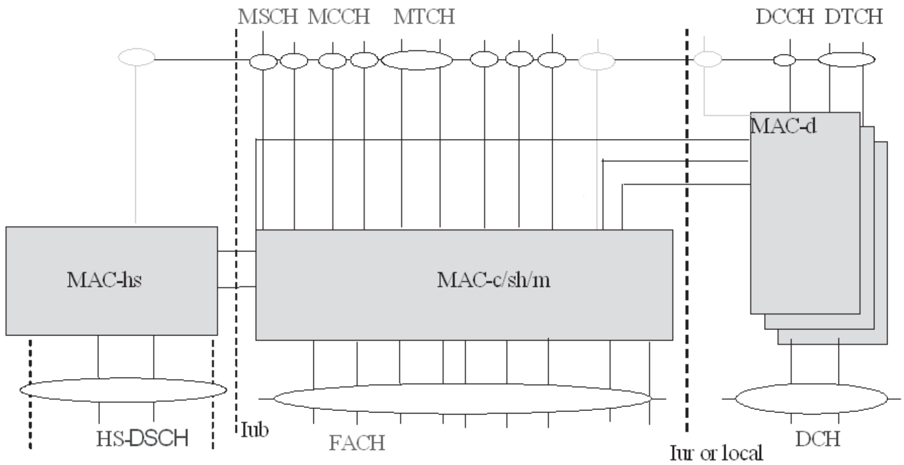

FIGURE 25 Simplified UTRAN MAC architecture with logical MBMS channels, [10]

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 Fig. 25.

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 [10] 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 Fig. 25.

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.

3.7 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 [10].

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 [10] 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.

4 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 is not tolerable for packet loss, high 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.

4.1 Introduction

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 ([25]) 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.

4.2 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 [12].

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 ones 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 thus cancellation is required [18].

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/queueing, transmission delay and the network delays. According to ITU e-model [41], the maximum acceptable mouth-to-ear delay for voice is on the order of 250 ms as the the reference values in Fig. 26 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-

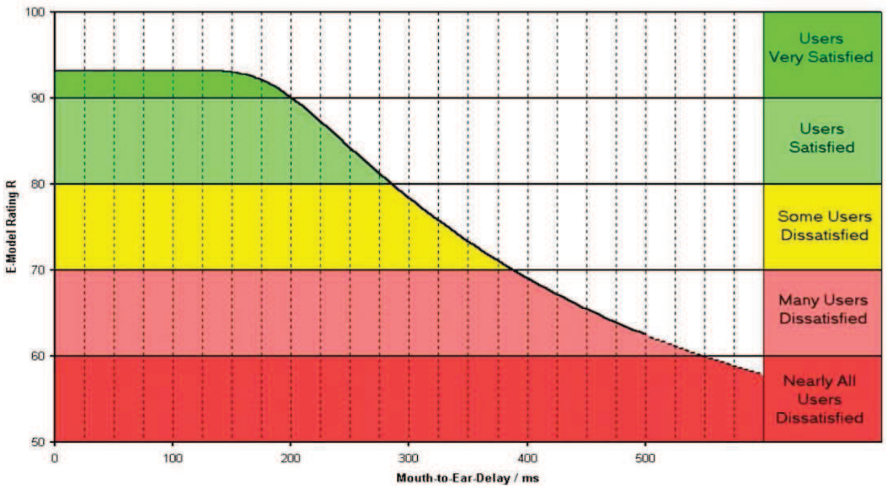

FIGURE 26 E-Model rating as a function of Mouth-to-ear delay [ms], [41]

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 helds in the jitter buffer for some time 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 the 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 2.11.2.

4.3 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)* [33], which has been standardized by *Internet Engineering Task Force (IETF)*. Transport protocols on the another hand are needed for the actual media transport. Generally, IETF's *Real Time Transport Protocol (RTP)* [35] 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.

4.3.1 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, [33], 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 underlaying 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, [34], which is carried by the SIP describes the session details such as:

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

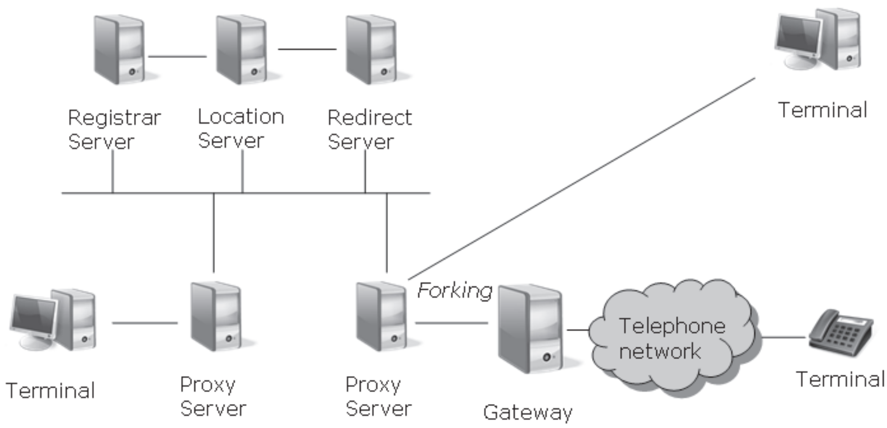

FIGURE 27 SIP Architecture

3. Which media are included in the session,

4. Information needed to receive those media (addresses, ports, formats, etc.).

Architecture of SIP, illustrated in Fig. 27, 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 the 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.

