University of Glamorgan

A Simulation Framework for Service Continuity over Multi Access Wireless Networks

by

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Declaration of Authorship

I, Belal Saleh Abuhaija, declare that this thesis titled, 'A Simulation Framework for Service Continuity over Multi Access Wireless Networks' and the work presented in it are my own. I can confirm that

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Abstract

Mobile communication systems have continued to evolve by the release of new standards for HSPA and the release of new standards for LTE in release 8 and release 9. The new releases aim at providing higher data rates to accommodate the envisioned services of 3GPP in voice, data, image transmission, video, multimedia service and broadband services. Catering for a wide variety of services to satisfy the demands imposed on mobile networks by the user diversity and demanding applications, the air interface has been identified as the major bottleneck in the mobile networks. Network planning engineers and operators are deploying the new air interfaces in the same cell sites, which gives rise to several internetworking issues among the different air interfaces from radio resources management to service call continuity issues due to user's mobility and changing point of attachment. Deployment of different air interfaces coupled with traffic diversity requests further complicates the managements of the mobile networks.

One of The main objectives of this research is to propose and evaluate solutions that address the internetworking of the different radio air interfaces in proposing a call admission control algorithm that can utilize different air interfaces capabilities and to determine which traffic types are better suited to be serviced by an interface. The proposed algorithm will consider the availability of the interface, the load of the network and the user equipment capabilities. Another main objective of this research is to propose and evaluate solutions that address changing the point of attachment of the users due to mobility in the form of handover algorithm. The proposed algorithm will consider the coverage of the node, direction and speed of the user, the network load, the air interface availability and the user equipment capabilities. Another main objective of this research is to design and implement a simulation system which includes all 3GPP standardized technologies. The simulation tool was designed as a discrete event simulation (DES) system which includes all the standardized air interface technologies and services. The simulation tool was designed using Visual C# to take advantage of the object oriented capabilities of the Windows environment and libraries. The simulation tool was essential in evaluating the proposed algorithms in the first two objectives.

Keywords: *Call Admission Control, Vertical Handover, Horizontal Handover, DES, GSM/GPRS, UMTS, HSDPA, LTE.*

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In this life, we as human beings go through experiences by communications and interactions with other people, which influence our life in a certain way as we are exposed to different personalities, mentalities and attitudes. Some people are doing favors to others, expecting at some time in the future the favors are returned, those are the majority. It is rarely to find somebody who will do you a favor after another and expecting nothing in return and that is my director of studies Prof. Khalid Al-Begain. Prof. Al-Begain was not only instrumental to the success of this research project through his guidance, support and mentoring but also he was and still looked at for spirit uplifting when bad days loom and for that I will ever be in debt.
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Abbreviations

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
ACIR	Adjacent Channel Interference Ratio
ACLR	Adjacent Channel Leakage Power Ratio
AMC	Adaptive Modulation and Coding
AMPS	Advance Mobile Phone Service
AMR	Adaptive Multi-Rate
ARQ	Automatic Repeat Request
AWGN	Adaptive White Gaussian Noise
BE	Best Effort
BER	Bit Error Rate
BLER	Block Error Rate
BSC	Base Station Controller
BSS	Base Station Subsystem
BTS	Base Transceiver Station
CAC	Call Admission Control
CBR	Constant Bit Rate
CIR	Carrier-to-Interference Ratio
CN	Core Network
CPICH	Common Pilot Channel
CQI	Channel Quality Indicator
CRRM	Common Radio Resources Management
CS	Circuit Switched
DES	Discrete Event Simulation
DP	Dynamic Partition
DRNC	Drift Radio Network Controller
EDGE	Enhanced Data Rates for GSM Evolution
EPC	Evolved Packet Core
EPS	Evolved Packet System
ETSI	European Telecommunications standards Institute
E-UTRAN	Evolved UTRAN
FDMA	Frequency Division Multiple Access

FTP	File Transfer Protocol
GERAN	GSM EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GMSC	Gateway Mobile Switching Centre
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSCSD	High-Speed Circuit-Switched Data
HSDPA	High Speed Downlink Packet Access
HS-DPCCH	High Speed Physical Control Channel
HS-DSCH	High Speed Downlink Shared Channel
HSPA	High Speed Packet Access
HS-PDSCH	High Speed Physical Downlink Shared Channel
HSS	Home Subscriber Server
HS-SCCH	High Speed Shared Control Channel
HSUPA	High Speed Uplink Packet Access
LAN	Local Area Network
LB	Load Balancing
LFG	Limited Fractional Guard channel
LTE	Long Term Evolution
MAC-hs	Medium Access Control High Speed
MAPL	Maximum Allowable Propagation Loss
MIMO	Multi Input Multi Output
MME	Mobility Management Entity
MSC	Mobile Switching Centre
NAS	Non Access Stratum
NRT	Non Real Time
OFDMA	Orthogonal Frequency Division Multiple Access
PCEF	Policy and Charging Enforcement Function
PCRF	Policy and Charging Resource Function
PCU	Packet Control Unit
PGW	Packet Data Network Gateway
PS	Packet Switched
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quaternary Phase Shift Keying
RAN	Radio Access Network
RAT	Radio Access Technology
RNC	Radio Network Controller

RNS	Radio Network Subsystem
RRM	Radio Resource Management
RSCP	Received Signal Code Power
RSS	Received Signal Strength
RSSI	Received Signal Strength Indicator
RT	Real Time
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SGW	Serving Gateway
SINR	Signal to Interference-plus-Noise Ratio
SIR	Signal-to-Interference Ratio
SRNC	Serving RNC
TBS	Transport Block Size
TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
TFRC	Transport Format and Resource Combination
TTI	Time Transmission Interval
UBR	Unspecified Bit Rate
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VLR	Visitor Location Register
VoIP	Voice Over IP
WCDMA	Wide Code Division Multiple Access
WLAN	Wireless LAN

Chapter 1

3GPP Radio Access Technologies

1.1 Introduction

The main goal of the mobile cellular technologies was to deliver voice services with quality of service (QoS) comparable to that of the wire line networks, however after the first utilization of a packet data application and the widely success of short messages service (SMS) application, the operators noticed the value of delivering data applications utilizing the infrastructure of cellular networks. Since then the demand for more bandwidth never stopped, fueled by consumer demand and rich multimedia applications. We are more connected than ever with the deployment of mobile terminals and laptops. At the same time, the consumers are demanding that audio, video and data be packaged into one application. The operators are pressured by the consumers to bundle these services together in one new and on demand multimedia services. Several radio access technologies have been standardized or in the process of deployment to meet the consumer need for higher data rates. Mobile networks deployment, planning and optimization have become an important area of research for the network operators.

The first generation of mobile telecommunications was based on analog circuit switched technology and called Advanced Mobile Phone Service (AMPS) as shown in Figure 1.1. The second generation of telecommunications systems was based on digital technology and in 1990 the first phase of Global Mobile for communications (GSM) was standardized by the European Telecommunications Standards Institute (ETSI), however nowadays 3rd Generation Partnership Project (3GPP) standardizes cellular wireless technologies and in 1999 the standard is called Release 99. In 2002 Release 5 was rolled out, which included High Speed Downlink Packet Access (HSDPA), and commercially deployed in 2005; subsequently Release 6 was deployed in 2005 which included standards for High Speed Uplink Packet Access (HSUPA). Further evolution and deployments of standards took place in 2007 which is called Release 7, and 2008 which is called Release 8.

Figure 1.1 shows the evolution of 3GPP Radio access technologies and the data rates they offer.

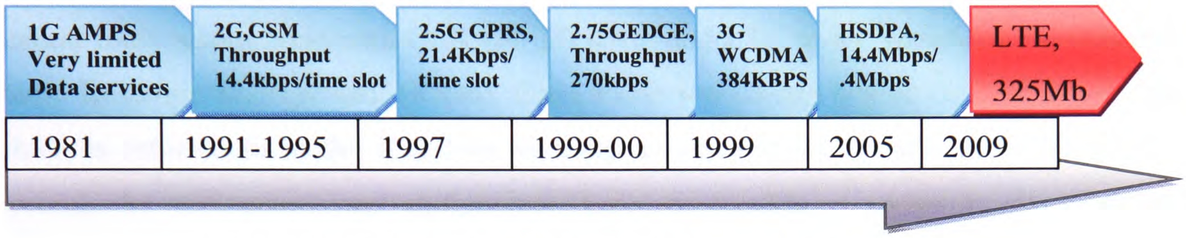

Figure 1.1 Evolution of 3GPP technologies and Peak Data Rates

GSM has evolved to provide low data rates known as General Packet Radio System (GPRS) and called 2.5G. GPRS brought the packet switched service capabilities to the existing GSM circuit switched system with theoretical data rates up to 170Kbps. This in turn has also evolved as discussed in the technology background chapter to become Enhanced Data rates for GSM Evolution (EDGE) or 2.75G. This radio interface is now called GSM/EDGE Radio Access Network (GERAN). GERAN interface capacity theoretically is around 270Kbps, practically it is around 60Kbps [1]. The deployment of 3G increased the capacity up to 384kbps. The evolution of 3G through the deployment of HSDPA which is known as 3.5G in Release 5 of the 3GPP standards has in theory increased the capacity up to 14.4Mbps, while the uplink data peak rate reached 5.7Mbps in Release 6. However, in Release 7 the down link peak data rate is 28Mbps and the uplink peak data rate is 11Mbps with the existence of multiple antenna and 64 Quadrature Amplitude modulations (64QAM). This is referred to in the standards as Universal Mobile Telecommunications Services (UMTS) Radio Access Network (UTRAN), based on Wide Code Division Multiple Access (WCDMA). Both GERAN and UTRAN terminate in the same core network. At the same time, the demand for higher data rates is still on the rise. By the advent of the new standards for Long Term Evolution (LTE), another air interface will be added to the mix. LTE is supposed to be a real broadband all-IP technology that will provide data rates in excess of 325Mbps, however LTE radio interface does not terminate at the same core network elements as GERAN and UTRAN.

Operators are stepping up the deployment of WCDMA and HSDPA cells to provide reliable, ubiquitous and higher data rate coverage. WCDMA and HSDPA cells are sharing the same cell site as GERAN cells. LTE is suppose to share the same cell sites in its early deployments. Such deployment will highlight four issues; first, the existence of several radio access networks GERAN, UTRAN and LTE at the same site complicates Radio Resource Management (RRM). Each Radio technology has its own RRM scheme which is referred to in the literature as multi-access and multi-radio networks [2, 3]. Second, the customers' user equipments (UE) are capable of accessing the different radio networks. Usually the UEs are backward compatible (i.e. the UMTS UE can access GSM/GPRS technology but not the other way around. Third, the interoperability between the different radio access technologies is further complicated by the existence of several services competing for the radio resources. The QoS with four traffic classes such as conversational, streaming, interactive and background services have been standardized [4-6]. Four, providing service continuity with the existence of several radio access networks can add to the radio resource pool but can also complicate resources management.

1.2 System Inter-Operability

In 3GPP standardization up to Release 4, the perception was to have two separate core networks, one for the circuit switched services connected to the Mobile Switching Centre (MSC) and the second for packet switched services connected to Serving GPRS Support Node (SGSN). Most of the core network functionalities of GERAN have been adopted by UMTS in Release 99, and through the introduction of the Iu interface, a functional split between the core network and the Radio Access Network (RAN) has been cemented. The demand to streamline service continuity between the two interfaces by the operators who are deploying both technologies led to the modifications of the base stations in GSM/GPRS and to the adoption of the Iu interface by GERAN. Changes in the core network have enabled the operators to deliver such services through the

GSM/EDGE air interface (GERAN). The new UMTS Terrestrial Radio Access Network (UTRAN) technology has offered new radio access network architecture and new QoS designed to efficiently provide packet switched multimedia services. The new air interface has higher spectral efficiency and bandwidth, and in Release 5 the two air interfaces UTRAN and GERAN have been developed to integrate with one another and to merge as one UMTS multi-radio network [7].

Integrating these two radio air interfaces is a common sense decision as the operator can have versatility and flexibility to offer the customers more options, better service and higher quality. This interoperability can enhance service continuity when users are changing point of attachment because of mobility.

The multi-radio architecture (GERAN/UTRAN) optimization may provide cost efficient service delivery through one network managed by the same QoS architecture [8].

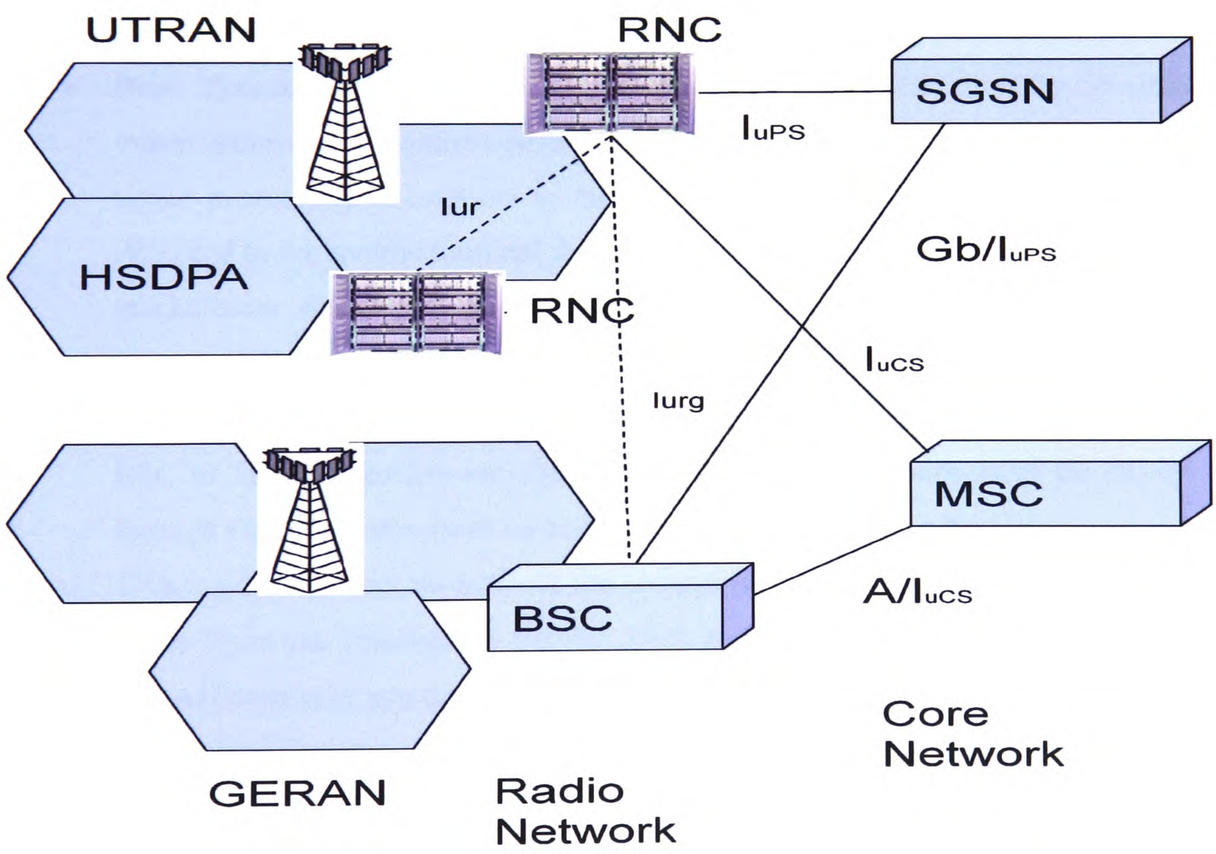

Figure 1.2 Network Architecture overview Release 5

Figure 1.2 components illustrate how GERAN and UTRAN are terminated in the core network. Below is a description of the components and its interfaces in the GERAN:

- **Base Station Subsystem** (BSS): A GSM network consists of many BSSs, each BSS is controlled by base station controller (BSC). BSS is responsible for all the functions necessary to management and maintenance of radio connections to the mobile terminal. BSS also responsible for coding and decoding of voice. BSS is comprised of several BSCs and several base station transceivers(BTSs)
- **Base Station Controller** (BSC): BSC manages the base transceiver station (BTS). It is responsible for radio frequency reservation and handle handover between base transceiver stations. It is also responsible for multiplexing radio channels onto the fixed network connection at the A interface (not shown in Figure 1.2, it will be discussed in the Chapter 2). BSC may control one or more Base Station Transceiver.
- **Base Transceiver Station** (BTS): It comprises all the functionality of radio transmission and reception equipments of the cell like, antennas, amplifiers and signal processing. It connects to the BSC via a dedicated fixed line called the Abis and to the mobile terminal through Um interface. Um interface contains all mechanisms necessary for wireless transmission (Time Division Multiple Access (TDMA) or Frequency Division Multiple Access (FDMA)).
- **Packet Control Unit** (PCU): This hardware equipment is an addition to the BSC to facilitate packet switched data transmission. It connects to the SGSN through Gb connection (will be discussed later in the Chapter 2).

In the UTRAN side of the access network the components are:

- **Radio Network Subsystem** (RNS): RNS consists of one or more RNC. The UTRAN consist of one or more RNSs.
- **Radio Network Controller** (RNC): RNC controls the radio resources of node Bs. RNC is responsible for QoS mapping, scheduling of the dedicated channels, handover control and power control (will be discussed in Chapter 2). RNC can be connected to other RNCs via Iur interface (not shown in Figure 1.2) and can be connected to BSC via Iur-g interface.

- **Node B**: It is responsible for the actual reception and transmission of the radio signals from User Equipments (UE). Node B controls the frequency channels allocated to the node and the output power of the base station. It is connected to RNC through Iu interface and to the UE through Uu interface. Node B is responsible for the inner loop power control as discussed in next chapter and should be under the control of one RNC.

In the core network the components are:

- **Home Location Register** (HLR): It is the database that holds the user's information profile in the home system. The user information profile may include the services allowed, cell access rights, and subscriber location [1].
- **Mobile Switching Center/ Visitor Location Register** (MSC/VLR): The switch that provides Circuit Switched (CS) services for the subscriber in this current location, while the VLR is holding a copy of the subscriber profile.
- **Gateway MSC** (GMSC): The gateway switch that is connected to external CS networks. All egress and ingress connections for the CS services (Domain) have to go through GMSC.
- **Serving GPRS Support Node** (SGSN): The entity that provides Packet Switched (PS) services for the subscriber.
- **Gateway GPRS Support Node** (GGSN): The gateway for all subscriber PS services to/from the current location network.