4.3.2 Real-time Transport Protocol

Real-time transport protocol [35] 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 Con-*

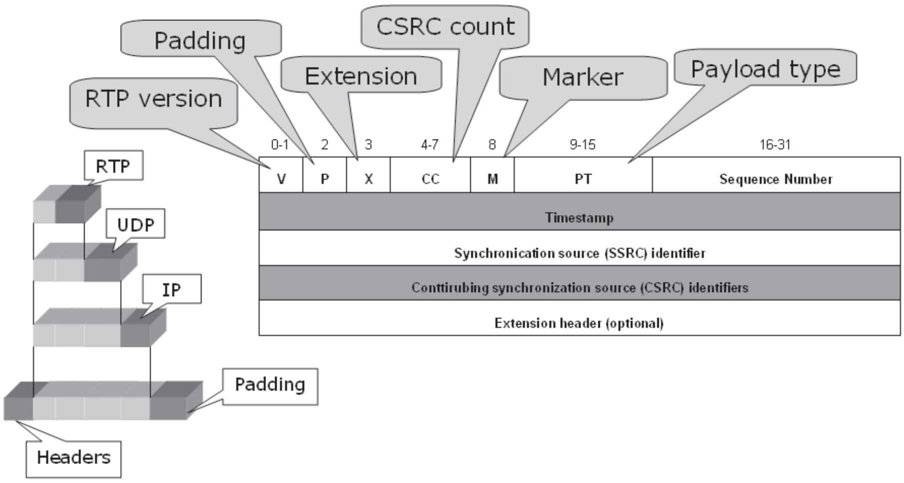

FIGURE 28 RTP Frame Structure

trol Protocol (RTCP) which allows monitoring of the data delivery and provides minimal control and identification functionality.

The RTP frame structure illustrated in Fig. 28 in 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. [36], [37] and [38].
- 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 underlaying 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, [35], 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.

4.3.3 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 [39] and *Robust Header Compression (ROCH)* [40], 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 misordered 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 [40]. Robust header compression introduced new compression mechanisms with the primary objective to achieve the combination of robustness against packet loss and maximal compression efficiency.

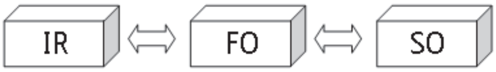

FIGURE 29 ROCH state transitions

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 Fig. 29.

According to [40], 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.

4.4 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 [11]. 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 Fig. 30. 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 [11] and [22].

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 Fig. 31. 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 Fig. 31 illustrates the core of IMS comprises of two main nodes: the

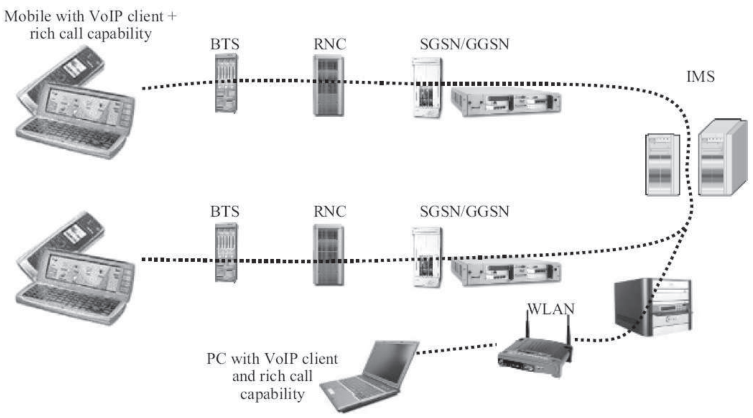

FIGURE 30 Example of IMS system, [27]

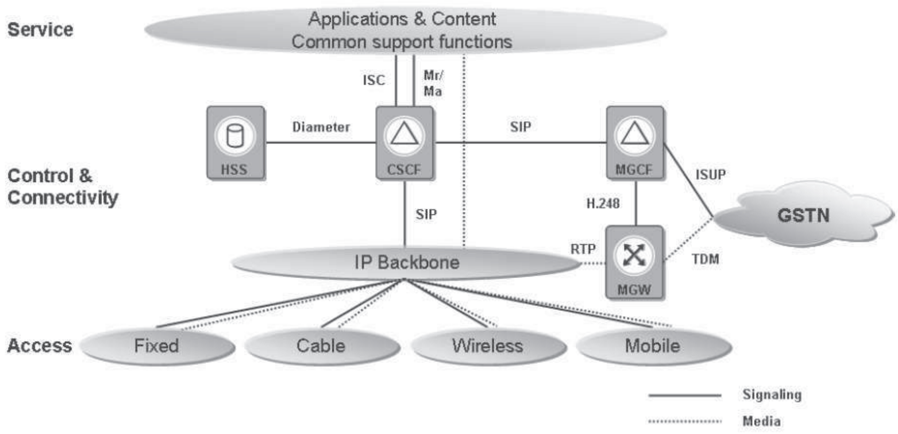

FIGURE 31 IMS Architecture overview, [22]

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 Fig. 32 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

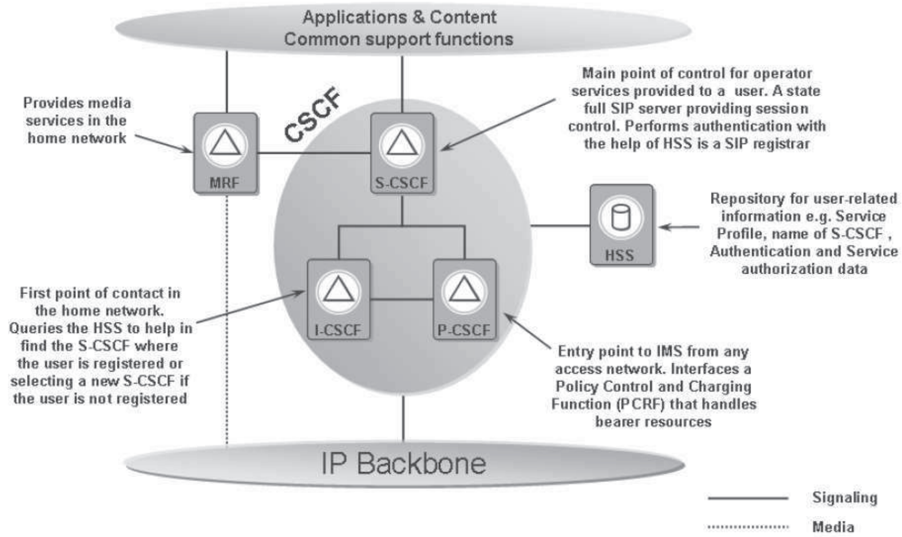

FIGURE 32 Common IMS core overview, [22]

for authenticating the IMS subscriber in the core network and maintains the IMS 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.