The addition of SGSN and GGSN in the core network along with the Packet Control Unit (PCU) in GSM base station controller enabled the delivery of the packet switched services to customer. The core network has two domains, CS domain that provides circuit switched services like plain old telephone conversations and PS domain that provides packet switched services like web surfing and emails. Both are connected to UTRAN and GERAN through the Iu-CS interface and Iu-PS interface[1, 8, 19].

High Speed Downlink Packet Access (HSDPA) evolved from 3G through utilising a number of existing technologies. Several techniques have been employed to compensate for the changing link conditions. The main theme is based on link adaptation by modifying the transmission parameters of the system to adapt to the instantaneous transmission conditions. Several properties of WCDMA have been adapted to enable

HSDPA such as fixed spreading factor, Adaptive Modulation and Coding (AMC), fast scheduling and physical layer retransmission by applying Hybrid ARQ to provide high speed downlink packet access by means of High Speed Downlink Shared Channel (HS-DSCH). All of this required a modification to the MAC layer by splitting the functionality between RNC and NodeB. With packet scheduling embedded in a new MAC sub-layer known as the MAC-hs as shown in Figure 1.3. It can be seen that the MAC-hs is now residing in Node B and the physical shared channels are associated with this entity which will allow fast retransmission as it is closer to the UE.

Three of the fundamental properties of WCDMA have been disabled i.e. soft handover, variable Spreading Factor (SF) and fast power control. In HSDPA, only hard handover is allowed and the spreading factor is fixed at SF16 under one scrambling code [11, 19].

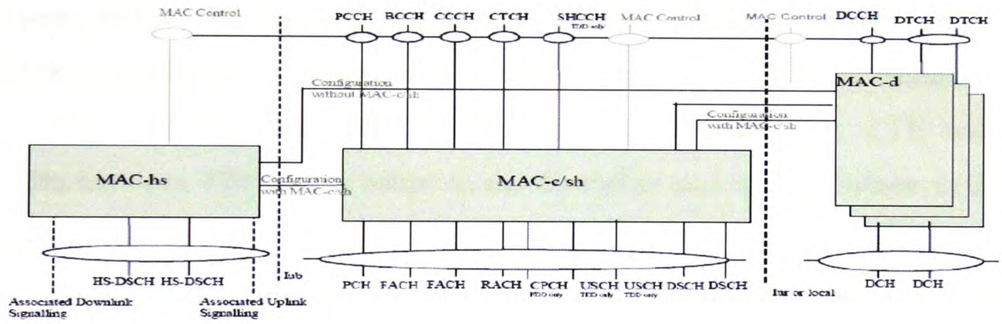

Figure 1.3 MAC-hs Architecture Node B [11]

There are 16 channelization codes available for HSDPA, of which 15 channel codes are used for data transmission and one code is used for signalling with spreading factor of 128. Adaptive Modulation and Coding (AMC) techniques have been implemented. Depending on the channel condition the scheduler can have a choice between Quadrature Phase Shift Keying (QPSK) or Quadrature Amplitude Modulations

(16QAM) modulation. In QPSK, each signalling element represents 2 bits; however in 16QAM each signalling element represents 4 bits which in theory can double the amount of data carried in a frame. Turbo coding is employed in HSDPA ranging from R=1/4 to R=3/4 depending on the channel condition. Spreading is applied to the physical channel after coding and modulation. In theory HSDPA is suppose to deliver bandwidth up to 14.4Mbps in the downlink and on the other hand, High Speed Uplink Packet Access (HSUPA) is suppose to deliver bandwidth of 1-2 Mbps in the uplink depending on the standard release.

The roll out of LTE standards to provide broadband connectivity to the mobile customers has begun. What is called 4G is at the door steps, based on OFDMA and a frequency band between 1.4 to 20 MHz, the frequency is divided into resource blocks, each block consists of 12 sub carriers of 15 KHz. The minimum bandwidth in the uplink is one resource block or 180 KHz. In the downlink the bandwidth is divided into 180 KHz resource blocks, 100 resource blocks are available if the bandwidth is 20MHz, 50 for 10MHz and so on. Three modulation techniques are envisioned 64QAM, 16QAM and QPSK. Depending on the propagation conditions each signalling element can carry 2, 4 or 6 bits. In fact this could double or triple the data rate. LTE will provide bandwidth between 170 to 325 Mbps in the downlink and up to 75Mbps in the uplink. LTE is an all-IP network that can bring between 5-10 Mbps to a subscriber mobile. This would truly enable quality video experience and multimedia mobility at a fraction of the cost to the operator and consumer.

Several studies have been conducted to insure that LTE can be deployed in the same geographical area as other mobile systems [12]. The studies show that the loss of capacity is minimal for UMTS. The deployment of LTE in the same cell site is also expected. This will further complicate the radio resource management issues, since there will be three different radio access networks to manage comprising of several technologies.

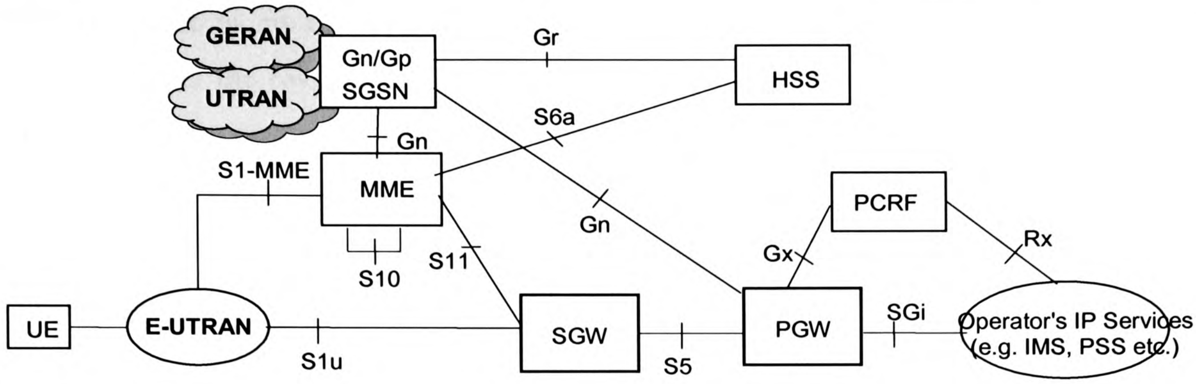

Figure 1.4 Non-roaming Architecture for interoperation with Gn/Gp SGSNs [34]

As depicted in Figure 1.4 the inter-operability between LTE and GERAN/UTRAN is enabled by the signaling interfaces as shown above, this is in accordance with Rel.7, other signaling interfaces are added to the architecture in Release 8 and 9. However, inter-operability is expected by the 3GPP standards and all these signaling interfaces are included to facilitate inter-access mobility, provide service continuity and handover between the radio systems, the actual algorithms are not standardized.

1.3 Motivation

Mobile communications is going through a rapid transformation that has not seen before. The deployment of GERAN, UTRAN and the potential deployment of LTE in the same cell site brought to the forefront several research issues. As mentioned above, these multiple radio technologies are connected via the signaling interfaces to facilitate interoperability. Multi-access network planning and radio resource management are essential aspects in the multi-radio/multi-access technology deployment, which are further complicated by the requirements to support diverse traffic classes and services standardized by 3GPP community.

There is a lack of algorithms for managing multiple radio resources in an environment where support for multiple services is expected [50-55]. Algorithms for call admission control (CAC) in multi radio environment based on traffic classes are needed as one size fits all does not work anymore. CAC algorithms have to consider several criterions before taking the decision to admit a new call.

Call retention is an essential part of any operator network algorithms. Drop calls severely affect the revenues of the operator as bad news travel fast. Drop calls have a major impact on customer perception of the service as call retention is more critical than new call admission and should have higher priority. Handover algorithms have to be sophisticated enough to coup with the diversity of RATs, the diversity of services and the knowledge of network load to be able to take the appropriate decision when handover is required. The deployment of several radio technologies complicates the handover process.

Even when the proposed algorithms are inclusive of multi-radio technologies, there is a lack of the proper simulator to investigate the proposed algorithms, which is an issue that is needed to be addressed as well. Proper simulation tool will provide a frame work for current and future mobile network deployment and optimization, this work will enable operators to save time and money when planning their networks and will have a great academic value on investigating different internetworking issues.

Hence, the work in this thesis was motivated by the need to find solutions to address the problem of interoperability among different radio access technologies in the form of developing CAC algorithms to facilitate the initial radio access technology to provide the requested service.

Another problem this thesis will address is service continuity, since most customers are mobile and changing point of attachment is expected, then the need to devise an algorithm for call retention and handover is essential. Service continuity is one of the operators' goals in providing ubiquitous service. Service continuity is also influenced by the capabilities of the user equipments devices.

Moreover, this thesis will address the lack of proper simulation tool to group all radio access technologies that have been standardized by 3GPP in a single simulator system.

1.4 Objectives

The main aim of this thesis is to propose solutions that address the problem of interoperability of multi radio technologies and design algorithms to efficiently utilize the radio resources in a multi service environment.

Thus the objectives of the thesis are as follows:

- To develop a full understanding of the interoperability issues within 3GPP radio access technologies as in GSM, GPRS, 3G, 3.5G and LTE.
- To acquire deep understanding of the problems associated with different radio access technologies in particular these related to coverage/capacity planning, optimization of multi radio networks.
- To develop a sophisticated multi-access Call Admission Control algorithm (CAC) of multi-services taking into account the availability of the radio resources, the minimum QoS requirements and the user equipment device capabilities.
- To develop a complete vertical and horizontal handover algorithm for wireless multi-access technology network taking into account the availability of the radio networks, the requested QoS, the device capabilities and the speed of the terminal.
- To design a detailed multi-radio wireless access discrete event simulation system comprising the full range of access technologies based on 3GPP standards in which the above developed multi-access CAC and handover algorithms are implemented and investigated.
- To develop an extension to the CAC algorithm to include the new radio access technology in LTE.
- To carry on detailed investigation on various scenarios to prove the efficiency of the proposed algorithms and the planning simulation tool.

1.5 Contributions

The main contributions of the thesis are as follows:

1. Development of sophisticated multi-access CAC algorithm taking into account the availability of the radio access network, the requested QoS traffic class, the load of the network and the capabilities of the user equipments devices. **(Chapter 4, Chapter 5).**
2. Develop a complete vertical and horizontal handover algorithm for wireless networks multi access technologies taking into account user mobility, traffic

class QoS, available radio access networks, the congestion in the network and the device capabilities. **(Chapter 4 and Chapter 5).**

3. Design and implement a comprehensive discrete event system simulator (DES) comprising the full range of the 3GPP radio access technologies like GSM, GPRS, 3G, HSDPA and LTE. The system simulator is designed in Visual Studio platform using C# as a programming language. The Windows platform provides a rich base class library that has been used for modeling the design objects. C# provides an object oriented modeling that was of a great value throughout this thesis. The simulator was designed with modularity in mind; the functionalities are executed when needed. The design is proven to be scalable where other technologies in the future can be added to it with minimal programming effort. **(Chapter 5).**
4. The performance of the above mentioned algorithms has been illustrated by the simulator. Several scenarios have been developed and investigated by the simulator and the algorithms. **(Chapter 6).**
5. An extension for the system simulator to include LTE radio access technology with additional traffic services as specified by the literature and the standards. **(Chapter 5).**
6. Develop an extension to the multi-access CAC algorithm in points 2 to include the new radio access technology LTE that is standardized by the 3GPP. **(Chapter 7).**
7. Investigated scenarios are presented to study the impact of the new addition of LTE radio access technology to the network and the mobile wireless network systems. **(Chapter 7).**

1.6 Thesis outline

The outline of the thesis organization is as follows:

Chapter 1: Gives a brief introduction to the mobile communications evolutions, explains the thesis motivation, describes the research objectives and lists the outline of the main contributions.

Chapter 2: The radio technologies standardized by 3GPP are discussed, the architectures for each radio access technology is presented, the interoperability of these technologies are also discussed through the signaling interfaces included in the standards and the quality of service architecture as specified by 3GPP standards is presented.

Chapter 3: Provides a literature review of the available algorithms to solve the problems presented in chapter 2.

Chapter 4: Presents a general description of the proposed algorithms designed to provide a solution for the problems presented in chapters 1, 2 and 3. A full design of the multi access call admission control in the presence of multi services and the design for handover algorithm to provide service continuity are described.

Chapter 5: The discrete system simulation tool is described and implemented along with all the entities that comprise the simulation system.

Chapter 6: Several scenarios are studied to provide insights into the simulator system and the proposed algorithms.

Chapter 7: A new module for the Long Term Evolution (LTE) radio access technology is added to the simulation system and an extension of the multi-access CAC is presented. Several scenarios are presented to study the effect of the deployment of new technology on the mobile network.

Chapter 8: Draws the main conclusion of the thesis and discusses areas for future possible investigations.

1.7 Author's publications

Book Chapter Contributions

- K. Al-Begain, S.Y. Yerima, B. Abuhaija, "***Packet Scheduling and Buffer Management,***" contributed chapter in Handbook of HSDPA/HSUPA Technology. (To be published by CRC group Taylor and Francis, June 2010).

Conference Proceedings

- B. Abuhaija, K. Al-Begain, "***Designing of Internetworking simulator to enhance GSM/WCDMA Inter networking,***" Proceedings of the 9th Annual

Postgraduate Symposium on The Convergence Telecommunications, Networking and Broadcasting(PGNet), Liverpool, U.K, June 2008 ISBN:978-1-902560-19-9.

- B. Abuhaija, K. Al-Begain, "***Enhanced Common Radio Resources Managements Algorithm in Heterogeneous Cellular Networks,***" Proceedings of the third International Conference on Next Generation Mobile Applications, Services and Technologies, Cardiff, Wales, U.K. September 2009, ISBN: 978-0-7695-3786-3.
- B. Abuhaija, K. Al-Begain, "***An Algorithm for Multi Service Continuity in Multi Radio Access Cellular Networks,***" Proceedings of the 12th International Conference on Computer Modeling and Simulation. (To be published in March 2010).
- B. Abuhaija, K. Al-Begain, "***Performance Evaluation of Service Based Call Admission Control Algorithm in Multi Radio Access Technologies,***" Proceedings of 5th Faculty of Advanced Technology Workshop, University of Glamorgan, U.K. February 2010.

Chapter 2

Technology Background

2.1 Introduction

Initially, mobile communication system was designed to deliver one service which is voice; this is no longer the case. Mobile services evolved from one traffic type to many traffic types with different set of QoS requirements. Such services include conversational, video conferencing, gaming and interactive services, all of these services use mobile radio access networks as a communications platform to deliver the required information. One of the goals of 3GPP standards has been the delivery of multimedia services while maintaining user connectivity. To deliver ubiquitous and diversify services, mobile radio networks have evolved in two directions. First, evolving from within the radio access technology; Global System for Mobile Telecommunications (GSM) circuit switched networks was designed to efficiently deliver voice services. GSM has evolved through General Packet Radio Service (GPRS) to provide packet switched services. An Enhanced Data rate for Global Evolution (EDGE) is a further evolution of GPRS. This radio access technology is referred to as GSM/EDGE Radio Access Technology (GERAN).

As the demands for higher data rate throughput required by new services and applications, Universal Mobile Telecommunications services (UMTS) was designed to deliver voice and multimedia services. This led to the development of new air interface capable of delivering such services based on Wideband Code Division Multiple Access (WCDMA), the new air interface is called UMTS Terrestrial Radio Access Networks (UTRAN) which has been standardized by 3GPP. WCDMA has been designed to support multimedia applications.

Planning and deployment of WCDMA networks is a challenging task. WCDMA is different than GSM networks because coverage and capacity needs to be done at the same time, interference plays a major role in planning WCDMA cell coverage. By increasing the cell coverage the interference increases. An evolution inside the UTRAN interface through the introduction of High Speed Downlink Packet Access (HSDPA) has increased the downlink data rate up to 14.4 Mbps in theory.

To deliver broadband services a new air interface has been standardized by 3GPP. Based on Orthogonal Frequency Division Multiple Access (OFDMA), Long Term Evolution (LTE) networks have been standardized to support data rates in excess of more than 300Mbps. This chapter will give an overview of the Radio Access Technologies (RATs), the Quality of Service (QoS) requirements and characteristics for the standardized services, deployment and internetworking of RATs.

2.2 GSM/GPRS

GSM media access control implements TDMA and (FDMA). Each frequency is 200 KHz allowing 124 frequencies in the 900 MHz band and 374 frequencies in the 1800MHz band. Each frequency is divided in the time domain using TDMA into frames; each frame has 8 time slots/channels [1]. Figure 2.1 illustrates the time and frequency structure of the 200 KHz channel.

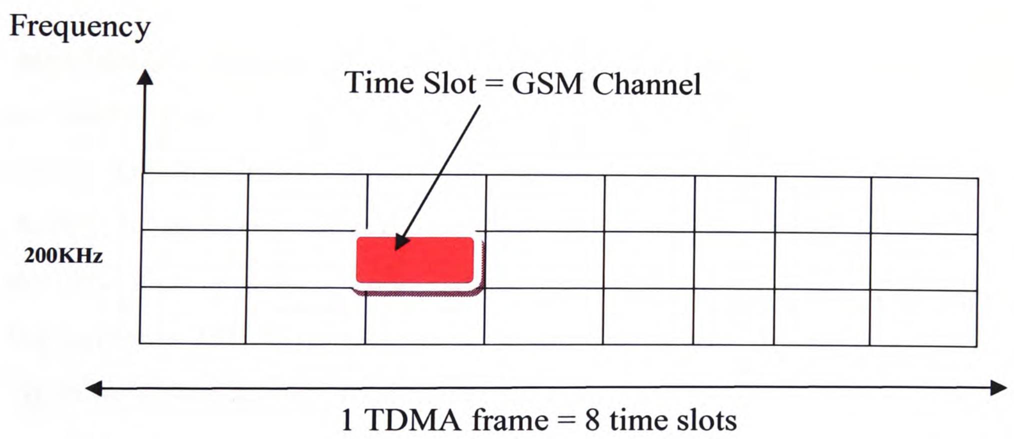

Figure 2.1 Multiple Access in GSM

GSM technology has evolved from all circuit switched network to a hybrid of circuit switched and packet switched technology network. Each time slot can be used to carry a voice call at full rate, half rate or to carry control information. The evolution of GSM circuit switched services to allow packet switched services was enabled through utilizing additional modulation and coding schemes and base station hardware enhancement. The introduction of Serving General Packet radio service (GPRS) Service Node (SGSN) and the Gateway (GPRS) Service Node (GGSN) in the core network allowed packet access to the internet. In the Base Station Controller, a Packet Control Unit (PCU) has been added to enable connection to SGSN through a new Gb interface. On the radio bearer side, time slots have been allocated dynamically and shared by packet data transmission users. The introduction of High Speed Circuit Switched Data (HSCSD) allowed the use of several time slots in parallel to achieve higher data rates. For GPRS the standards have defined four coding schemes as depicted in Table 2.1 below

Table 2.1 Coding schemes for GPRS

Scheme	Code Rate	Data Rate(Kbps) per time Slot
CS-1	½	8
CS-2	2/3	12
CS-3	¾	14.4
CS-4	1	20

This will in theory put the cell throughput at 160Kbps. GSM/GPRS has evolved through the introduction of a new modulation technique and link quality control. The octagonal Phase Shift Keying (8-PSK) modulation can carry 3bits per modulated symbol while Gaussian Minimum Shift Keying (MGSK) can carry only 1 bit per signalling symbol. An 8-PSK modulation increased the data rate of the air interface up to 60 kbps on a single time slot in theory. Link quality control includes 9 different modulation and coding schemes (MCS) as shown in Table 2.2 below, even though EGPRS utilizes GMSK scheme but this is different from the scheme used in GPRS in which the GMSK modes have incremental redundancy.

Table 2.2 EGPRS Coding and Modulation Techniques

Modulation and Coding Technique	Code Rate	Modulation	Data Rate per Time slot
MCS-9	1	8-KPSK	59.2
MCS-8	.92		54.4
MCS-7	.76		44.8
MCS-6	.49		29.6
MCS-5	.37		22.4
MCS-4	1	GMSK	17.6
MCS-3	.8		14.8
MCS-2	.66		11.2
MCS-1	.53		8.8

As depicted in **Figure 2.2**, the legacy interfaces of GSM/EDGE are still used in the standards. The A interface is Circuit Switched (CS) side of the radio access subsystem and it is connected to the MSC in the Core Network (CN) [1].

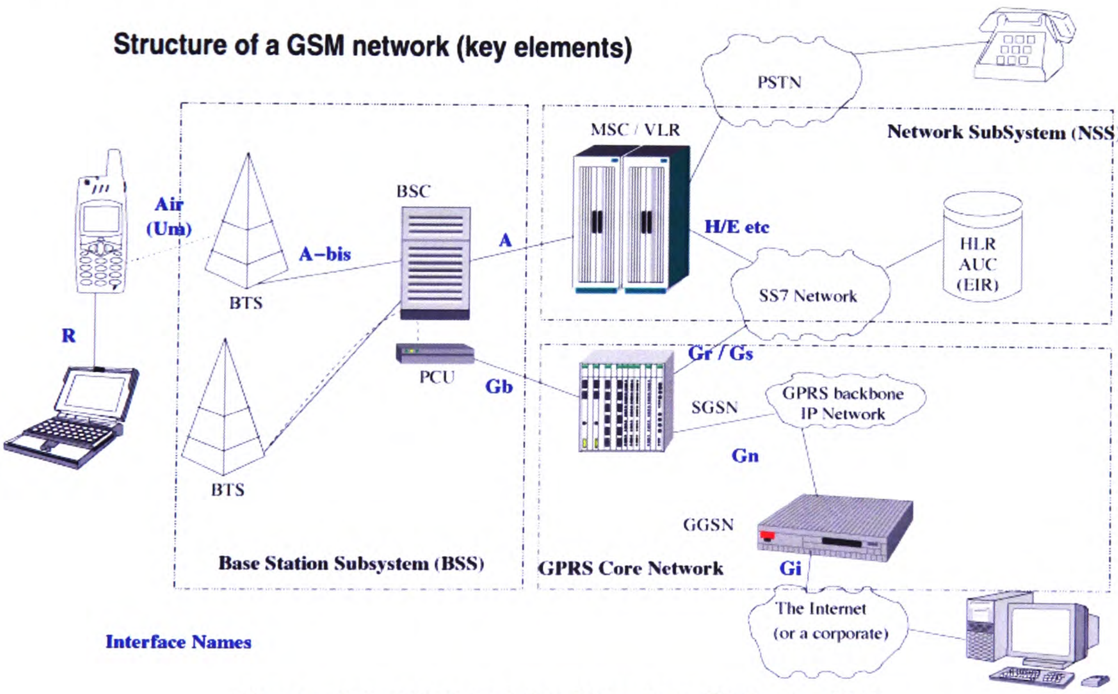

Figure 2.2 GERAN Architecture Release 4[1]

The Gb interface is the Packet Switched (PS) side of the radio access subsystem and it is connected to SGSN in the CN. This radio interface is called GERAN, **Figure 2.2** shows architecture of the radio interface [13]. The functional split between the radio network GERAN and the core network is completed by Release 5 introduction of Iu-CS and Iu-PS interfaces which enabled two outcomes, first, GERAN and UTRAN can terminate at the same core network for efficient resource optimization, second, it allowed the operators to efficiently manage the radio resources and the available spectrum while the radio technology is invisible to the end user [2].

2.3 3G/WCDMA

The data capability of the GERAN is limited. The envisioned services and applications demand higher data rates. To satisfy the need for higher bandwidth 3GPP standardized a new air interface based on WCDMA technology in Europe. UMTS Terrestrial Radio Access Technology (UTRAN) is the name of the new air interface to enable the new services offered by the operators. WCDMA uses 5MHz carrier frequency, user information is multiplied with the spreading code over the entire 5 MHz bandwidth using spreading codes. WCDMA air interface has a chip rate of 3.84 Mbps, supports variable user data rates based on the spreading factor as in (Section 2.3.2). The frame duration is kept at 10ms each with 15 slots [10, 19]. With the support of variable spreading code which translates into supporting variable data rates for users, WCDMA is well suited to support bandwidth on demand concept. The user data is kept constant for the each 10ms frame, however the data capacity can change for users on a frame by frame basis. WCDMA is capable of service multiplexing, which means multiple services with different QoS requirements can be multiplexed on one connection. From functionality point of view, the radio access network elements are grouped in UTRAN and CN elements as depicted in Figure 2.3. UTRAN elements are responsible for all radio related functionality like radio resources allocations, signaling and QoS requirements. The core elements like MSC and SGSN are responsible for switching and routing of packets and services. In the following sections a description of the UTRAN architecture and WCDMA technologies will be demonstrated.

2.3.1 UTRAN Architecture

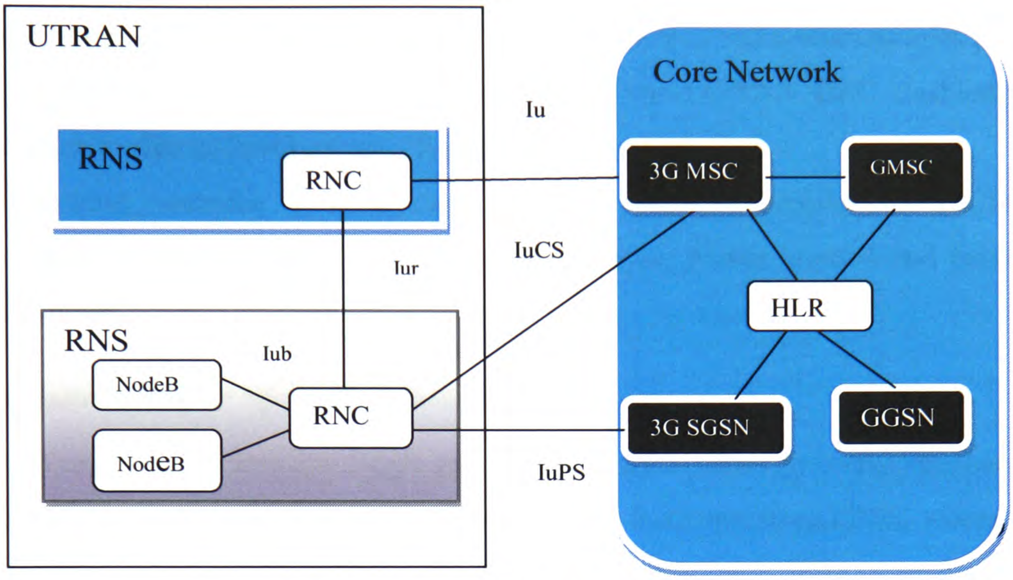

Figure 2.3 UTRAN Architecture

The components of the UTRAN and the core network entities have been discussed in Chapter 1, Figure 1.2. The Iur interface between the RNC,s is used by the UTRAN network to support mobility. RNC is responsible for the radio resources of the UTRAN and is terminating to the MSC or the SGSN as depicted in Figure 2.3. Node B can only be under the control of one RNC as depicted in Figure 2.3, however the RNC can have two roles logically, if the resources are consumed from a Node B that is connected to RNC which means that RNC is in control of the data transfer and the signaling connections to the mobile terminal, the RNC in this case is called the Serving RNC (SRNC). However, if the mobile terminal is drifting away from the SRNC and start consuming resources from a Node B under the control of a different RNC, in this case the RNC is called drift RNC (DRNC).

As explained in [14], the basic functionality of Iur interface is to support Serving Radio Network Controller (SRNC) relocation and inter RNC area update. A second functionality is to support global resource management in the form of transferring cell information and radio measurements between RNC's. The introduction of the Iu interface in the UMTS has allowed the reuse of the core network elements for UTRAN and GERAN. Iur-g interface is a control plane protocol which connects two BSSs or

BSS and RNC for exchange of signaling information. This interface is optional and from a logical point of view can be setup even in the absence of a physical interface [6]. This allowed the optimization of internetworking between the two radio technologies [6, 15]. The Iur-g interface between the UTRAN RNC and the GERAN BSC enabled inter system handover without involvement from the core network [19].

In the following sections, a brief review of the basic principles of WCDMA, channelizing codes (Spreading codes), scrambling codes, power control and handover will be discussed.

2.3.2 Channelizing or Spreading Codes (SF)

In WCDMA, users in the same cell area are distinguished by their spreading factors (SF) or channelizing codes in the downlink within one cell. Different from GSM, where each user is assigned a time slot in a frame, the spreading codes are unique to each user. At the receiver end the unique SF is de-spread to recover the user data. The spreading codes are based on Orthogonal Variable Spreading Factor (OVSF) technique [10, 18-19] as depicted in Figure 2.4. The use of OVSF permits the use of different length codes while maintaining the orthogonality between them. From Figure 2.4, it can be deduced that the more chips are used to represent the user signal the lower the user data rates for the service and the bigger spreading factor used. In the downlink the spreading factor for voice traffic is 512. Even though that this is counter intuitive but it does make sense. One important criterion of the spreading codes is that, the code root is represented by its smallest spreading factor to the root. In other words, the other spreading factors to the root and to the other end of the code on the same branch are not used. Spreading of the codes does increase the transmission bandwidth. In the uplink the spreading codes used are between 4-256 chips while in the downlink the codes from 4-512 chips are used. Each cell has its own tree and the codes are managed by the RNC.

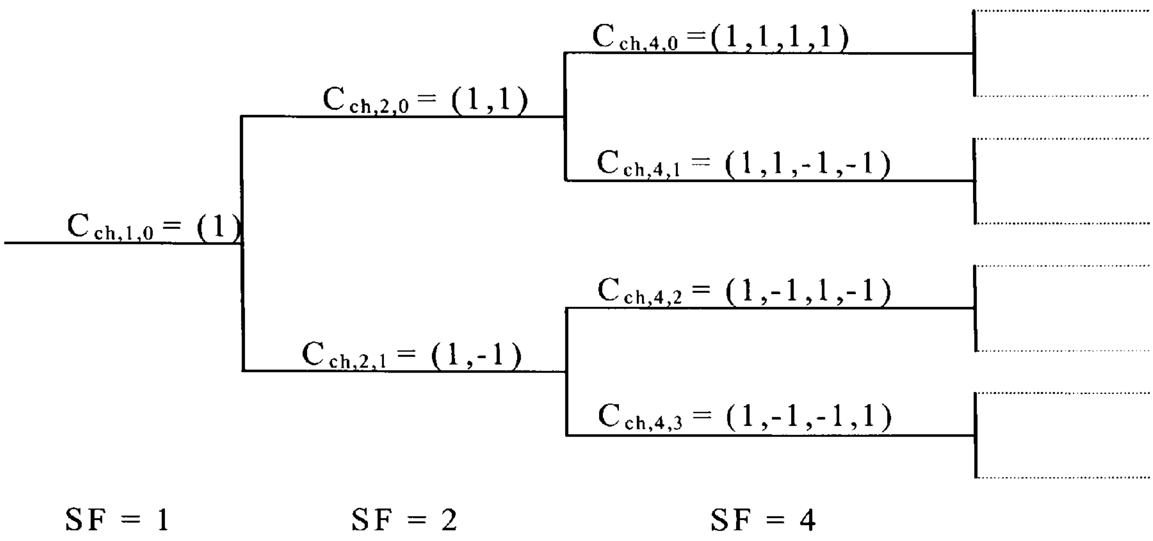

Figure 2.4 Spreading Codes

The downlink uses QPSK modulation. In QPSK modulation each symbol carries 2 bits.

2.3.3 Scrambling Codes

The scrambling codes are used to distinguish between Node Bs in the downlink and users in the uplink. In a non synchronized system like WCDMA, scrambling codes are assigned by the system for the uplink calls. In the downlink the network planner assigns a scrambling code for each cell. The transmission rate is the same as the chip rate of WCDMA and used on top of the spreading code. In other word, scrambling codes do not add any extra bandwidth to the data rate [10, 18].

2.3.4 Planning WCDMA

Planning for GSM coverage does not depend on the traffic volume or user distribution, it is centered on the number of sites required to provide radio frequency signal area coverage. Once the uplink and downlink signal strength is determined, additional capacity can be added to cover the estimated traffic. Interference is considered in GSM planning in the scope of frequency allocation to the sites. However in WCDMA, coverage depends on user distribution, service data rate requests and traffic intensity. In the subsections below factors that affect coverage and capacity of WCDMA are discussed.

2.3.4.1 Fast Power Control

WCDMA technology is power sensitive, if a terminal transmits at a full power while it is close to the base station, it can block the signal from all other terminals due to high interference contribution and give rise to the near-far problem in WCDMA [19]. Each terminal contributes to the total interference in the cell. That is why power control is critical in the uplink. There are two power control loops in the system, first, open loop power control which is used when the terminal first enters the cell coverage area. An attempt to estimate the path loss by sending signal beacon in the downlink is possible. However, open loop is not sufficient as it is not accurate enough because there is no correlation between uplink and downlink transmission. Second, the closed loop power control. There are two closed power control loops, inner loop and outer loop. The inner loop span between the UE and Node B, and it is responsible for compensating for the fading signal by providing accurate Signal to Interference Ratio (SIR) measurements. If the received signal is weak from the UE then the base station sends power up command to the UE, if the signal is strong then it will send power down command. The inner loop insures that just enough power is received for the required service. The outer loop control resides in RNC [1, 17-19]. The outer loop is responsible for ensuring the (E_b / N_0) requirement of the service. The Bit Error Rate (BER) is the number of bits that are received in error while Block Error Rate (BLER) is the number of transmission frames received in error. This in turn is defined by the target BER on the dedicated physical channel or BLER on the dedicated transport channel to ensure constant quality of service [3]. The Radio Network Controller (RNC) will command Node B to send the power commands up or down based on comparing the received quality of service with the target quality of service. In the downlink, closed power control is used to provide additional power for users at the cell edge to compensate for other cell interference. The power is shared among the different channel types whether they are common channels or dedicated channels [8, 18].

2.3.4.2 Capacity and Coverage

In general GSM coverage can be balanced on the uplink and the downlink, however, in WCDMA it is not an easy task. There are several parameters that play major role in

WCDMA. To name a few, traffic intensity, user distribution and user service requirements. In WCDMA each user admitted to the system and transmitting to node B is contributing to the total noise. As the user move further from node B, the power needed for the signal will increase. As the users leave the cell the coverage of the cell increases, as more users move in the cell the coverage decreases [8, 19]. This phenomenon is called cell breathing.

The interference at node B for a particular user is the sum of all the received power of all users multiplied by the orthogonality factor (will be discussed later). So each new admitted user to the WCDMA cell is seen as an extra noise level by other users. That is why WCDMA in the uplink is interference limited [8,17-19]. Since most of WCDMA systems have frequency reuse 1, the same frequency is used in every cell in the system. In the downlink the limitations are different. It is assumed that the power is divided among radio bearers relative to each channel capacity and data rate. In other words, the power of each channel is constant relative to other channels. Therefore, capacity is downlink limited [17-19]. To strike a balance between coverage and capacity is a delicate matter in the deployment of WCDMA. In the following two sections, the uplink and downlink loads are presented in light of the above discussion to try to estimate cell capacity.

2.3.4.3 Uplink Load factor and Downlink Load factor

In [8, 20-21] the standards and literature specified the minimum required Quality of Service (QoS) in terms of E_0/N_t for services offered as depicted in Table 2.3 below.

Table 2.3 Eb/N0 Minimum Requirements

Service	BLER %	Static	Case1*	Case2*	Case3*
12.2Kbps	1%	5.1	11.9	9.0	7.2
144Kbps	10%	.8	5.4	3.7	2.8
384Kbps	10%	.9	5.8	4.1	3.2

* Case numbers are corresponding to the multipath profiles as in [26].