5 ACHIEVED RESULTS

The purpose of this chapter is to first provide understanding of the used research tool and research scenario. After providing basic understanding of those issues this chapter presents analysis on the basis of achieved results.

5.1 Research Tool

The research tool used to perform studies regarding this thesis is a comprehensive dynamic WCDMA/HSPA radio network simulator. Simulator is designed for studying RRM algorithms in order to obtain capacity and coverage estimates, to support standardization work and network planning by optimizing and tuning the radio network parameters. Simulation tool enables detailed simulation of users in multiple cells with realistic call generation, propagation, fading and mobility models (adopted from [13]).

There are at least two factors advocating the usage of dynamic system simulations: on the one hand, the complexity of the dynamic 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, fully dynamic system level simulations where user mobility, RRM functionalities and their inter-actions are taken explicitly into account are essential to the performance evaluations. Moreover as the used simulator is highly configurable with parameters for network traffic, mobile terminals, network configuration, RRM algorithms and physical layer in addition to vast expandability possibilities it truly is an indispensable tool for studying radio network performance and possible enhancements.

Implementation of the used simulator itself is a compromise between building an accurate model of a WCDMA/HSPA network and keeping manageable computing efforts. In order to keep computing efforts on reasonable levels and to reduce the complexity of the simulator, simulator has 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. Then produced link level data of receiver *Frame Error Rate (FER)* (or *Bit Error Rate (BER)*) performance is used through a so called *Actual Value Interface (AVI)* in the system level simulator as presented in [32].

Simulation flow includes three fundamental phases:

1. Setup
2. Simulation
3. Destruct

During 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 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 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 is reached. In the end, final statistics will be saved and then the simulation is finished by destroying all created elements, referred as third phase.

The modeling and reliability analysis of the used dynamic system simulator has been presented earlier in many international articles and conferences, see e.g. [31] and [57], as well as in doctoral theses such as [45], [42] and [30]. The reliability analysis for this thesis is addressed (mainly) in the Appendix 1 at the end of this thesis. The purpose of the following sections is to address modeling issues from the perspective of MBMS and VoIP.

5.1.1 MBMS Modeling

MBMS sessions and subscribers

Simulator supports static and dynamic call setup processes for MBMS 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.

Also, from the perspective of MBMS sessions, two alternatives exist. 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.

MBMS transmission and reception

For each sector MTCH is created and it is 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.

5.1.2 VoIP Modeling

VoIP traffic

The VoIP traffic model that is assumed in this thesis is illustrated in Fig. 33. The main features of the model are:

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

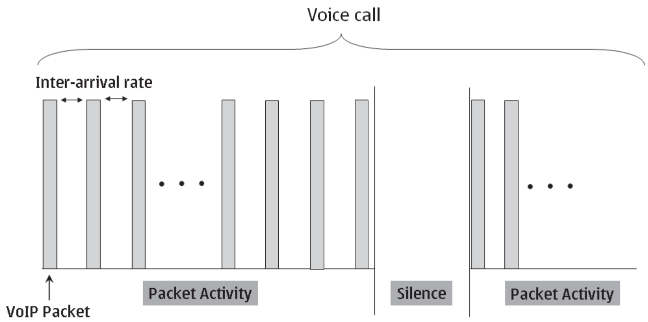

FIGURE 33 Example of VoIP traffic model

- 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 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 Fig. 33. During silent intervals SID frames are sent in order to maintain the connection. SID frames contain information needed to generate background (comfort) noise. They are sent in the beginning of the silent period and every 160 ms during the silence.

VoIP calls

In order to be able to evaluate how many users can be served in a cell before QoS criteria is exceed 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 in terms of VoIP can be interpolated. Users are placed uniforminly around the simulation area.

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 raised in any point of the call, the call is counted to be in outage. In addition, a VoIP call can be dropped if the quality of the transmission drops significantly. Dropping of VoIP calls is based

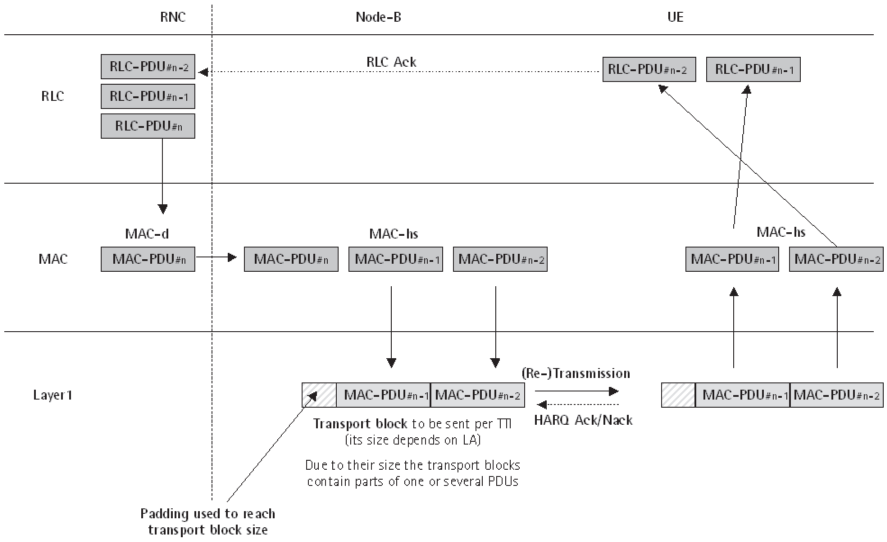

FIGURE 34 Data flow for HSDPA

on similar procedure as with outage but the parameters are different. Finally, the overall outage percentage and moreover system capacity is calculated on the basis of total number of successful and unsuccessful (incl. dropped) calls.