In [20] 3GPP standard, the minimum Eb/N0 requirements has been specified and defined as

$$\frac{Eb}{N0} = \frac{Ec}{N0} * \frac{Lc}{Linf} \tag{1}$$

where Ec is the received total energy of the corresponding traffic channel per the noise power received by the antenna, N0 is the total noise power spectral density, Lc is the number of chips per frame and Linf is the total information bits in the frame excluding the Cyclic Redundancy Check (CRC) bits. Signal propagations depend on the geographic environment and mobility. In free space there are no obstacles in the line of sight of the signal as such the received power is proportional to the inverse of the square of the distance between the antenna and the user equipment. However, in the presence of buildings, trees and moving objects like cars, most probably that the line of sight is not there anymore and this raise the concept of multipath. The signal strength fluctuation is known as fading, if the fading is frequent it is called fast fading and if it happens over longer time period it is called slow fading. Received signal power strength is influenced by slow fading which is largely influenced by the distance between the sender and receiver. If the receiver be in a shadow of a hill or building, it is called shadowing and modeled by the lognormal distribution with a standard deviation depending on the operating frequency of the sender. Fading can also be short term and caused by reflection of a signal from large obstacles such as buildings, or refraction which is when the signal travel through two mediums of different densities, or by scattering from small objects and it can be because of a sharp edges of an object like hills or mountains which is called diffractions. Four cases of multipath fading are listed in Table 2.3, depending on the mobility or speed of the UE and the operating frequency band. The static case presented in Table 2.3 is for a propagation conditions in the presence of Additive White Gaussian Noise (AWGN) environment where no fading or multi-paths exists for this propagation model. Cases 1 to 3 are for different mobility average speed between the sender and receiver.

As mentioned in Section 2.3.2 above, the signal is spread over the 3.84Mcps by the spreading code. This produces a processing gain after the de-spreading of the signal and calculated using Equation (2).

The processing gain as defined in [19] is given as follows:

$$Processing\ Gain = 10 * Log10((3.84 * 10^6)/(\upsilon * \mathrm{R})) \tag{2}$$

where R is the service bit rate; υ is the activity factor of the service. Usually the activity factor is 0.5 for voice and 1 for data service. For the minimum required E0/N0 the user power signal has to be as[19].

$$\frac{E_b}{N0} = \text{ProcessinGain} * \frac{\text{power of user signal}}{(\text{Ioc+Ihc})-\text{power of user signal}} \tag{3}$$

where I_{oc} is other cells users interference power, I_{hc} is home cell users interference power. As discussed in the coverage and capacity Section 2.3.4.2, each user admitted to the cell contributes a certain amount of interference (noise) to other users in the cell and in the neighboring cells. This gives rise to the term noise rise and it is defined as the total noise plus the thermal noise in the cell compare to the total thermal noise as

$$\text{Noise rise} = \frac{\text{Ioc+Ihc+Pn}}{\text{Pn}} \tag{4}$$

Where, P_n is the thermal noise power of the base station (Node B). Let us define I_{total} as the numerator in Equation (4). The load factor can also be defined as in [17] as the following:

$$Load\ Factor = \frac{Ioc+Ihc}{I\ total} \tag{5}$$

where the Load factor can be written as ηul and represents the load in the uplink while in the downlink it can be written as η_{DL}.

From Equation (4) and (5) the noise rise can be deduced as:

$$\text{Noise rise} = \frac{1}{1-\eta\text{ul}} \tag{6}$$

it is clear that if the uplink load reaches close to 1 then the noise rise can reach infinity and in this case the coverage of the cell is close to zero. In turn this requires an infinite power supply to maintain the service. Usually the cell is constraint to 50-60% uplink load if the constraint is coverage. However, if the deployment is capacity constraint then the uplink load is configured at 75-80% [18-19].

In [19] the uplink load has been estimated as the following:

$$\eta ul = \frac{Eb/N0}{W/R} * N * \upsilon * (1 + i) \tag{7}$$

where η_{UL} is the load factor in the uplink, this value is less than 1, and depends on the noise rise as per Equation (6). The noise rise in dB is $-10\log_{10}(1-\eta ul)$. W is the chip rate (3.84Mcps), R is the bit rate for the user for that specific service. N is the number of users in the cell, υ is the activity factor and i is the other cell to own cell interference factor which has a typical value for Omni directional antenna of 55% [19]. From Equation (6), the number of users based on the service profile is calculated [18, 19].
In the downlink [8, 19, 20] the following Equation is used to estimate the downlink load

$$\eta\, DL = \sum_{I=1}^{N} \upsilon * \frac{\left(\frac{Eb}{No}\right)}{\frac{W}{R}} * [(1 - \acute{\alpha}) + i)] \tag{8}$$

Where, the only parameter that is different from the uplink Equation is $\acute{\alpha}$ which represent the orthogonality factor and considered 50% for vehicle channel and 90% for pedestrian channels. Different amount of power is used for each user based on location and transmission conditions. The sum of all individual powers plus the power used for common channels (typically 3.6Watts) is a very effective way to estimate base station dedicated channel power transmission [8, 18, 19].

2.3.4.4 Radio Resource Management (RRM) Modeling in UMTS

From the discussion in the previous section, a balance based on QoS requirements (Eb/N0) as specified in the standards [20] (shown in Table 2.3) and the number of users serviced by the cell can be calculated. The uplink bound of the base station load in the uplink is the interference contribution. This is illustrated in Equations (5) and (6). The number of users that a WCDMA cell can admit and serve depends on the QoS requested, the required data rate and the interference generated by the user service if admitted and the intensity of traffic. There are different traffic classes as specified by the standards, real time and non real time traffic.

System level modeling of UMTS can be thought of two types, static system level or dynamic system level. In static system the dynamic aspect is neglected as in mobility model and average speed. Static system level modeling uses snapshots analysis to carry out system studies. The snapshots consist of how many users are serviced and the individual user location, service requests and user equipment capabilities. Several link level measurements, one of these measurements is Carrier to Interference Ratio (CIR), and the required CIR determines the relationship between the BLER and the Eb/N0 of a specific radio bearer or user service request. The other measurements are the orthogonality factor in the downlink, RNC chooses the orthogonal codes for each service, these codes as explained earlier are mutually orthogonal and are not suppose to interfere with each other but because of multipath propagation as explained in Section 2.3.4.3, the signal loses some of this property and interference from the same cell is modeled by the loss of orthogonality which is a value between 0 and 1. The propagation data is also modeled in static system level simulators, the channel variation depends on the fading in the form of shadowing and modeled after the lognormal distribution and typically is given 8dB for outdoor system. The mobile terminal is classified based on the power used by the terminal as the maximum power is the limiting factor in the uplink coverage. Static system level simulators lose the small scale dynamic effects as in fading and power control, and the large scale dynamic effects like handover and load control. On the other hand, dynamic system level simulators can incorporate system mechanisms in greater details such as, the call arrival and departure, mobility of the users is also modeled to facilitate the handover algorithms studies. Radio resources management mechanism can be modeled in addition to system capacity. When modeling different traffic classes with distinct QoS requirements dynamic simulators are better suited than static simulators, but this comes at the expense of complicated analysis and huge storage requirements.

2.4 HSDPA

High Speed Downlink Packet Access (HSDPA) evolved from WCDMA through utilizing a number of existing technologies. Several techniques have been employed to compensate for the change of link conditions. The main theme is based link adaptation by modifying the transmission parameters of the system to adapt to the instantaneous transmission conditions. Among many, HSDPA employs fixed spreading factor, Adaptive Modulation and Coding (AMC), fast scheduling and physical layer retransmission by applying Hybrid ARQ to provide high speed downlink packet access by means of High Speed Downlink Shared Channel (HS-DSCH) [22]. All this implies that substantial changes have been made to Node B to enhance Release 99 WCDMA with packet scheduling embedded in a new MAC sub-layer known as the MAC-hs [11]. It is important to note that HSDPA is not a different air interface than WCDMA, but physical layer and Data link layer enhancements have been adopted to allow for higher data rates up to 14.4Mbps and more [25].

In the following, some of the technology changes that enabled HSDPA will be discussed.

2.4.1 Physical Layer

On the radio system level several WCDMA functionalities are adapted to enable HSDPA. Three of the fundamental properties of WCDMA have been disabled i.e. soft handover, variable spreading factor (SF) and fast power control. In HSDPA only hard handover is allowed. The spreading factor is fixed at SF16 under one scrambling code [10]. Figure 2.5 depicts the spreading factor and the number of channelization codes available for HSDPA, when the total cell power is used for HSDPA, all 15 channel codes are used for data transmission while one code is used for signaling with spreading factor of 128 [22].

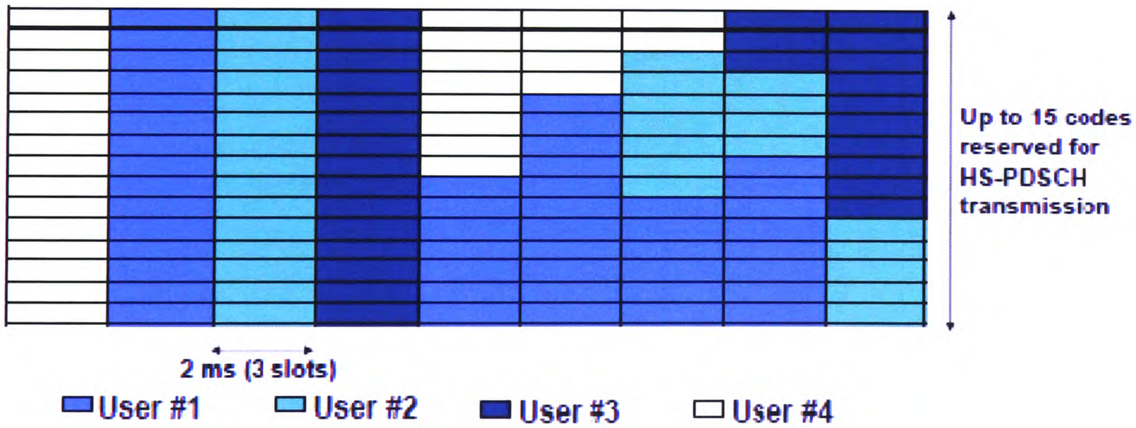

Figure 2.5 HSDPA Spreading Codes

However, the example in Figure 2.6 illustrates that the cell is shared between WCDMA services and HSDPA services, where 8 codes are used for HSDPA while the rest are used for WCDMA.

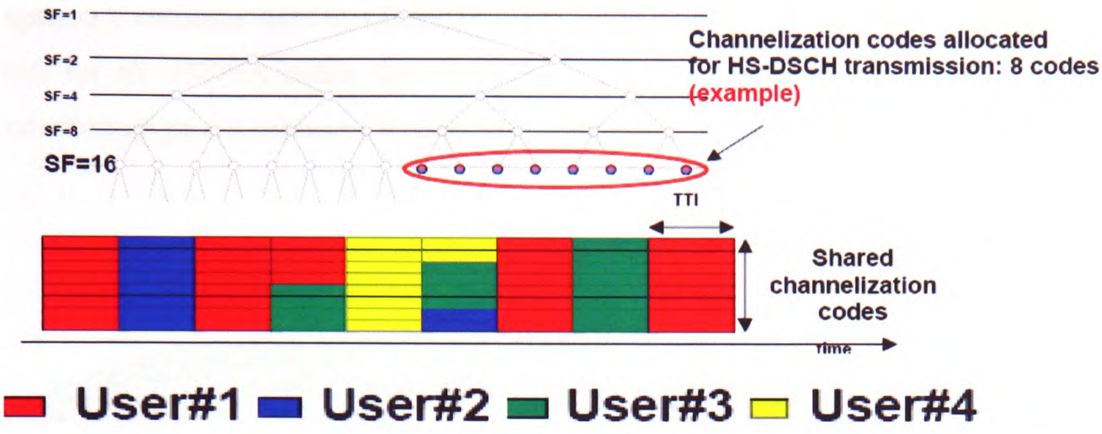

Figure 2.6 Spreading Factor

The concept of cell code sharing among Release 99 WCDMA cells and HSDPA highlights the power management issue in the cell among the two technologies [25, 106]. Release 99 WCDMA power control is based on, if the user in favorable propagation conditions then less power is transmitted to the user, however if the user is at the edge of the node or in less favorable propagation conditions then more power is transmitted to the user as depicted in Figure 2.7.

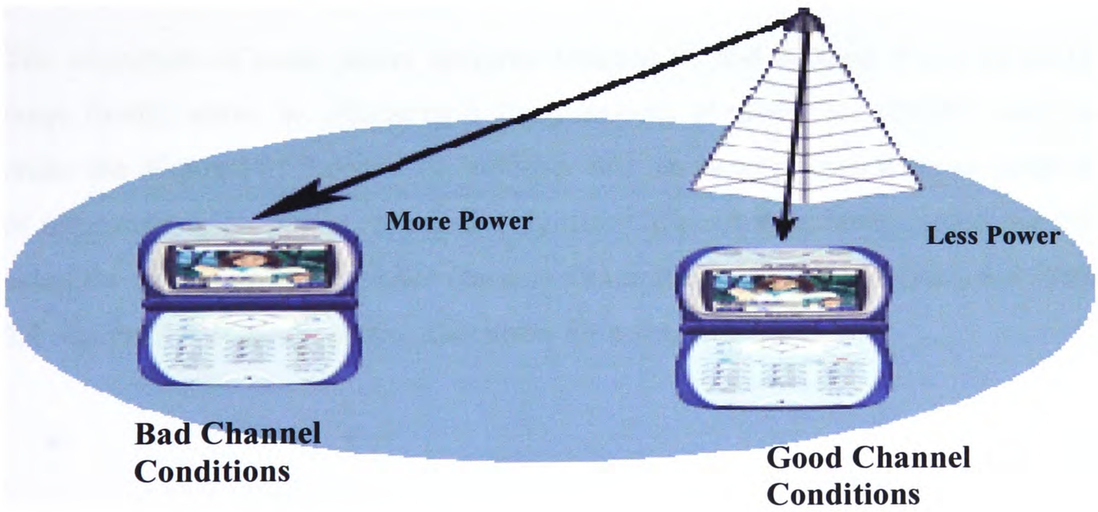

Figure 2.7 Release 99 Power Control

However, in Release 5 with the introduction of HSDPA, if the customer is in good propagation conditions then this translates into more data rate throughput as the power is constant for all HSDPA codes, and if the user is in less favorable conditions then less data rate throughput is achieved in this case as depicted in Figure 2.8.

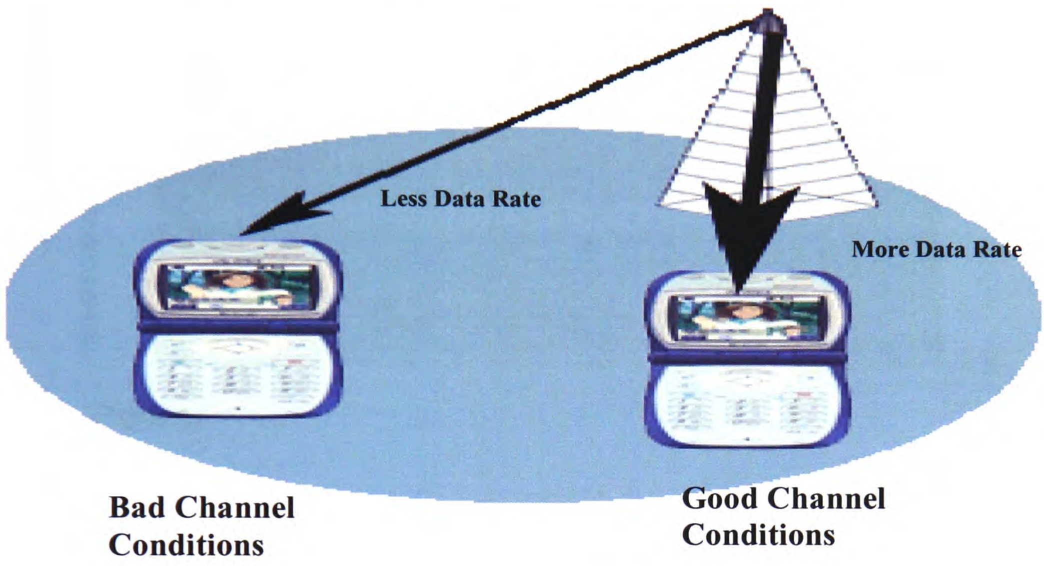

Figure 2.8 Rate Adaptations with HSDPA Nodes

The allocation of node power between Release 99 and Release 5 can be done in two ways firstly, static, by allocating a fixed amount of power for HSDPA and the rest is under the disposal of Release 99; however this has been proven to be an inefficient way of allocating node power. Secondly, dynamic allocation is better suited for efficiently using the excess power left after Release 99 requirements has been satisfied [25]. Figure 2.9 depicts the dynamic power allocation for a shared node B.

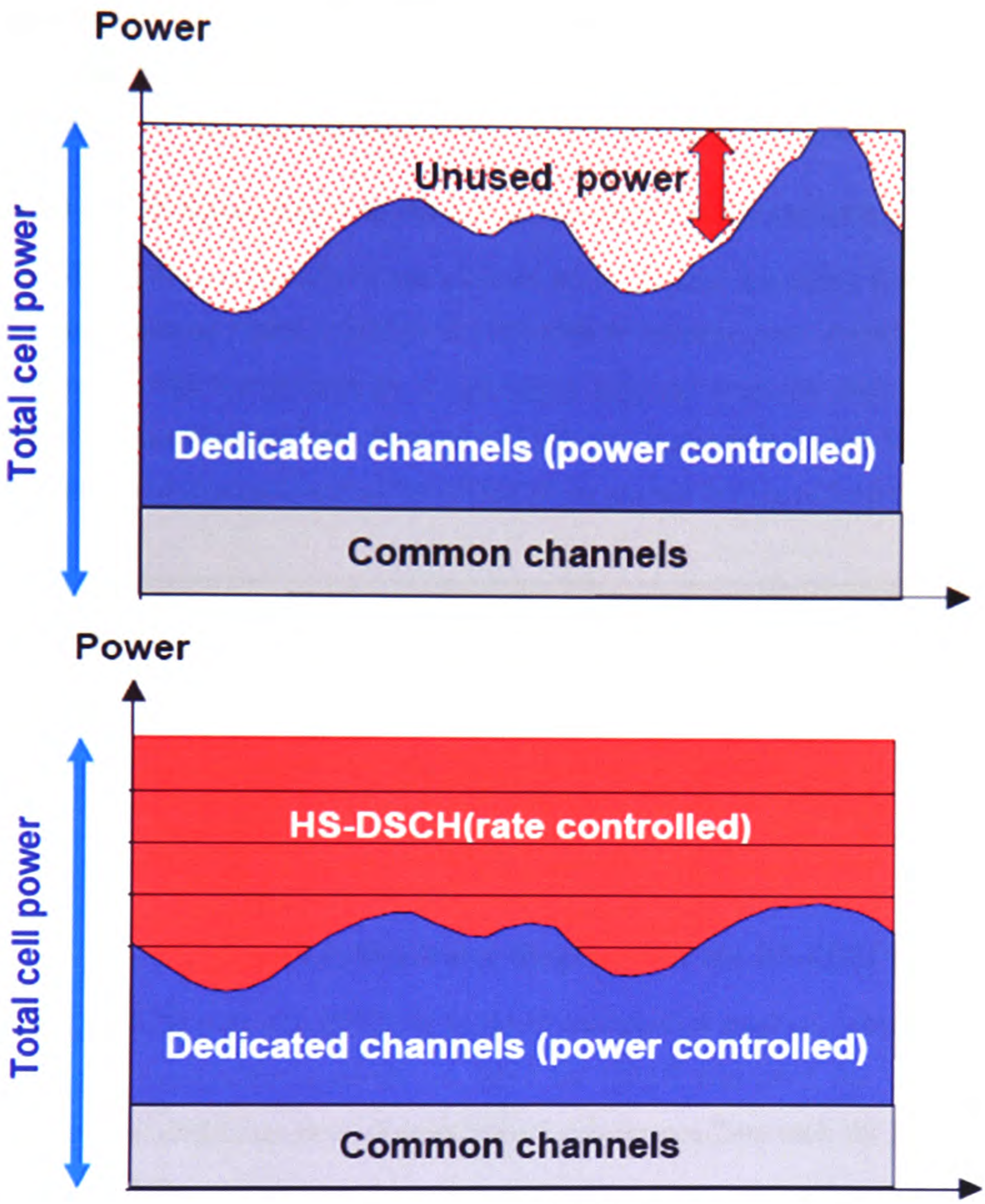

Figure 2.9 Node B Transmit Power Allocation [25]

Adaptive Modulation and Coding (AMC) techniques have been implemented, depending on the channel condition, the scheduler can have a choice between QPSK or 16QAM modulation [10, 23, 25]. In QPSK each signaling element represents 2 bits. However, in 16QAM each signaling element represents 4 bits which in theory can double the amount of data carried in a frame. Turbo coding is employed in HSDPA ranging from R=1/4 to R=3/4 depending on the channel condition as depicted in Table 2.3. Spreading is applied to the physical channel after coding and modulation which in effect increases the bandwidth of the signal. Scrambling operation is applied to the signal and this operation is a mean to separate base stations as mentioned in Section 2.3.3. Three new physical channels have been defined for HSDPA. Below is a brief description of them.

2.4.1.1 High Speed Physical Downlink Shared Channel (HS-PDSCH)

A HS-PDSCH corresponds to one channel code of fixed spreading factor from the set of 15 channelization codes available for HS-DSCH transmission. Multi-code transmission is allowed, which translates to a UE being assigned multiple codes in the same HS-PDSCH Transmission Time Interval (TTI=2ms), depending on the UE capability. The sub-frame and slot structure of HS-PDSCH are shown in Figure 2.10 below.

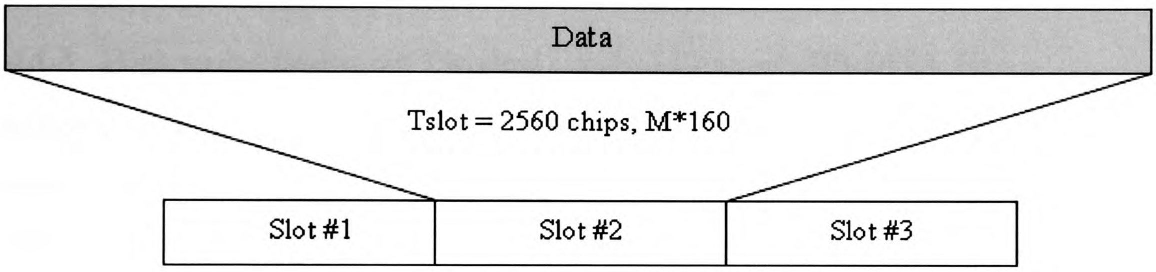

Figure 2.10 **Sub-frame Structure for the HS-PDSCH**

An HS-PDSCH may use QPSK or 16QAM modulation symbols, where M is the number of bits per modulation symbols i.e. M=2 for QPSK and M=4 for 16QAM. The 10ms frame of WCDMA has been divided into 5 sub-frames 2ms each for fast scheduling and retransmission.

2.4.1.2 High Speed Shared Control Channel (HS-SCCH)

The HS-SCCH is a fixed rate (60 kbps, SF=128) downlink physical channel used to carry downlink signaling related to HS-DSCH transmission. Figure 2.11 shows the sub-frame structure of the HS-SCCH.

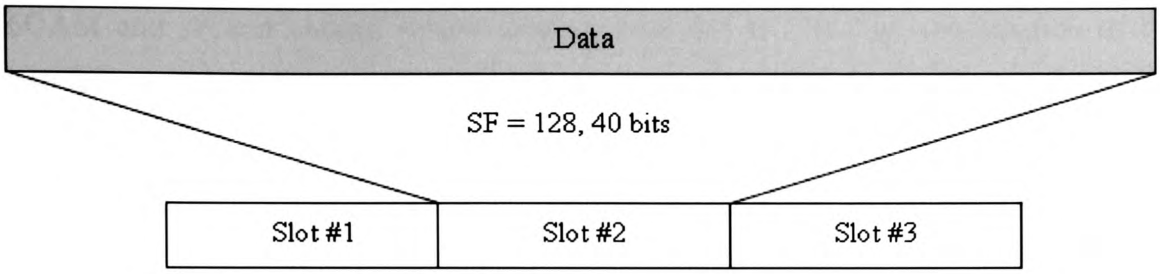

Figure 2.11 Sub-frame structure for the HS-SCCH

HS-SCCH is transmitted two time slots ahead of HS-PDSCH. It carries critical signaling information to the UE such as the channelization code set; modulation scheme and transport block size. HS-SCCH also carries the HARQ process number, redundancy version and UE identity among other information [22-24].

2.4.1.3 High speed Dedicated Physical Control Channel (HS-DPCCH)

HS-DPCCH is used to carry the signaling information in the uplink direction. The channel has SF= 256, 10 bits per slot and 30 bits for the sub-frame (15Kbps). There is at most one HS-DPCCH per UE. HS-DPCCH carries the Hybrid ARQ ACK, in one slot and Channel Quality Indicator (CQI) in the other two slots. Figure 2.12 shows the sub-frame structure for the HS-DPCCH.

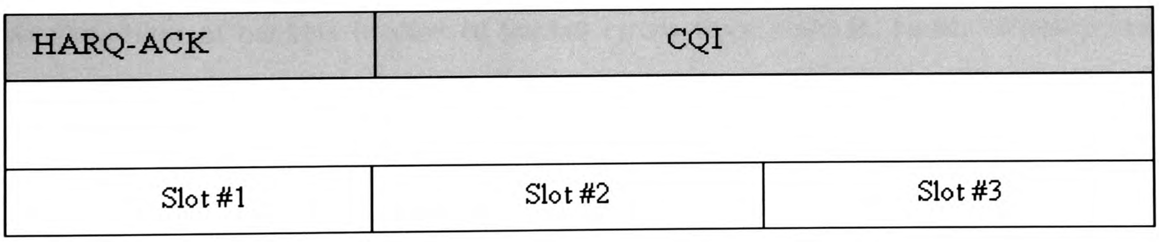

Figure 2.12 Sub-frame Structure for the HS-DPCCH

The CQI represented by 5 bits for a value is between 1..30; based on this feedback from UE, the scheduler in Node B decides on the modulation and transport block size to send in the next TTI transmission if there is available data for the UE in the buffer [11]. HS-DPCCH is transmitted 7.5 time slots (5ms) after the reception of the HS-PDSCH.

Depending on the reported channel quality condition Transport Format and Resource Combinations (TFRC) is determined. There are two values for modulation, QPSK and

16QAM and several coding values ranging from 1/4 to 3/4. The combination of the modulation and coding are adjusted based on the channel quality conditions as mentioned earlier. TFRC largely depends on the UE capabilities, modulation and coding rate [10, 22].

Table 2.4 Theoretical bit rates with 15 Multi-codes

TFRC	Modulation	Effective Code Rate	Maximum throughput (Mbps)
1	QPSK	¼	1.8
2	QPSK	2/4	3.6
3	QPSK	¾	5.3
4	16QAM	2/4	7.2
5	16QAM	¾	10.7

The maximum throughput in Table 2.4 was used as a guide when modeling HSDPA.

2.4.2 HSDPA MAC-hs Architecture

In Figure 2.13, the functional entities of the MAC-hs are depicted. The MAC-hs entity has been moved to Node B. This functional split in the MAC layer compared with Release 99 has made several enhancements possible. First, this allows fast retransmission of packets in case of packet errors from node B. Faster retransmission means shorter delays for packet data operations. Second, Adaptive Modulation and Coding (AMC) technique is employed since the scheduling decision has been moved to Node B. Table 2.4 shows modulation and coding rates used with HSDPA. Third, fast HARQ mechanism at layer 1, allows for chase combining or incremental redundancy retransmissions in case of errors. When re-transmitting using chase combining each frame is self decoded as it is an exact replica of the first transmission, however when using incremental redundancy the retransmission is for an additional redundant parity bits.

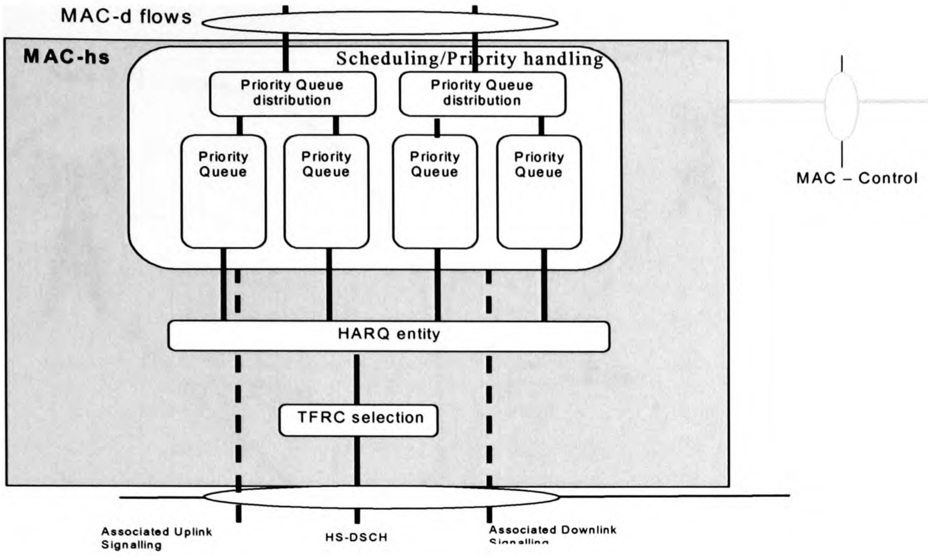

Figure 2.13 MAC-hs UTRAN [11]

Below is a brief description of the functional entities inside MAC-hs [11]:

- **Transport Format and resource combination (TFRC)**: Selects the appropriate bit rate based on information received from the HS-DPCCH in the uplink.
- **HARQ entity**: There is one HARQ entity per user in Node B. There shall be one HARQ process per HS-DSCH per TTI, 2ms in HSDPA. HARQ entity can support multiple processes (up to 8 for each user).
- **Priority queues**: Are used for differentiating services. If flow has a high priority (i.e.VoIP), then the flow should be forwarded to a high priority queue [8, 11, and 19].

2.4.3 Radio Resource Management Modeling of HSDPA

In WCDMA Release 99, the received energy per user bit to noise ratio (E_b/N_0) was used as a guide for the quality of service required on the link as mentioned above in Section 2.3.4.3. The (E_b/N_0) corresponds to a certain BLER for a given service data rate where spreading gain is the only adaptation parameter as in Equation (7) and (8) in Section 2.3.4.3. HSDPA employs adaptive modulation and coding which is changing with each transmission of 2ms sub-frame along with the number of HS-PDSCH codes used in the sub-frame. This is why (E_b/N_0) cannot be used as a defined metric for HSDPA [25].

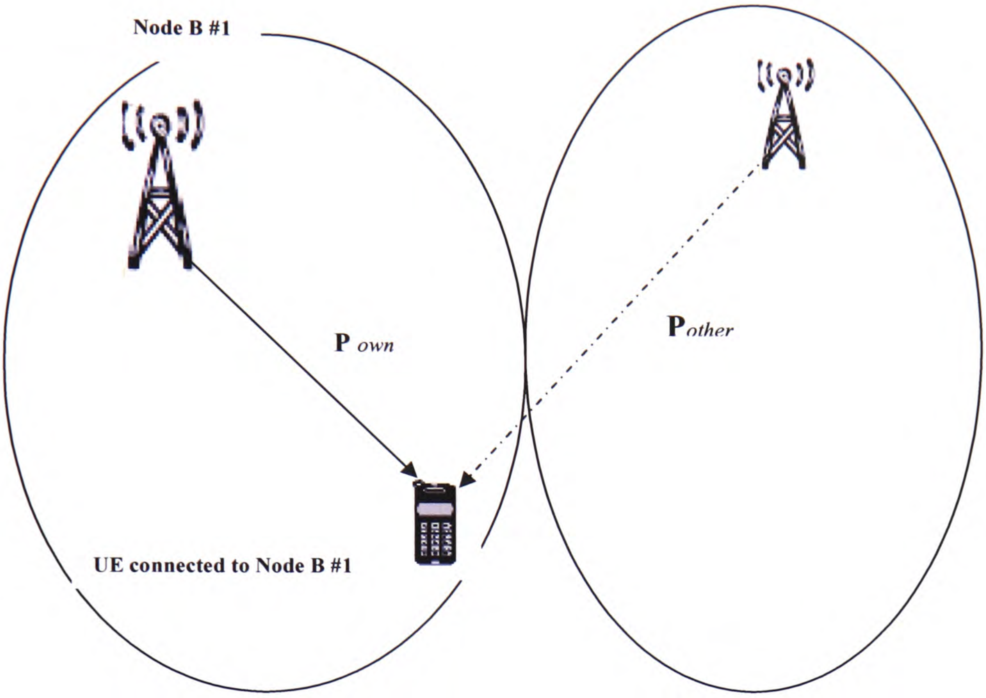

Figure 2.14 HSDPA performance factors

HSDPA link adaptation functionality AMC, multi code transmission and fast physical layer transmission and re transmission every TTI of 2ms renders (Eb/N0) useless for HSDPA. Therefore an average Signal to Interference Plus Noise Ratio (SINR) is used as in the following Equation (9) and as depicted in Figure 2.14:

$$SINR = SF16 * \frac{P\ HS\text{-}DSCH}{(1-\acute{\alpha})*Pown+Pother+Pnoise} \tag{9}$$

where SINR is the signal to interference plus noise ratio after de-spreading, SF16 is the spreading factor for HSDPA codes which is fixed at 16. $P_{HS\text{-}DSCH}$ is the received power for HS-DSCH codes. P_{own} is the base station power. P_{other} is the received other cell interference and P_{noise} is the receiver thermal noise. Table 2.4 above provides a guide into HSDPA cell bit rate budget, the transmission throughput depends on the instantaneous radio link condition (every 2ms).

2.5 Long Term Evolution (LTE)

Data customers have been boosted by the introduction of HSDPA. The end user exponential demand for data services shows the value end user finds in wireless data networks. The demand for higher data rates regardless of the end user mobility and the exponential consumption of data rates as experienced by HSDPA led the standardization community to deploy broadband technology under 3GPP standards. The 3GPP community has roled out a new air interface standards called Long Term Evolution (LTE) to provide broadband services [29]. LTE is a broadband technology, based on Orthogonal Frequency Division Multiplexing Access (OFDMA). Wireless broadband solution can provide low cost broadband coverage in the absence of wire line infrastructure. Wireless broadband is a common sense alternative in rural areas, urban areas and in growing markets. It has been reported in Fierce Broadband wireless online magazine in Dec. 14, 2009 "Over in Oslo, consumers can get TeliaSonera's LTE service, including the modem, for just 1 Norwegian Kronor ($0.17) per month until April 1, 2010, after which the monthly tariff goes up to 699 Norwegian Kronor ($120) and there will be a fee for the modem". The same article has reported "The first customers will help to develop this pioneering service and will have very attractive terms and conditions to do that, We'll have an open dialogue with early customers about how they experience our products and services."

Below is an overview of LTE Architecture and Radio resources.

2.5.1 Overall Architecture

LTE designed as an all IP network. In Figure 2.15 below, the functional entities of the architecture are depicted along with some data and signalling interfaces.

The Evolved Packet System (EPS) elements are shown in Figure 2.15. In the Evolved Packet Core (EPC) network side resides the following elements [27, 29, 34]:

- **Serving Gateway** (S-GW): All user IP packets are transferred through the S-GW. S-GW can be in control of one or more e-Node B which enables the serving gateway to become the local mobility anchor for the data bearers when the UE moves between e-Node Bs.