Regarding HSDPA, VoIP call QoS based on outage criteria is monitored in the MAC-hs after possible reordering of packets, see Fig. 34 for reference. Packets can arrive out-of-order if for instance the initial transmission for the transport block does not succeed and thus it requires HARQ retransmission. The time how long UE waits for packets to arrive is set with a parameter 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 as Fig. 35 illustrates. The delay between Node B and RNC is not explicitly modeled but it is taken into account in the delay budget. Another way to monitor the QoS in uplink is through noise rise generated by the users in the system. According to that criteria, the system can support as many of VoIP users per cell that can be served so that NR remains on average below certain level. Threshold of 6 dB is generally considered as a reasonable threshold for NR [16], also see Section 2.11.5 for more information. In this thesis a worst-case scenario regarding the NR criteria is evaluated as the requirement is set so that the system must stay 99 % of time below the NR threshold.

5.2 Simulation Scenario

The studies presented in this thesis are based on two different scenarios. Mainly simulations are conducted in a so called wrap-around macro cell scenario with 21

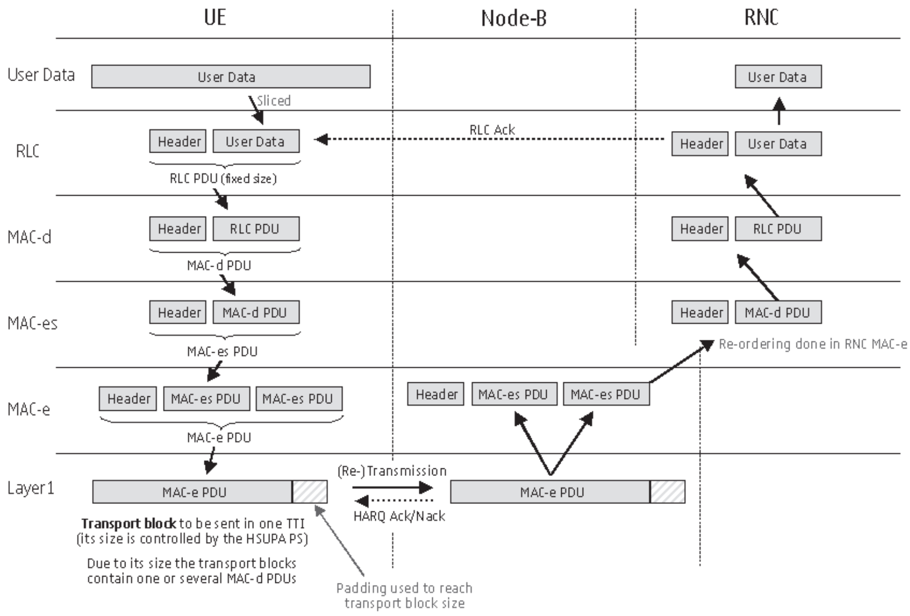

FIGURE 35 Data flow for HSUPA

cells and 7 base stations. Fig. 36 depicts the cellular structure of that wrap-around scenario. The purpose of wrap-around is to model the interference correctly also for outer cell by limiting the UE mobility around the center cells, but the cell transmissions are replicated outside the mobility area to offer more realistic interference situation for every UE in the network. 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. VoIP over HSDPA simulations, on the another hand, were initially based on scenario with 18 cells and 9 base stations without wrap-around. Latter scenario is illustrated in Fig. 37. However, a part of the results included to those early studies with VoIP over HSDPA presented in article **[PV]** were re-simulated for this thesis in a wrap-around scenario to exclude e.g. the impact of border-effects.

The used power delay profiles are selected in coherence with the ITU recommendations i.e. *Vehicular A (VehA)* and *Pedestrian A (PedA)* channel models are used. The power profiles are modified from the original ITU power delay profiles so that the delay between paths is at least one chip-time, [13]. Average path powers in Pedestrian A channel are in decibels [-0.2, -13.5] and in Vehicular A channel [-3.1, -5.0, -10.4, -13.4, -13.9, -20.4]. The delays of the channel profiles are [0, 1] slots for PedA and [0, 1, 3, 4, 7, 10] slots for VehA.

The main simulation parameters for MBMS and VoIP simulations are presented in the Appendix 2 at the end of this thesis and also in conjunction with included articles. In the Appendix, Table 15 presents simulation parameters for MBMS simulations. Table 16 sums up common VoIP parameters used for VoIP studies and Tables 18 and 17 present HSDPA and HSUPA assumptions, respec-

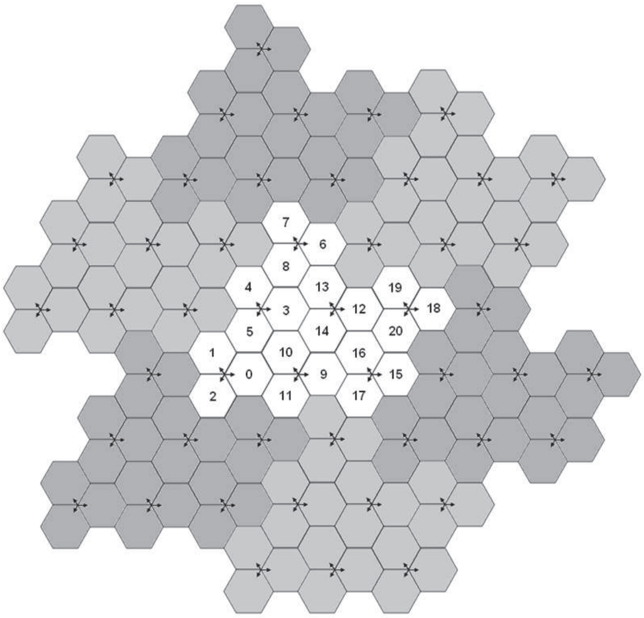

FIGURE 36 Simulation scenario with 21 cells and wrap-around

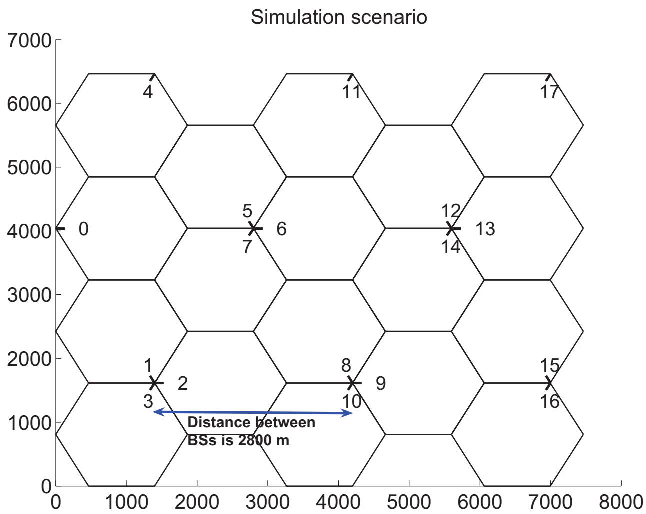

FIGURE 37 Simulation scenario with 18 cells and no wrap-around

tively.