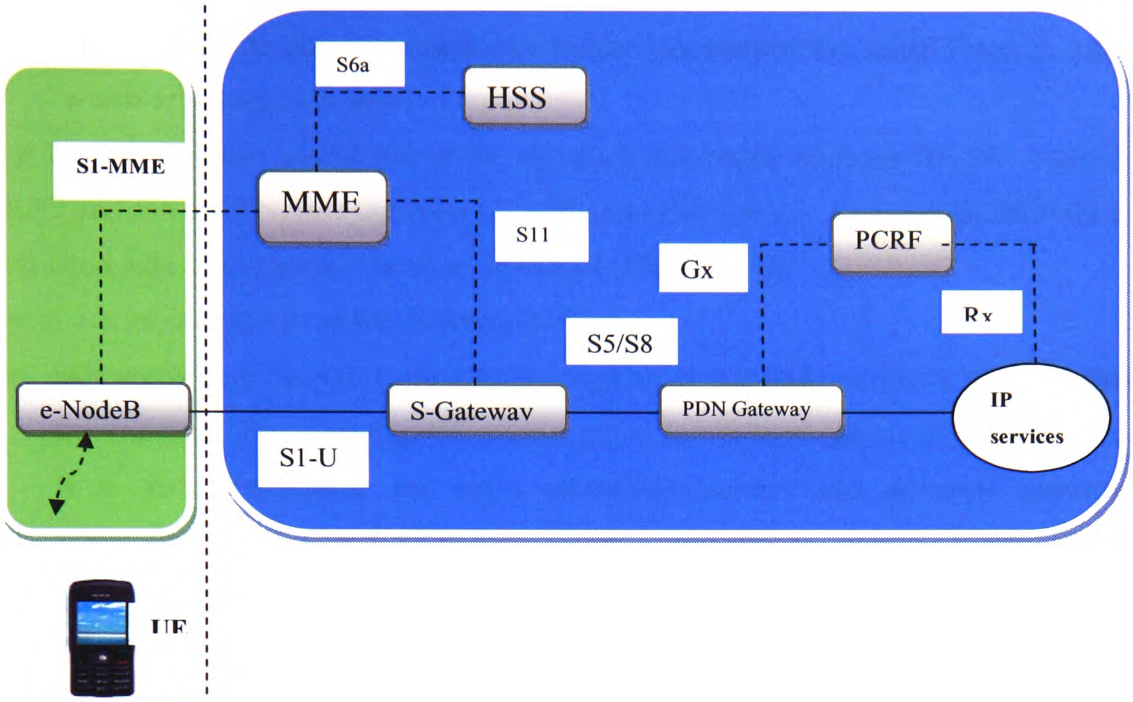

Figure 2.15 EPS Network Elements [30]

- **Mobility Management Entity** (MME): Is in control of the Non-Access Stratum (NAS) protocol that runs between the UE and the EPC. The main function of the MME includes the establishment, maintenance and release of the bearers which is handled by the session management layer in NAS [34]. Another function that is handled by MME is the establishment of the connection and security between UE and NAS which is handled by the mobility management layer in NAS [29].
- **Packet Data Network Gateway** (PDN-GW): It is the edge router between the EPS and the external data network. It allocates IP addresses and UE uses these addresses to communicate with other IP hosts in different networks. It serves as the last mobility anchor in the operator's network.
- **Policy and Charging Resource Function** (PCRF): Is responsible for policy and charging control. It makes the decision on how to handle the connection QoS. It communicates with Policy Control Enforcement Function (PCEF) in the PDN-GW to enforce to communicate these decisions.

- **Home Subscription Server** (HSS): All permanent users' profiles are kept in the database. It also keep track of the user locations at the MME level, in other words at a track area level.

Figure 2.15 also shows partial list of the signaling interfaces between the EPC entities and EPC and E-UTRAN, and between UTRAN and E-UTRAN. As far as E-UTRAN in the left hand side of Figure 2.15 and as depicted in Figure 2.16.

Below is a brief discussion of the different entities:

- **e-Node B**: It is connected to EPC through S1-U interface to S-GW for user data transmission. For signaling it is connected to MME thorough S1-MME interface. It is also responsible for radio admission control and dynamic resource scheduling. IP Header compression which is essential for real time services and data ciphering and integrity functionalities are also residing in e-Node B. The functional split between the core network and e-Node B is completed by transferring all the radio functionality to e-Node B [27-28, 30].
- X2 Interface: EUTRAN can have one or more e-Node B, e-Node Bs can be connected through X2 interface. This signaling interface can is used for load balancing, interference information exchange and handover preparation and execution stages [11, 27].

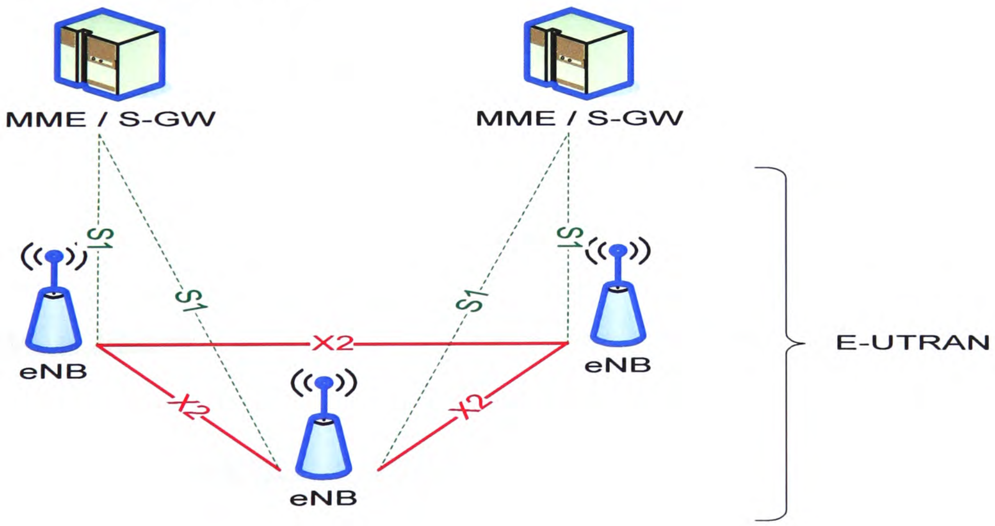

Figure 2.16 **E-UTRAN Architecture [30]**

2.5.2 OFDMA Principles

Orthogonal Frequency Division Multiple Access (OFDMA) has the following properties [27]:

- Good performance in frequency selective fading.
- Low complexity of base band receiver.
- Good spectral properties.
- Link adaptation and frequency domain scheduling.
- Compatibility with advanced receiver and antenna technologies.

OFDMA has already been adopted in Wireless Local Area Networks (WLAN) and digital TV. The basic deployment can be on 1.25, 5, 10, 15 and 20 MHz. The frequency is divided into resource blocks of 180 KHz each, which is the minimum bandwidth allocation. Each 180 KHz is divided into sub carriers of 15 KHz. In the time domain, the frame is 10ms and further subdivided to sub frames of 1ms, each is divided into two time slots of 0.5ms. There are 6, 25, 50, 75, and 100 resource blocks respectively for each frequency bandwidth. The minimum frequency bandwidth has 6 resource blocks. There are 72 (1.25MHz / 180KHz) subcarriers when using extended cyclic prefix, each resource block consists of 12 subcarriers, each of the 12 subcarriers (15 KHz) is a Resource Element (RE) which is the smallest element in 0.5ms slot. In the downlink the resource blocks allocation and Transport Block Sizes (TBS) are signaled to UE through the Downlink Channel Indicator (DCI) which is mapped in the downlink, on the Physical Downlink Control Channel (PDCCH) [31-33]. The downlink peak data rates are given in Table 2.5 for different Modulation and coding techniques.

Table 2.5 **Downlink peak bit rate with TBS considered (Mbps) [27]**

Modulation and coding	MIMO usage	5MHz/ 25 RB	10MHz/ 50 RB	15MHZ/ 75 RB	20MHZ/ 100 RB
QPSK	Single stream	4	8.0	11.8	15.8
16QAM	Single stream	7.7	15.3	22.9	30.6
64QAM	Single stream	18.3	36.7	55.1	75.4
64QAM	2x2 MIMO	36.7	73.7	110.1	149.8

In Table 2.5, values for 1.25MHz and 3MHz are not included. When modeling LTE several overheads are necessary to include, naming a few of the overheads, such as guard band, broadcast channel and synchronization channel. The effective downlink peak data rate is one half of the values in Table 2.5 and will serve as a guide for our simulation tool peak bit rate budget in the downlink.

2.6 Inter- RAT Inter-Networking

In Table 2.6 below, an overview of the evolution of 3GPP Radio Access Technologies (RATs) is presented.

Table 2.6 Evolution of 3GPP Radio Access Technologies

Year	Technology	Standards Body	Capacity	Services	Domain
1992	2G, GSM based on TDMA/FDMA	3GPP	One frequency /200KHz /8 Time Slots	Voice Mainly and SMS services	Circuit Switched (CS)
1995 - 1997	2.5G, GPRS based on TDMA/FDMA	3GPP	Time Slots, shared, 170 Kbps	Voice, limited internet services and SMS	Circuit Switched (CS) and Packet Switched (PS)
1999	3G, UMTS based on WCDMA	3GPP	Spreading Codes, 384Kbps	• Conversational • Streaming • Interactive • Background	CS and PS
2002 - 2005	HSDPA/HSUPA based on WCDMA	3GPP	Spreading codes, 14.4Mbps downlink, 384Kbps uplink	• Conversational • Streaming • Interactive • Background	CS,PS and The introduct ion of IMS.
2010	LTE based on OFDMA	3GPP	Resource Blocks (180 Kbps), 320Mbps.	• Streaming • Interactive • Background	Packet Switched (PS) only.

Each RAT has its own radio resources. The collection of these radio resources are a radio resource pool. To utilize the radio resource pool efficiently, the operators need to devise advanced Call Admission Control (CAC) algorithms. This has proven to be a daunting task. This problem has been further complicated by two other issues; first, the diversity of the services and the availability at the moment of three radio access networks to choose from to deliver these services (services will be discussed later in this Chapter). Second, mobility of users while maintaining service connectivity for voice and real time services at all times. For data transmission a reliable connection should be maintained when the user is moving from one coverage area to another. In Figure 2.17 the internetworking architecture of GERAN, UTRAN and LTE is presented.

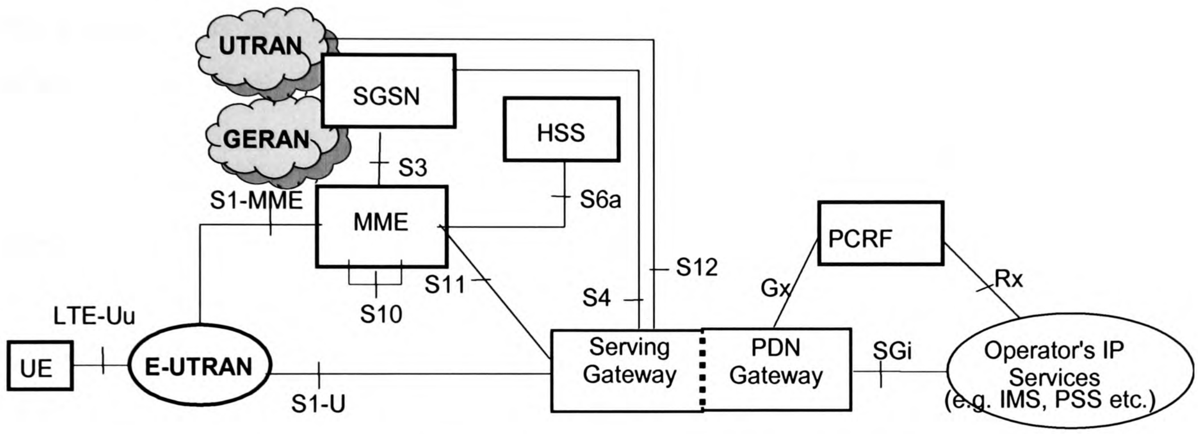

Figure 2.17 Non-Roaming Architecture of 3GPP RAT [34]

In Section 2.3.1, the existence of Iur-g interface has been discussed to enable the optimization of radio resources in GERAN and UTRAN without the involvement of the core network. In LTE there are two signaling interfaces connected to SGSN, namely S4 and S3. S4 is connecting the SGW to SGSN for user data transfer and signaling in connected mode. It provides related control and mobility support between GPRS Core and the 3GPP Anchor function of Serving GW. In addition, if Direct Tunnel is not established, it provides the user plane tunnelling. S3 connectes MME to SGSN and It enables user and bearer information exchange for inter 3GPP access network mobility in idle and/or active state [34]. For supporting interoperation between GERAN/UTRAN and E-UTRAN, when the SGSN interfaces have not evolved to S3/S4, Gn/Gp interfaces are used, Gn is between MME and SGSN and Gp is between P-GW and SGSN or GGSN and the use of GPRS tunneling protocol is invoked [13]. Figure 2.17 shows the signaling interfaces when the core network elements and the Radio elements are under the control of one operator. Other signaling interfaces have been standardized for roaming users; S6a interface connects the MME entity to the HSS in the visiting network which enables the transfer of subscription and authentication data between the two entities. S8 is another interface used as inter networking reference point providing user and control plane data between the Serving GW in the visiting network and the PDN-GW in the home network [34]. Nevertheless, some sort of inter-networking between 3GPP RATs is going to take shape. Early deployment of LTE is offered at low prices in Norway, to boost the customer base. The inter-networking of GERAN/UTRAN/LTE as they are deployed in the same cell site will enhance the radio resources of the network. However the management and optimization of these resources is far from easy task especially when it is complicated with different service requirements and customer mobility.

2.7 RAT Deployment

Operators have gained an extensive experience from GSM as the network is mature and optimized and has an advanced architecture through deployment of macro, micro and indoor cells where all communicate and interact together. This hierarchical cell structure enabled operators to manage their networks efficiently [8].

Similar experiences have been gained through the deployment of 3G (WCDMA) air interface systems. As such, coverage and capacity plays a major role in WCDMA deployment. More WCDMA cells are needed for each GSM coverage cell; however WCDMA improves the spectral efficiency as the whole spectrum is allocate for users data rates separated by SF. Deploying WCDMA on the existing GSM networks has its advantages, there is no need for acquisition sites and the operators experience with traffic densities on GSM has helped in initial deployment as it makes more economical sense to the operators and it is easy to implement. With the potential introduction of LTE, coexistence of GERAN/UTRAN and LTE becomes critical and therefore further studies have been conducted by 3GPP RAN Working Group 4 [35]. The studies showed that the effect of LTE RAN deployment in an existing GERAN/UTRAN site is minimal. UMTS throughput loss is around 3 percent when considering the total interference caused by transmitter to the adjacent channel receiver, Adjacent Channel Interference Ratio (ACIR) and the transmitter mean power, Adjacent Channel Leakage Power Ratio

(ACLR). ACIR is dominated by ACLR [36-37]. LTE coexistence with GERAN/UTRAN is very important because there is no dedicated frequency band assigned for LTE. From a geographical location deployment, it is obvious and common sense the economical benefits of sharing sites with other RATs.

2.8 Quality of Service (QoS)

The quality of service architecture as defined by 3GPP is depicted in Figure 2.18. Providing end to end QoS at the application layer as in Figure 2.18, the user has to use the layers below to insure the correct mapping of the service parameters. The management and the mapping of the end to end services QoS to the radio access bearer services is done at the UMTS bearer service layer. The UMTS bearer service employs the services of the radio access bearer service to map the high level requirements to the radio bearer service. In turn, the radio bearer service employs the physical radio service bearer service. This management function is not standardized by 3GPP; it is implementation specific [20].

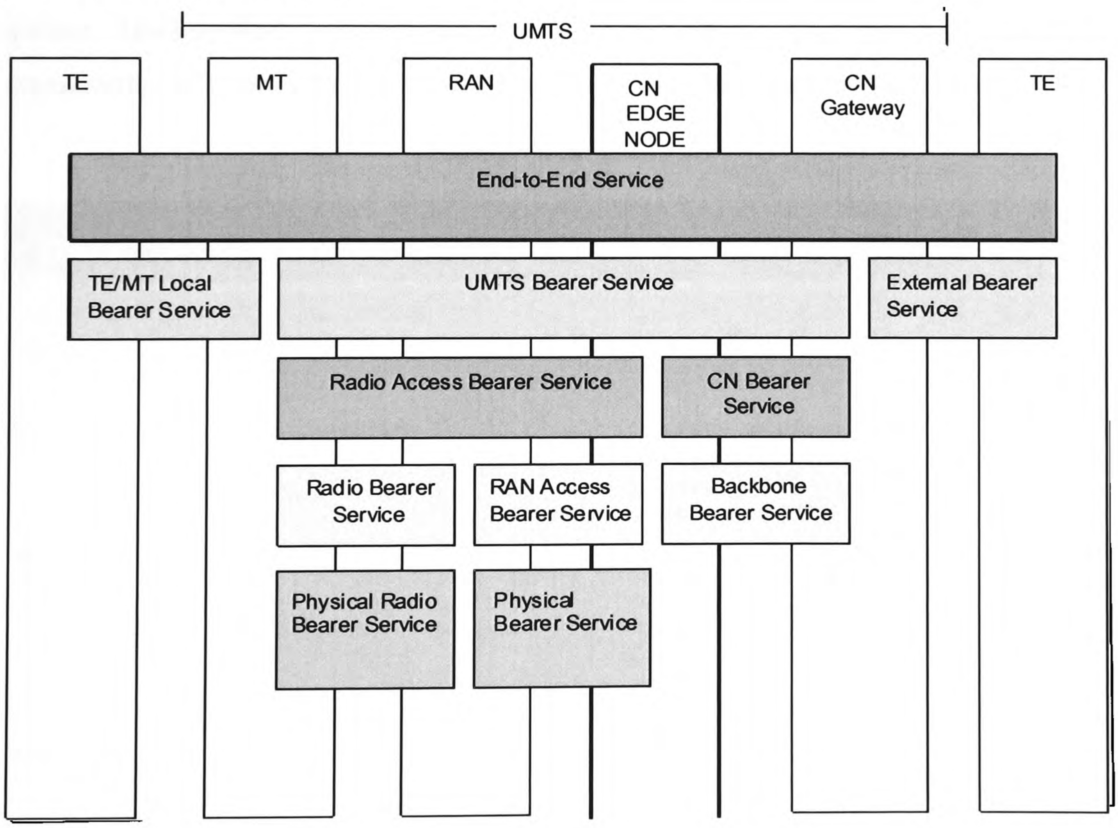

Figure 2.18 UMTS QoS Architecture

There are four QoS classes according to [36], as depicted in Table 2.7, conversational class, streaming class, interactive class and background class. They are distinguished by several parameters, but mostly by service delay sensitivity. Conversational class is prone to delays while background class is delay-tolerant class. Conversational class is real time services example of voice calls. However a new set of applications fall into delay sensitive application has been developed for the internet, to list some, VoIP, video conferencing and video gaming. The characteristics of these services are order of delivery, time correlation and delay sensitivity. Streaming class is real time audio and video downloads from streaming server. The main characteristic of this service is to preserve time relations among entities. Interactive class is a request/ response class. The requester could be human or a machine like data base retrieval, or remote login services. The request/ response have to be within a certain time but most importantly the bit error rate has to be very low. The integrity of the data transfer is more important than the time of transfer. The characteristics of this service are data integrity and time for request/ response. Background services are example of SMS messages and emails. The characteristics of those services are data integrity and time is almost irrelevant.