5.3 Simulation Result Analysis

The object of this thesis is to first analyze MBMS performance in WCDMA networks and after that address how Voice over IP can perform in HSPA evolution of WCDMA networks. Both MBMS and VoIP are key technologies when packet based cellular networks are considered. However, by introducing these services into the wireless cellular networks, new range of challenges can be encountered. The purpose of this thesis is to address some of those challenges, especially regarding user's mobility in the network. The following sections present the main findings of this thesis.

5.3.1 Macro diversity combining impact to the MBMS performance

Macro diversity and how it impacts to the MBMS p-t-m performance in WCDMA networks was presented in included articles **[PI]** and **[PIII]**. Article **[PII]** addressed the performance of MBMS together with macro diversity combining and receive diversity.

Studies indicated that macro diversity combining can improve MBMS performance significantly. By adding merely one link into the combining set roughly 5 dB gain with soft combining and 3 dB gain with selective combining (in Vehicular A channel) can be acquired when compared to situation without combining. Moreover, the results showed that even if MBMS terminals would be able to utilize receive diversity macro diversity combining still brought quite significant gain to the performance. Gain acquired from macro diversity was found to be depended on the amount of links in the combining set which can be controlled in some sense by adjusting combining thresholds. Studies demonstrated that soft combining provides more gain than selective combining due to fact that soft combining benefits also from poor links whereas selective combining benefits only from good links.

5.3.2 Impact of Receive Diversity to MBMS performance

MBMS performance studies in WCDMA networks were extended in articles **[PII]** and **[PIII]** to cover the situation where terminals would be deployed with two receive antennas instead of 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 showed that receive diversity can bring substantial benefit for MBMS transmissions. The gain of 2Rx over 1Rx was roughly 4 dB depending on the other simulation parameters. Moreover, as mentioned already before, even though receive diversity itself brings quite significant gain it was also shown that by utilizing macro diversity MBMS p-t-m performance can be enhanced even more.

5.3.3 MBMS with Transmit Diversity

Transmit diversity is also considered as one way to improve the transmission capacity. It exploits the principle of providing the receiver with multiple faded replicas of the same information-bearing signal. The impact of transmit diversity with MBMS was studied in article **[PIII]**.

The results of that study showed that transmit diversity narrows E_s/N_0 distribution and provides slight improvement to lower tail of E_s/N_0 distribution with and without soft combining. In terms of coverage this slight improvement leads to around 2 dB gain without combining. The gain was reduced to less than 1.5 dB with soft combining due to the presence of additional source for diversity.

5.3.4 Battery saving opportunities with MBMS 2Rx terminals

When network is dimensioned to 1Rx and the system contains also 2Rx terminals power savings opportunities can in theory be accomplished by switching off additional receiver branches when radio conditions allow. Possibilities for power saving opportunities was addressed in article **[PIII]**.

The study demonstrated that it is indeed possible to achieve very similar coverage levels with MBMS in situations where UE is either allowed to reconfigure between 1Rx Rake and 2Rx Rake receiver or to use only 2Rx Rake receiver. To have this similarity in coverage levels one only needs to have suitable switching thresholds and C/I filtering windows in use. Apart from similar network performance, antenna usage statistics indicated that there is possibility to spend a significant amount of time using the 1Rx Rake configuration. This phenomenon suggests that there are definite power saving opportunities relative to a UE using 2Rx Rake continuously.

5.3.5 VoIP over HSUPA performance

Uplink is considered most often to limit the overall coverage of services due to fact that uplink direction is dominated by small handheld power limited terminals and high interference levels. Moreover, if speech service is considered then in 'worst-case-scenario' the cell can be very highly loaded with terminals and if mobility is also taken into account more challenges can be faced. Packet based voice service i.e. Voice over IP performance was studied over HSUPA in article **[PIV]**. The article included studies with various handover delays, active set sizes and user velocities. Additionally, the impact of SID frames was studied.

The study indicated that the system performance can be highly affected by user's mobility. Handover delay proved to have rather minor impact on the performance with low user velocities as the performance loss remained within 10 % provided that the delay stayed within reasonable limits. The difference between a single link and two-way soft handover performance was quite noticeable. However, limiting the active set size was observed to reduce the VoIP capacity only slightly as long as at least two links were allowed in the active set. In addition, silence indication frames specified for VoIP caused only a slight additional performance loss in mobility scenarios.

5.3.6 VoIP over HSDPA performance

With HSDPA only hard handover is supported which can result into excessive packet losses and delays if handover is not executed in time or on the another hand is 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 simulation results indicated moderate sensitivity of VoIP capacity to the UE velocity. Very high velocities lead to more noticeable capacity loss for VoIP over HSDPA. However, tuned handover parametrization for different velocities might provide better performance but it would require some mechanisms for UE speed detection. The results showed also that handover delays started to have significant effect 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 was concluded not to pose problems.

Quite recently VoIP over HSDPA studies were re-opened to study e.g. power consumption related matters. As quite considerable amount of time had past 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. 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 Fig. 38 in terms of user velocity. As that figure shows the absolute capacity numbers are somewhat improved with the updated simulator version, however depending heavily on the user velocity and thus of the handover performance which was revised in the new version. For instance, with 3 km/h the capacity numbers are still in the same order the difference being only couple of UEs/cell but for higher velocities the VoIP over HSDPA capacity is higher than before. Therefore, also the relative drop caused by user velocity is somewhat lower than 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 Fig. 39. As that figure shows, again the low velocities are practically unchanged but with higher velocities the absolute numbers are higher with the new version. However, the relative impact of the delay stays in the same order as before and thus the conclusions apply on that matter as well.