Table 2.7 Traffic Classes

Service Class	Service Type	Characteristics	Example of Service
Conversational Class	Real time Services	Low delay, Symmetric traffic, No buffering and guaranteed bit rate	Voice Calls
Streaming Class	Real time streams	Minimum variable delay, buffering allowed, asymmetric traffic, guaranteed bit rate	VoIP. Video Streaming.
Interactive Class	Non Real Time but Round trip delay time is a key to this service since request/response pattern	Moderate variable delay, buffering allowed, asymmetric traffic, no guaranteed bit rate	web browsing, data base retrieval, server access
Background Class	Non Real Time	Big variable delay, buffering allowed, asymmetric traffic, no guaranteed bit rate	E-mails, SMS

At the physical radio level bearer and according to [20], the QoS as a measure of (E_b/N_0) for Additive White Gaussian Noise (AWGN) channel is given in Table 2.7 below:

Table 2.8 Performance requirements in AWGN channel

Measurement channel	Received E_b/N_0 For BS with Rx diversity	Received E_b/N_0 For BS without Rx diversity	Required BLER
12.2 kbps	n.a.	n.a.	$< 10^{-1}$
	5.1 dB	8.3 dB	$< 10^{-2}$
64 kbps	1.5 dB	4.7 dB	$< 10^{-1}$
	1.7 dB	4.8 dB	$< 10^{-2}$
144 kbps	0.8 dB	3.8 dB	$< 10^{-1}$
	0.9 dB	4 dB	$< 10^{-2}$
384 kbps	0.9 dB	4 dB	$< 10^{-1}$
	1.0 dB	4.1 dB	$< 10^{-2}$

The minimum QoS requirements for the different multi-path fading channels are presented in [20] as well. Mapping UMTS bearer services to physical bearer service in accordance with [20], further better enhances the end user experience and utilizes of operator network elements. Parameters for the physical bearer service can be traffic class, maximum bit rate, guaranteed bit rate, delay, bit error rate and allocation/ retention priority.

2.9 Mobility and Handover

Handover is the process of maintaining session Quality of Service (QoS) while the customer is changing radio access network point of attachment. Handover due to mobility, while seamlessly maintaining service continuity and QoS is one of the 3GPP standard requirements. Handover is maintaining service continuity to a mobile station while moving from one coverage area to another associated with different base station or Node B. There are two types of handover, horizontal and vertical. Horizontal handover is the process of providing service continuity while handing the service to cells of the same technology, in other words the target cell is of the same radio type of the source cell. Vertical handover is the process of handing the service to a target cell that is of different type than the source cell. Service continuity is applicable only to any service that can be maintained by the target domain. Service continuity should be able to support voice calls within UMTS Radio Access Networks (UTRAN), within GSM EDGE Radio

Access Network (GERAN), between UTRAN and GERAN, between LTE and UTRAN/GERAN and vis versa. Moving a call/session from one system to another (Vertical Handover) requires more processing power and several networking entities will be involved than transferring a call/session within the same system. Mobility Management Entity (MME) in EPC is responsible for tracking the UE in idle mode. However in connected mode SGW and PGW are anchoring handover mobility to other RAT networks. The existing and maturity of GSM networks have a great influence on WCDMA deployment. The two systems operate at different frequency bands making the handover based on the system level and not on the frequency level. The LTE radio interface addition to the cell site will bring to the forefront many deployment scenarios.

2.10 Chapter Summary

In this Chapter, a technology overview was provided, the demand for higher data rates, bandwidth on demand services and the popularity of user equipments that are capable of handling multimedia services has led to the standardization of diverse services on Universal Mobile Telecommunication Service (UMTS). Delivery of such services was not possible using the existing GSM/GPRS (GERAN) networks because of limited data throughput of the system. This has led to the introduction and deployment of WCDMA radio air interface to satisfy the ever growing demand for bandwidth. As the demand kept rising fueled by the increased customer base for such services, the 3GPP community responded by evolving 3G functionality and the introduction of High Speed Downlink Packet Access (HSDPA) in Release 5. To stream line the service delivery, a functional split between the core network and the radio network was introduced in Release 5, this has led to the introduction of new signaling interfaces between the radio access technologies.

3GPP standards outline the signaling interfaces to facilitate interoperability among the different radio access technologies; such interoperability will present two problems to the operators. First, to efficiently utilize the diverse radio resources provided by multi-radio access technologies, the operators need to design a complicated CAC algorithm. Second, the operators need to provide service continuity through the design of handover algorithms in a multi-service environment.

Chapter 3 provides a literature overview of the proposed algorithms to provide solutions to the abovementioned problems.

Chapter 3

Literature Review

3.1 Introduction

Cellular mobile telecommunications networks objective was to deliver voice service with QoS comparable to that of wire line circuit switched telephone systems. Radio resources management for GSM was straight forward in a sense that the system has to deliver one main service for real time delivery as voice and one Non Real Time (NRT) as Short Message Services (SMS) data application. The introduction of a new set of data services by the operators led to the enhancements of radio resources management algorithms. The need for higher data rates has been fueled by the introduction of new bandwidth hungry applications and handheld wireless devices (laptops and media devices). This led to the introduction of a new set of services under the 3GPP standards in the form of UMTS. To deliver these services 3GPP standardized a new radio access technology which has been called later UTRAN (as discussed in Chapter 2). The deployment of 3G and 3.5G initially was in the same cell sites with GSM. The inter-networking between the two systems was inevitable. Network planning and deployment of GSM was straight forward being one radio access technology, homogenous network and services.

The operators collected plenty of traffic information over the years. The operators have a new set of radio resources at their disposal by deploying new RATs, complicated by a new set of services to deliver to end users. Before deploying new devices to their access and core networks, the operators need to make sure that no disruption is possible to their system. The core network is increasingly getting complex by the deployment of new RATs, modeling and simulation of cellular networks are increasingly becoming of premium value because of the system complexity. To facilitate the deployment of new devices and services, the operators revert to modeling and simulation tools at their disposal to gain insight into the performance of the new services and radio resources. The operators are investing heavily in simulation studies and design because the simulators can examine the system behavior under different deployment scenarios, identify the bottlenecks if they exist. Relatively it is a cost effective process when compared to field trials. The reason behind the simulation studies exponential growth in value is the deployment of several radio access technologies (RATs) and the wide range of services offered by this deployment in addition to the increasing complexity of the core network. The standardization and the potential introduction of the new air interface Long Term Evolution (LTE) technology to deliver broadband services further complicated the access and core network. In this Chapter, a review of the research literature regarding mobile network simulation tools, Radio Resource Management, Call Admission Control, mobility and handover.

3.2 Call Admission Control and Radio Resource Management

Radio Resources Management (RRM) and Call Admission Control (CAC) are inter-related, because CAC functionality is part of the RRM entity. The 3GPP standards do not specify algorithms for CAC algorithms as this is considered vendor specific and in general the radio resources are technology specific. In GSM for example the radio resources are the TDMA time slots where each frequency carries 8 slots/channels that are allocated to user data and/or signaling. In WCDMA this is not an efficient way of considering radio resources, since the radio resources span layer 1 to layer 3. However, scheduling the resources are divided into two types [11, 25] from functionality point of view, dynamic and semi-dynamic. CAC is considered a semi-dynamic function, however, when the call/session is admitted the resources scheduled at layer 1 and 2 in a dynamic fashion.

In [39], the authors presented an approach that considered two types of services in WCDMA, Real Time (RT) services as in voice and NRT services. The authors developed a mathematical model to estimate the number of real time connections influenced by the Adaptive Multi Rate (AMR) codec which offers eight different rates between 4.75 kb/s and 12.2 kb/s. To stop NRT services from starving for bandwidth, a reservation scheme of the total capacity was implemented. The available excess capacity is assigned to NRT in the absence of RT users or the remaining bandwidth after satisfying RT requests. The authors estimated the interference values by running simulations to measure the average interference generated by each mobile and no path loss calculations were configured. The authors only considered WCDMA in their simulation and only one node analysis was carried which eliminates the effect of other cells interference on the system. A simple mathematical model was developed and an extension for their numerical and mathematical model was also developed to include handover analysis. The authors studied the blocking probability of RT applications and the NRT service time influence on the blocking probability was presented. The authors claim that reserving some of the bandwidth for NRT can be achieved with minimal influence on real time traffic. However, this mathematical model was developed for one technology and the handover presented was horizontal handover as no other technology was considered. The lack of proper simulation tool to address the interference issues and the authors were reverting to CAC and handover algorithm numerical analysis cannot be substantiated without a simulation tool to shed light on the proposed model.

In [38], the authors proposed an adaptive Call Admission Control algorithm based on traffic priority, the algorithm has three components; bandwidth allocation/reallocation policy, CAC algorithm and bandwidth adaptation algorithm. They defined 6 priority traffic classes starting from voice with highest priority to file transfer and retrieval services with the lowest priority of 1. The authors presented three types of traffic in general, Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Unspecified Bit Rate (UBR), each one of these classes have two subclasses. Each of the six traffic classes has a minimum and a maximum bandwidth allocation. However, depending on the priority of the traffic the adaptation bandwidth algorithm decides to upgrade or degrade the service, but the algorithm is bound by a minimum value set for each connection service. The authors study was in the context of bandwidth multimedia applications where they claim that handover calls have priority over new call admission as this is consistent with the industry standards of retention of the existing calls/sessions have priority over the new attempted calls/sessions. They devised an algorithm on the assumption that the system available bandwidth can be shared and allocated based on the traffic classes they have devised. No consideration was given to the radio access technology point of attachment as the claim that system capacity can be interpreted in equivalent bandwidth and it has been assumed that the system uses fixed channel allocation. However, the services that are supported by each RAT are different and sometime they overlap. It would be inefficient to assume that all systems resources will be equated with bandwidth [39]; this might be true for packet switching systems, but will not be efficient for 3GPP technologies where a combination of circuit switched and packet switched services need to be supported. From another perspective, the authors [38] discussed the handover of calls/sessions, even though the proposed algorithm include call/session handover scheme and it has priority over new calls, this algorithm was presented in the context of homogenous system handover or horizontal handover which limits the scope of the algorithm.

At present the 3GPP standardized body is rolling out new RATs to satisfy customers and applications demands and the operators are deploying these access technologies, therefore the system is far from being homogenous. The authors used a single path loss equation to calculate the received radio signal without consideration for the environment classes or terrain. The simplification of the radio resources is an indication of lack of proper simulation tool to study these algorithms properly.

In [40] the authors proposed a resource management and CAC for QoS support in cellular/WLAN internetworking. There are two ways to interconnect WLAN to cellular networks, the tight coupling architecture, where WLAN is connected via a signaling interface to the cellular network, and the loose coupling architecture where the two separate networks connect to each other through the internet Diff. Serv. Architecture [41]. In the loose coupling they employ the DiffServ platform to differentiate between traffic classes [42]. The assumption is that there are two traffic classes, voice and interactive data, and the nodes are in overlay deployment where WLAN coverage is smaller than cellular coverage. The authors are suggesting that all data traffic is not delay-sensitive and tolerant to transmission rate variations. However, this generalization is not entirely in line with the standards [20, 36, 61], as there are data traffic types that requires stringent QoS such as video telephony and gaming services. The authors claim that if the resources of the two interfaces are jointly considered factors such as available capacity, traffic characteristics are, user mobility then higher utilization of the resources can be achieved by the operators. However, first, the proposed deployment scenario is loose coupling deployment as argued by the authors which indicates that long signaling delay is the norm therefore an indication of dropped calls and sessions. Secondly, QoS architecture of WLAN is not in line with the required cellular networks as the most advanced architecture of WLAN cannot guarantee QoS for a voice call even with the deployment of 802.11e extension of the WLAN protocol. The data throughput for cellular node was assumed to be 2 Mbps while for wireless LAN it was 5 Mbps. The proposed CAC algorithm is based on the two traffic classes and a limited fractional guard channel policy (LFG) [43] where a certain amount of bandwidth is reserved for voice handover calls was implemented in the algorithm. The bandwidth allocation for voice calls is not defined clearly and the handover mechanism for heterogeneous deployment was not considered, as LFG policy was envisioned for homogenous handover. The QoS only defined in the scope of giving priority to voice calls over data sessions, however as in [20, 61] a clear definition for QoS requirements is presented by the 3GPP standards.

In [44] the authors used SIR as a constraint for CAC modeled as a semi-Markov decision process. The new user SIR is calculated and if it is below the requirement for the service (voice/data) then the total SIR is checked for all users in that cell. The second criterion is that the network blocking probability for a certain service is bounded by a particular value. To facilitate this second criterion each node B must have segregated active number of users for each service. If the two criterions are satisfied then the call is admitted else the call is rejected. The authors only considered WCDMA technology.

In [43, 45, 46] the guard channel and fractional guard channel approaches for CAC were introduced and then a variation of the scheme was studied and analyzed. The variations of giving priority to handover while at the same time have a number of channels reserved for handover was evaluated and analyzed. Mostly, these algorithms were envisioned for circuit switched calls and homogenous networks, however with the deployment of 3G/3.5G and the potential for LTE these algorithms are not adequate.

In [47-51], the complete sharing approach and a complete partition approach has been proposed, two traffic types environment were envisioned in all the research. The capacity was treated as a bandwidth only bit rate budget. The first proposal was to allow all traffic channels a complete sharing and access to the bandwidth, then a partitioned bandwidth for voice and data traffic was proposed. To provide efficiency to the partition region a Dynamic Partition (DP) was proposed by [52]. DP reserves a portion of the capacity for data sessions and another portion for handover voice calls, also new voice calls have their share of the bandwidth, and the rest of the capacity is for shared channels. A new voice call and a new data call have access to their designated capacity and to the shared capacity. Handover voice calls have access to their portion of the capacity, to the new voice call capacity and to the shared channel capacity as depicted below in Figure 3.1.

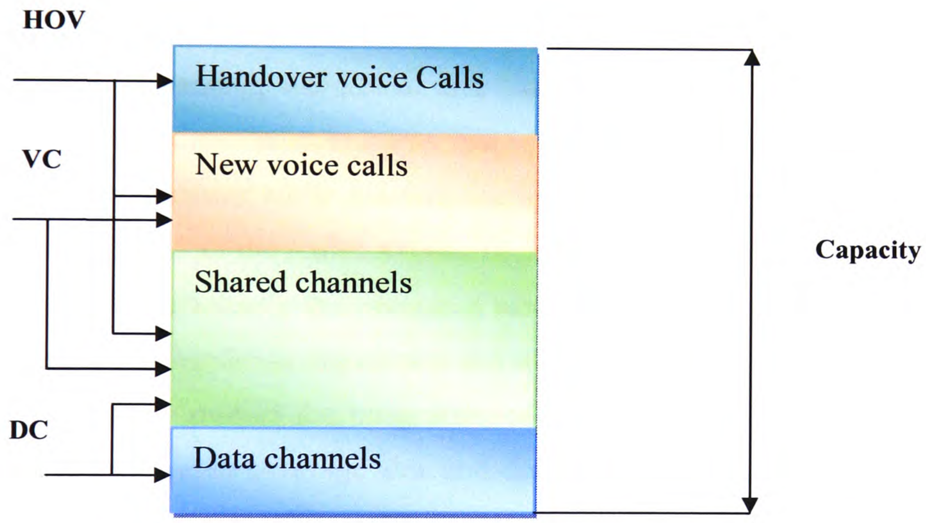

Figure 3.1 DP control Algorithm [52]

A dual threshold bandwidth reservation scheme [49] was introduced for CAC, and it is an enhancement to the proposed algorithm in Figure 3.1. The variation is that the data channels are merged with the shared channels, in this case voice and handover voice calls will have more resources to tap on, the authors in [49] claim, that their algorithm achieves a lower blocking probability percentage.

In [47-50, 52-54], the radio resources are treated as bandwidth only and the resources are divided in some fashion between the traffic classes. At the same time limiting the actual radio resources in this manner is not realistic or consistent with the deployment of different RATs in 3GPP. As explained in Chapter 2, the radio resources are different for each RAT and the traffic classes and QoS requirements [36, 61] are more complicated than what has been proposed in the above research. The introduction of a new access interface, UTRAN, and the initial deployment of this interface side by side with GERAN and in the same cell site fuelled academic research in two directions, first, Common Radio Resource Management (CRRM) because the two interfaces are under the control of one core network which also bring about other research issues in the form of load balancing. CRRM has been standardized by 3GPP, one of the requirements of CRRM algorithms is to enhance the trunking gain for real time services and lower the delay for non real time services [2]. Based on the service requested by the customer, the network operators need to envision policies to maximize active customers' satisfaction and minimize blocking probability of new customers through the efficient use of radio resources. These traffic services can be grouped into four main categories; conversational, streaming, interactive and Best Effort. The objective of CRRM is to manage the network resources in a way that insures the Quality of Service (QoS) for the existing calls or sessions while accommodating new call arrivals through coordinating the pool resources of all the Radio Access Technology (RAT) involved.

Second, to utilize efficiently the two radio resources, inter-system handover has to take place, as in vertical handover algorithms and studies.