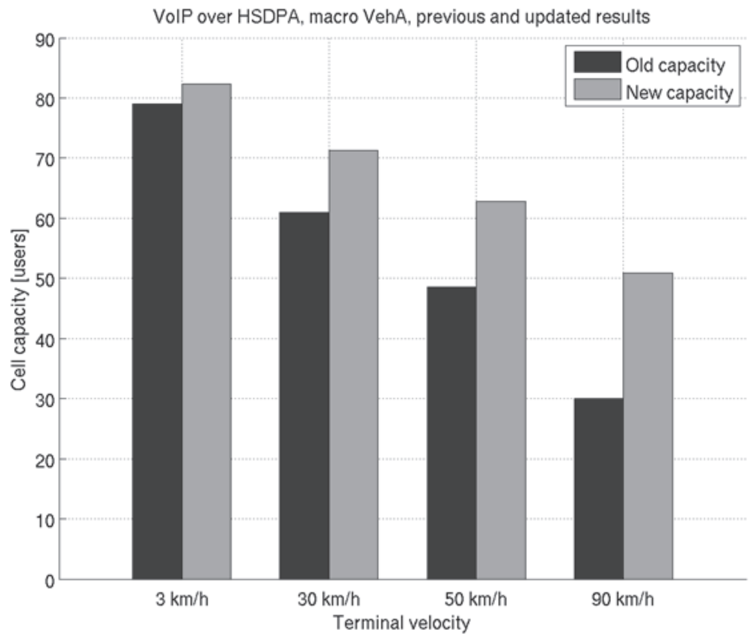

FIGURE 38 VoIP over HSDPA 95 % outage level comparison with different velocities

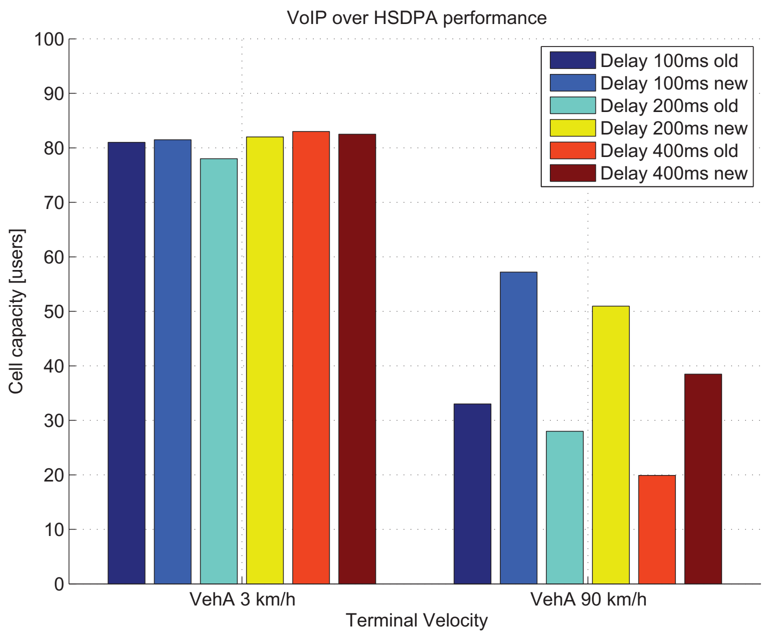

FIGURE 39 VoIP over HSDPA 95 % outage level comparison with different delays

6 CONCLUSIONS

The purpose of this thesis was to address the performance of MBMS VoIP services in WCDMA/HSPA networks. This thesis elaborated that MBMS and VoIP services have already gained vast popularity in the fixed networks and thus it is very likely that they will also be key concepts in wireless cellular networks in the near future. However, from practical perspective, wireless network sets very strict environment for this kind of services. Thus, this thesis aimed to evaluate the need of radio resources, possible enhancements and vast number of practical network parameters so that cellular network and these services operate in optimal manner in different situations. Special attention was be paid on the mobility aspect of both of those services. This thesis evaluated the performance with help of fully dynamic system simulator where user mobility, fading, propagation and RRM functionalities are explicitly taken into account.

This thesis showed that MBMS can benefit significantly from macro diversity combining. Macro diversity i.e. combining transmissions from multiple sources is likely option for MBMS performance improvement as MBMS is expected to be available through large parts of the network. In addition to macro diversity, also receive diversity was found to be very efficient method to improve the performance of MBMS. However, mobile terminals with two or more receive antennas are not likely be among the first devises to support MBMS but start to emerge slowly which means that the networks are likely to be dimensioned to 1Rx terminals. Thus, this thesis showed that when the network is optimized for 1Rx then 2Rx terminals can have power saving opportunities and still have high QoS by turning off the other receive antenna in good channel conditions.

The results presented by this thesis also showed the sensitivity of VoIP to handover/active set update delays. Delays were found to be rather small if the user velocity is low. However, when the terminals are moving at high speeds then the delays start to have an impact. This was found to be caused by the fact that with high speed the serving link is deteriorating much faster and thus the handover procedure should be faster. With the respect of VoIP over enhanced uplink it was concluded that the cell capacity is not limited by uplink as long as soft handover is supported. Finally, this thesis showed that VoIP capacity can suffer quite a bit if all of the terminals would be moving at very high speeds. Thus, at the end of the day these issues needs to be taken into account e.g. in the network planning to avoid some of the potential problems this thesis raised.

YHTEENVETO (FINNISH SUMMARY)

Ensimmäiset kolmannen sukupolven langattomat matkaviestinverkot otettiin kaupalliseen käyttöön vuonna 2002. Nämä verkot pohjautuvat pääasiassa WCDMA-tekniikkaan (engl. Wideband Code Division Multiple Access), joka mahdollistaa käytössä olevan taajuuden tehokkaamman käytön. Siinä missä toisen sukupolven (2G) verkot tarjosivat pääasiassa mahdollisuuden puheliikenteelle, 3G-verkot mahdollistivat ensimmäistä kertaa innovatiivisten pakettipohjaisten palveluiden tarjoamisen matkaviestinverkoissa. Mahdollisten palveluiden kirjo laajeni entisestään, kun kolmannen sukupolven verkkojen suorituskykyyn esiteltiin laajennukset, jotka tarjoavat entistä suurempia tiedonsiirtonopeuksia ja pienempiä vasteaikoja. Nämä evoluutiot kulkevat nimellä HSDPA (engl. High Speed Downlink Packet Access) ja HSUPA (engl. High Speed Uplink Packet Access) tai jos viitataan yhteisesti molempiin HSPA (engl. High Speed Packet Access). Odotetusti WCDMA- ja HSPA-verkkojen suosio on kasvanut lähes räjähdysmäisesti, sillä vuoden 2004 lopussa WCDMA-tilaajia oli 17 miljoonaa, mutta jo vuoden 2008 ensimmäisen neljänneksen aikana tilaajia oli jo yli 250 miljoonaa. Tulevaisuudessa käyttäjien määrän uskotaan kasvavan entisestään, sillä kasvua vauhdittaa alati laajeneva kirjo innovatiivisia palveluita aina videokuvan siirrosta ja tv-lähetyksistä erilaisten interaktiivisten palveluiden tarjoamiseen.

Tässä lisensiaatintyössä tutkittiin erityisesti pakettiliikenteeseen perustuvan puheliikenteen (Voice over IP) toimintaa HSPA-verkoissa sekä ryhmälähetyksiä käsittelevän MBMS-tekniikan (engl. Multimedia Broadcast Multicast Service) suorituskyvyn ja sen parannuksien analysointia WCDMA-verkoissa. Matkaviestinverkkojen puolella tällaiset tekniikat tarjoavat myös uusia näkökulmia verrattuna kiinteisiin verkkoihin. Esimerkiksi VoIP-puheluiden ja sen mahdollistamien lisäpalveluiden tuominen osaksi matkaviestinverkkoja tarjoaa operaattoreille mahdollisuuden aivan uudenlaisiin tulovirtoihin sekä mittaviin kustannussäästöihin siinä valossa, ettei tällöin enää tarvittaisi erillistä piirikytkentäistä verkkoa puheelle. Kuitenkin samalla langaton matkaviestinverkko asettaa näille palveluille hyvin kriittiset toimintaolosuhteet, jonka tuomiin haasteisiin tämä lisensiaatintyö pyrkii vastaamaan.

Tutkimusmenetelmänä käytettiin ohjelmistopohjaista radioverkkosimulaattoria, jolla pyritään mallintamaan reaalimaailman matkaviestinjärjestelmää hyvin tarkasti. Simulaattorissa käytetyt mallit ovat yleisesti hyväksyttyjä ja tunnettuja. Simulaattorin mallinnusta ja aikaisempia tuloksia on esitelty lukuisissa kansainvälisissä tieteellisissä konferensseissa ja lisäksi mm. kolmessa väitöskirjassa.

Yksityiskohtaisemmin sanottuna tutkimustyö kohdistui MBMS:n suorituskyvyn analysointiin WCDMA-verkoissa mahdollisten toimintaa parantavien tekniikoiden kanssa. Suorituskykyä arvioitiin pääasiassa ensimmäisten 3G-laitteiden näkökulmasta, ja täten HSPA-evoluutioiden tuomia näkökulmia ei erikseen käsitelty. Makro-, lähetys- ja vastaanotindiversiteetti olivat niiden tekniikoiden joukossa, jotka sisältyivät tutkimuksen liikkuvuuteen liittyvien MBMS-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. Työn tuloksien perusteella makrodiversiteetti on erinomainen keino parantaa MBMS:n suorituskykyä. Vastaanotindiversiteetti lisäsi myös MBMS-käyttäjien tyytyväisyyttä merkittävästi makrodiversiteetin kanssa tai ilman.

VoIP-tekniikan suorituskyvyn osalta erityisenä kiinnostuksen kohteena tutkimuksessa oli VoIP-lähetyksien käyttäytyminen yhteisvastuunvaihtojen (engl. handover) yhteydessä. VoIP-liikenteen käyttäytyminen siirryttäessä tukiaseman/-asemien peittoalueelta toiselle, eli yhteisvastuuvaihtojen yhteydessä, on erityistä huomiota vaativaa, sillä VoIP on hyvin kriittinen kaikille ylimääräisille viiveille sen reaaliaikaisuuden vuoksi. Tässä lisensiaatintyössä esitettyjen tuloksien perusteella voidaan todeta, että pienillä nopeuksilla kohtuulliset ylimääräiset viiveet yhteisvastuuvaihtojen aikana eivät tuota ongelmia. Kuitenkin suuremmilla nopeuksilla yhteisvastuuvaihdot pitäisi toteutua merkittävästi nopeammin, jotta suorituskyky säilyy hyvänä. Yleisesti ottaen nopeuden vaikutus VoIP-tekniikan suorituskykyyn todettiin olevan suhteellisen merkittävä, joten tämä on huomioitava esimerkiksi verkkoja suunniteltaessa.

APPENDIX 1 RELIABILITY ANALYSIS

Dynamic system simulator, which was used as a research tool in this thesis was presented briefly in Section 5.1 and, for instance, in [30], [42] and [45] the tool is verified and presented in very detailed level. This appendix is aimed to give the statistical confidence on simulation results presented in this thesis.

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 MBMS simulations

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 level which is acquired by interpolating these test cases. Main simulation parameters for these test cases are presented in Table 2.

The confidence interval can be used to evaluate the statistical confidence of simulation results. Tables 3, 4 and 5 present results acquired from $n = 31$ simulation runs with different random generator seeds.

Confidence intervals of 90 %, 95 % and 99 % for the reliability simulation results presented above are presented in Table 37, Table 38 and Table 39. As those tables show abnormalities are very minimal in every case, which confirms that MBMS results presented in this thesis are reliable.

TABLE 2 Simulation parameters for MBMS reliability analysis

Simulation parameter	Value(s)
Combining scheme	Soft combining
Receiver	Rake 1Rx
Channel models	Modified ITU Vehicular A
MTCH bit rate	128 kbps
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
Network synchronization	Ideal

APPENDIX 1.2 VoIP simulations

The purpose of this section is to provide statistical confidence analysis for Voice over IP simulations presented this thesis. The scenario for confidence analysis is 21 cell wrap-around macro cell layout presented in Section 5.2. In that scenario VoIP over HSDPA performance is studied with various seeds and user amounts in Vehicalar A channel. Mainly parameters for this analysis are presented in Table 9 and other than that simulation parameters follow the same set as defined in Table 16 and 18. In addition to varying the seeds a few test cases were run with different simulation lengths to verify the statistical confidence of chosen simulation length of 200 seconds.

The simulation results for this analysis are produced with more advanced version of the simulator that was used to conduct studies presented in included article **[PV]**. Refer to the Section 5.3.6 for more details about the updated simulator version.

Analysis from these cases is focused on VoIP outage as it is considered as the key performance indicator in this thesis. The performance is evaluated both on the outage level with certain loads as well as according to the 95 % coverage/outage level which is acquired by interpolating these test cases.

As mentioned earlier the confidence interval can be used to evaluate the statistical confidence of simulation results. Tables 10 and 11 present results acquired from $n = 31$ simulation runs with different random generator seeds. The impact of simulation length is additionally presented in Table 12 which shows only minor differences between interpolated capacity numbers. Thus, evaluation with default simulation length should be adequate.

Confidence intervals of 90 %, 95 % and 99 % for the reliability simulations

TABLE 3 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 4 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 5 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 6 Statistical confidence for 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 7 Statistical confidence for 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 8 Statistical confidence for 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 %

TABLE 9 Main simulation parameters for statistical analysis of VoIP simulations

Simulation parameter	Value(s)
Scenario	Wrap-around, 21 hexagonal cells
Cell radius	933 m
Channel profile	Vehicular A
Packet scheduling algorithm	VoIP optimized proportional fair
UE velocity	30 km/h
HSDPA HO delay	200 ms
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

TABLE 10 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 11 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 12 VoIP over HSDPA 95 % outage level with different simulation lengths

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

TABLE 13 Statistical confidence for VoIP over HSDPA outage level

	1400 UEs/cell, Outage [%]	1600 UEs/cell, Outage [%]
MEAN	1.68	9.47
STANDARD DEVIATION	0.33191	0.97242
CONFIDENCE INTERVAL, 90 %	± 5.85 %	± 3.03 %
CONFIDENCE INTERVAL, 95 %	± 6.97 %	± 3.61 %
CONFIDENCE INTERVAL, 99 %	± 9.16 %	± 4.75 %

TABLE 14 Statistical confidence for 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 %

presented above are gathered to Tables 13 and 14. As those tables show abnormalities are very minimal especially when interpolated capacity numbers are considered. Thus, this leads us to believe that VoIP results presented in this thesis are reliable.

APPENDIX 2 SIMULATION PARAMETERS

The purpose of this appendix is to present briefly main simulation parameters for MBMS and VoIP studies. Table 15 presents simulation parameters for MBMS simulations. Table 16 sums up common VoIP parameters used for VoIP studies and Tables 18 and 17 present HSDPA and HSUPA assumptions, respectively.

TABLE 15 Main simulation parameters for MBMS studies

Simulation parameter	Value(s)
Scenario	Wrap-around, 21 hexagonal cells
Cell radius	933 m
UE velocity	3 km/h
Combining scheme	Soft and selective combining
Receiver	Rake 1Rx and 2Rx
Pathloss model	Modified Okumura-Hata
Channel models	Modified ITU Vehicular A and Pedestrian A
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
MTCH bit rate	64, 128 and 256 kbps
MBMS session length	20 s
Number of sessions	18
Simulation time	360 s
Number of UEs per session	600
TTI length	80 ms for 64 kbps, 40 ms for 128 and 256 kbps
S-CCPCH E_c/I_{or}	From -18 dB to -1 dB
Combining set size	1, 2 and 3
Add window	[2, 4], [4, 6] and [10,12] dB
Add timer	40, 100, 200 and 480 ms
Drop window	6 dB
Drop timer	640 ms
Replace window	2 dB
Replace timer	40, 100, 200 and 480 ms
Combining set update delay	0, 1, 5 and 10 TTIs
Network synchronization	Ideal and non-synchronized

TABLE 16 Common simulation parameters for VoIP over HSPA simulations

Simulation parameter	Value(s)
VoIP Packet size	38 bytes
Average VoIP call length	60 s
Average activity period	3.0 s
Average silence period	3.0 s
Probability to start call in silence	0.5
Packet inter-arrival rate	20 ms
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

TABLE 17 Main simulation parameters for VoIP over HSUPA studies

Scenario	Wrap-around, 21 hexagonal cells
Cell radius	933 m
Pathloss model	Modified Okumura-Hata
Channel profiles	Pedestrian A and Vehicular A
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
TTI length	10 ms
Bit rate	32 kbps
HSUPA Scheduling mode	NST
UE velocity	3, 30 and 120 km/h
Maximum active set size	1, 2 and 3
Add, replace, drop window	[2 dB, 2 dB, 4 dB]
Max. number of HARQ transmissions	2
SID packet size	15 bytes
Simulation time	360 s

TABLE 18 Main simulation parameters for VoIP over HSDPA studies

Cellular layout	18 hexagonal macro cells
Cell radius	933 m
Receiver	Rake 1Rx
Pathloss model	Modified Okumura-Hata
Channel profile	Vehicular A
Slow fading deviation	8 dB
Slow fading correlation distance	50 m
TTI length	2 ms
Packet scheduling algorithm	VoIP optimized proportional fair
UE velocities	3, 30, 50 and 90 km/h
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	100, 200 and 400 ms
Max. number of HARQ transmissions	3
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
SID	disabled
Simulation time	200 s

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