In [55] the authors studied the integration of 3G and GSM/GPRS networks assuming that they have the same coverage size and deployed in the same cell site. The authors studied the data rate conversion, multiple data rate multimedia services and user mobility impact on the dwell time in each system. The number of time slots in GSM/GPRS systems is equated by the number of code channels needed for a comparable data rate. The two systems were assumed to have the same coverage area and the total system was in equilibrium homogenous state. There was no distinction between QoS traffic classes as two classes of traffic were considered voice and data where data traffic was subdivided into 11 subclasses. The proper data conversion was needed to facilitate channel requirements in inter system handover. They concluded that the system integration performance depends on the traffic conversion between the two systems and the channel holding time of the multimedia applications. The handoff failure depends on the resource availability in the neighboring cell and also on the dwell time the network takes in responding to the request. As mentioned above, the authors assumed homogenous deployment and equilibrium system state and they assumed that all channels are GPRS data type. However, this is far from the real deployment scenarios of 3G and GSM/GPRS as each deployment has its own limitations and requirements. Another issue is that WCDMA technology coverage is a complicated task in the uplink because of interference as discussed in Chapter 2. The need for proper simulation tool is apparent.

In [56], the author devised an architecture for the CRRM entity based on that different RATs coexist in the same cell sites in deployment of next generation networks. Each CRRM entity can manage one or more RRM in the network depending on the deployment scenario. CRRM entities can exchange information with RRM and CRRMs can exchange information among themselves. However, as far as decision making is concerned, the CRRM role might be as an advice entity and leave the decision making to RRM unit. Three solutions have been proposed for radio selection in this architecture. First, serviced based RAT selection as proposed in [57]; the authors proposed two scenarios for admitting traffic to GERAN and UTRAN as follows:

- The first scenario is to admit the voice users to GERAN and interactive users to UTRAN.
- The second scenario is to admit voice users to UTRAN and interactive users to GERAN.

The authors' findings were that the policy of allocating voice users to GERAN performs better in terms of total aggregate throughput, as they claim that in GERAN, interactive users are subject to scheduling. However, the authors did not consider in their study the UE capability, as some UE are not capable of accessing UTRAN. Moreover, the study was run only on two traffic classes, voice and interactive data, this case study approach limits the services offered by the operators as well as there was no flexibility to move to other RAT in case of load over flow. Second, the proposed radio selection was Load Balancing (LB) approach, where the load is distributed evenly in the RATs. UTRAN load can be measured; as far as GERAN is concerned, the average amount of time slots utilized can be deduced from the cells by measuring the slot busy time. This load metric can be a major factor traffic distribution as presented in [58]. However, LB approach and initial service class RAT selection are interdependent when traffic mix changes and at times LB does not enhance traffic performance. Third, the proposed radio selection policy was based on radio network selection, which depends on the user location if indoor or outdoor. It has been reported in this approach that indoor WCDMA capacity is highly degraded by indoor traffic as such indoor users are allocated to GERAN while outdoor users are allocated to UTRAN.

In [2, 59, 60], common radio resource managements algorithms are studied. The selection criteria for the mobile are on the bases of the RAT point of access, indoor coverage and load of the cells. Two classes of traffic are considered voice and data. The distinction between voice traffic carried on GSM and the voice traffic carried on UMTS was realized as an initial parameter. Services and service parameters are specified by 3GPP [4, 61]. The services envisioned by the above research only deals with partial list of the supported services as per 3GPP standards.

MATLAB was used in evaluating the above proposed algorithm. Even though MATLAB simulator is widely used and recognized academic tool, it is not the proper tool for such algorithms because of the lack of mobility and the static nature of MATLAB, that is why it is deemed not suitable for such studies. This brings to the forefront the lack of detailed simulation tool to properly evaluate such algorithms.

The following section presents an overview of the research on the handover process.

3.3 Inter-System and Intra-System Handover

One of the mobile cellular network operator's objectives is to provide ubiquitous service to their customers. Maintaining service connectivity, while the end user mobility pattern is unpredictable, is a challenge facing the network operators. The functionality of maintaining service continuity while the end user is moving from one cell coverage area to another is called handover. Handover can be due to mobility, load balancing or because of a different RAT is more suitable to support the required services. The QoS for the connection should be preserved or no measurable QoS degradations shall be experienced by the end user because of handover as long as the user stays within the UTRAN coverage area [62]. Radio Received Signal Strength (RSS) used to be the deciding factor for call handover based on the reported channel quality, because the networks used to be homogenous. Heterogeneous networks research started by the deployment of WLAN in the same area as cellular networks since WLAN can be deployed in hot spots while cellular nodes used as an overlay network since it has a better coverage, but as discussed above the lack of QoS compatibility between the two radio systems was a major obstacle in achieving full integration.

In [63, 64] the authors devised a three stage process for handover. First, initiation or handover decision based on certain measurements, for inter RAT, the UTRAN notifies the UE of which events should trigger the UE to send a measurement report [65], the UTRAN uses these events for handover evaluation function. For GERAN, these measurements could be the Received Signal Strength Indicator (RSSI) while for UTRAN it could be energy per chip divided by thermal noise (Ec/N0) or the received signal code power (RSCP). There are four inter RAT measurement events available in the standards, Event 3a means the estimated quality of the currently used UTRAN frequency is below a certain threshold and the estimated threshold of a different system is above a certain threshold, Event 3b which means the estimated quality of other system is below a certain threshold, Event 3c, which means the estimated quality of other system is above a certain threshold and finally, Event 4d, which is change of best cell in other system. For UTRAN the measurements can be of the downlink RSCP while for GSM it can be for RSSI. In the context of handover, the cell that holds the radio resources for the call is called the source cell while the cell that is identified to handover the call to is called the target cell. Establishing and reserving the required radio resources on the target cell is the network responsibility.

The second stage of handover is to transfer the existing radio link from the source cell to the target cell [64]. The authors considered the third stage of establishing the radio resources for the transferred handover calls/sessions as part of the CAC process; this is consistent with the industry standards as handover calls/sessions have priority over new calls/sessions trying to access the system. In the presence of several RATs at the same cell site, using RSS as the sole measurement and criterion to trigger handover will firstly, limit the network ability to move users in case of congestion. Secondly, this limits the user selection of the network point of attachment as it assumes only one RAT available. Thirdly, it is assumed that the network is homogenous and common signaling will suffice for the handover. The authors in [64] devised a metric function to evaluate whether a handover should be considered and refrained from the traditional received signal strength (RSS) measurement as a sole factor. They complemented RSS by several other factors which include service type, monetary cost, network bandwidth and load, mobile speed and network reliability i.e. the measured BER or BLER. The authors designed a vertical dynamic handover algorithm with network elimination in a multi network environment. Based on the above metrics the algorithm ranks these metrics as per their importance to the user, such as cost and to the network, such as if the required QoS can be guaranteed. The authors assumed three cells with varying bandwidth capabilities and devised a cost function for each network. They simulated a mobile terminal with CBR and additional variable ABR. The simulation was run using three different methods of enhancements to measure the handover of the mobile sessions i.e. CBR and ABR, the first run considered RSS only; the second run used the cost function and the third used the cost function with optimizations. They claim that the dynamic vertical algorithm using cost function and optimization outperformed the other two. The general idea of the algorithms is genuine however the practical implementation of the algorithms needs to be modified and verified; the assumption of the availability of three networks each with certain data throughput is not a realistic approach for two reasons. First, the authors assume that present and next generation networks capacity can be represented as bandwidth only, even though that this is a valid assumption but these networks differ in the distribution of their radio resources as have been discussed in Chapter 2 and in the section above. If the authors meant that the networks are of the same RAT then this will not be called vertical handover as it falls under horizontal handover. Second, the assumption that there is one streaming class of CBR and ABR might be correct for all packet switched networks but this falls short because of the nature of the 3GPP networks of mixed packet switched and circuit switched services as described in [61]. This research was done in the scope of context transfer dynamic handover utilizing mobile IP, even though that mobile IP is part of the 3GPP standards in the context of packet transport, it is not part of the access technology.

In [66, 67], the authors proposed a smart decision model based on a cost function to handover sessions to the best interface at the best time when considering traditional IP networks. The authors dealt with the problem without any consideration for the radio resources differences between GPRS networks and WLAN networks. Furthermore, the establishment of the IP tunnel from the handover server to the GPRS tower has to go through SGSN or the proposed handover server should be part of SGSN. The claim that by establishing the IP tunnel then the TCP end to end connection is kept intact might be correct, however, the offered bandwidth difference will certainly be an issue regarding TCP congestion window and retransmission of packets. The claim of seamless handover is at best not explained clearly since the decision making responsibility is with the user while he is engaging with an active session. Smart decision is envisioned as it might decide based on interference not to use 802.11b in the presence of 2.4GHz cordless phones; however, this scenario is not practical when one is trying to study mobility and vertical handover. In [68], a CAC network selector was proposed for new arriving calls/sessions based on the service requested and the best possible access network to provide the required QoS. The policy based access network includes network capacity surface to provide information to the policy decision point of the residual load of each network, when a service request arrives based on the information provided by the network capacity information the policy decision point chooses the best point of attachment for the service. It funnels this information to policy enforcement point which forces the user to a particular access point as proposed in [69, 70]. The research contribution of the paper was an inter-system handover initiation using fuzzy logic. The authors propose a handover process based on BLER, number of handovers the session executed, the time the session have been under the QoS threshold and a user classification has been included (Bronze, Silver and Gold) based on the network contract or service level agreement. The handover process decides when is the best time to handover based on user service level agreement, however, the author's simulation include EDGE and UMTS networks and they claim that this algorithm can be extended to WLAN. The authors claim that there is an improvement on the number of inter-system handover in the video and web services but they have not commented on the voice service which is still the predominant service in cellular 3GPP networks. An optimized service based vertical handover algorithm cost function was also proposed in [71], building on [69, 70], the authors incorporated a network elimination process in their cost function based on service type. Therefore, if a user has multiple services attached to it, some services might be handed over and others might drop. To provide proof of concept for the proposed algorithm, the authors revert to numerical analysis as they only considered WLAN and GPRS networks and considered the radio resources of the cells as a bandwidth budget. This is a clear indication for the lack of the proper simulation tool to evaluate these algorithms.

In [64, 72, 73], the authors have designed a cost function based on several metrics as in [69-71]; customer preferences, cost of the service, service type and others are examples of such metrics. The authors claim that the proposed multi service handover decision algorithm shows significant gains in the ability to satisfy customers with multi-service sessions. Their approach was mainly concerned with WLAN and GPRS; as a matter of fact, in some cases in their performance evaluation studies they have excluded GPRS from the mix. The lack of support in the studies for 3G/3.5G and proper GSM/GPRS is visible; the reason for that is the lack of the proper simulation tool to conduct and verify the proposed algorithms. Several research papers focused on vertical handover based on context aware concept and controlled by the mobile terminal as in [75]. Three phases of handover management algorithm were suggested, information gathering, handover decision and handover execution. The handover initiation stage is implemented using three levels fuzzy logic for received signal strength (RSS), Bandwidth availability, network coverage and velocity on the basis of weather handover is needed or not. The proposed algorithm claim is seamless handover execution based on mobile IP functionalities, while the handover decision was based on context awareness concept. The scheme gives the flexibility to the mobile terminal to check if handover is needed, as most of the complexity processing are delegated to the mobile terminal, the authors argue that the complexity has to move from the network side to the terminal. When evaluating their algorithm the authors used one WLAN node and one 3G/UMTS node and the study was done using numerical analysis. This clearly shows the lack of the existence of an appropriate simulator to study and properly evaluate the algorithm.

Most of the research concentrated on performance evaluation on one 3G/UMTS or one GPRS node and one or two WLAN nodes as in [68, 76, 77] , while the traffic classes envisioned are voice and data classes in general without any specificity to the need of each traffic class. The cost function and all its variances cannot be verified without proper simulation tool to include GERAN, UTRAN and LTE since these networks are supposed to be deployed initially in the same cell site. However, in [77], the authors proposed a fuzzy logic that takes multi criteria parameters using multi system dynamic event-driven simulation tool that is based on OPNET platform and the study extended only to UMTS Release 99 and WLAN nodes.

In [78], the authors proposed a cross layer approach, where layer 2 and layer 3 can exchange information and signaling regarding data transmission and adaptation. However, they proposed the extension of layer 2 capabilities to include routing and re-computing of routes. Layer 2 will notify layer 3 if there is no routes available and will provide signaling to layer 3 of available nodes, their bandwidth and stability. However, this is layer 3 functionality; it was not clear what the authors are proposing because if the functionality of layer 3 moves to layer 2 this is a redundant computations. Another cross layer design was introduced in [79] where the authors proposed Inter-RAT handover between UMTS and WiMAX, they designed a new TCP proxy layer that interacts with the internetworking sub layer in WiMAX.

In [80], the authors devised a load balancing algorithm based on radio resources management that causes several elements to influence the operators decision of a vertical handover. Four stages algorithm has been devised, triggering mechanism, identification of users/ sessions that can be handed over, selection users/session to move and a migration process. The initial triggering event might be that a new admission is requested while the resources are exhausted and there is no possibility of degradation of a certain services. The identification stage is to recognize the users/sessions that can be relocated to a different interface based on, for example, position of the user, interface capabilities, application sensitivity and other factors as well. The selection stage is the main process of selecting the users/sessions to move based on the triggering requirements. The migration stage includes all the signaling that takes place between the user/session and the network on several layers (i.e. physical, network, etc). The validation of the mathematical model was done considering two traffic classes with a certain bit data rates, using MATLAB simulator. However, first, the abstraction level of the different RATs radio resources as only bandwidth does not present a real implantation of the RAT. Second, in their performance evaluation the authors considered only two overlay nodes, WLAN and a cellular node both with certain bandwidth, which does not give a proper insight into the algorithms and how it will react to changes in the network conditions in general. The author captured the essence of the standards by assuming different RATs are supposed to be catering for different services as in [61] and the data rates depend on the service request. But they have not presented a complete algorithm which includes all services and other RATs standardized by 3GPP. The authors also assume equilibrium network and homogenous radio resources of the two networks and that is a common assumption also in the previous presented research. However, the resources in cellular mobile networks depend on the technology as has been presented in Chapter 2. Packet switched networks has its roots in the legacy data networks while circuit switched networks has its roots in the land line networks. Even though load balancing is used by the network operators to efficiently utilize their networks, but the resources consumed by different technologies is not the same. For example, a voice call might occupy a time slot in GSM/GPRS in full rate assignment, to move the same call to 3G system; other issues need to be addressed for that call, such as interference, spreading code availability, power consumption and mostly the capability of user equipment. These issues were never addressed in the research above except in its generalities when user equipment capability is an essential part of a successful handover process regardless of phases or stages of each algorithm.

In [81], an intersystem handover based on NRT service content was patented. The NRT can be WWW or WAP request. The UE is in dual mode 2G/3G, so when the request arrives at the SGSN, the SGSN can determine the Packet Data Protocol (PDP) context of the request and consults with the GGSN, which is envisioned as connected to two switches; Web switch and WAP switch. Based on the URL requested, the GGSN route the request. An inter-system handover algorithm based on RSSI for GSM and (Ec/N0) for UMTS was presented and evaluated in [82]. The algorithm basically defines two thresholds for the mobile user, if the user in GSM node falls below the RRSI threshold then it will look for another GSM nodes in the area, if no GSM nodes available then it will look for UMTS nodes, if there is a UMTS node available and the (Ec/N0) is above the threshold for a specific time then the call is moved to UMTS otherwise the call stays with GSM until the quality of the call degrades and eventually the call is dropped. The same is done if the call is in UMTS except now the first measuring factor is (Ec/N0). A handover performance was carried using out MATLAB simulator considering only voice calls with speeds of equal probability ranging from 3.6, 18, 36 and 54 Km/h. the simulation was carried using only two 3GPP technologies and one traffic service in voice. This might be sufficient for such deployment but as the envisioned of coexistence among other RANs as in HSDPA and LTE as well as the deployment of different traffic classes, thus this performance study is limited in its real deployment applications.

In [83], the authors proposed two algorithms for network inter operability between different radio access technologies, first algorithm for initial user system selection and the second is for inter-system handover, each with different cost function. The two algorithms were evaluated using system level simulator, using several parameters as triggering mechanism for the algorithms. The cost functions may include mobility, traffic load, terminal type and user preference. When evaluating the algorithms, the authors claim that initial selection of the system is independent of the network load and this makes their algorithm more robust to all traffic conditions. They concluded that initial user selection of radio access technology distributes the traffic to both networks and this makes the network more efficient in distributing traffic. They also concluded that a significant a achievement in inter-system handover drop rate was reported. The authors considered only two traffic classes FTP and web browsing and did not take in account the 3GPP standards traffic (i.e. conversational, video, interactive and best effort). A second issue with the authors approach is in inter-system handover algorithm; the QoS requirements for the proposed services, when both services are non real time services, such as FTP and web browsing are different. Since FTP traffic requires better reliability than web browsing. Another issue arise as it has been discussed earlier in this section is that the QoS architecture for HSDPA and WLAN are not compatible.

Monte Carlo static simulator is used in [84] to evaluate the deployment of WCDMA and HSDPA in macro and micro cell environments. The authors concluded that deploying

WCDMA in macro cells while deploying HSDPA in micro cells scenario has a better overall performance than other scenarios. However, because of the limited capabilities of the simulator used in the evaluation, handover was not evaluated properly in the research.

3.4 Simulation Tools

Simulation tools are used in complex systems to provide a guide for network engineers on the behavior of the network in general when a new algorithm or a device is added or will added to the system. With the complexity of the cellular mobile networks as described in Chapter 2, the simulation tool can provide rich information and statistical data for the operators to enhance the performance of the network and to enrich the process of decision making. Simulation tools are of two types, dynamic and static simulators. In static simulators, the users are generated randomly with their capacity demand and the users are randomly distributed over the geographical area based on user density information. The users get serviced, and each snapshot will have new users' position depending on random speed attached to the users. Static simulators need to run as many snapshots as possible to reach a converged state. Static simulators carry out the analysis based on one common network configuration and traffic distribution. There are no time correlations between the simulation events. The loss of dynamic information can have an impact on the final statistical results. The advantage of static simulator is that the statistical results are very easy to read and analyze.

Mobility and handover, which are essential for any wireless telecommunication systems, will be lost in static simulators. To compensate for lack of dynamic information, the use of short term dynamic simulations has been proposed in [85]. The main concept is to load the snapshot and to analyze the snapshot for a predefined amount of time. Then a new independent snapshot is loaded and for a period of time the dynamic effects (fast fading and power control) are analyzed.

Dynamic simulators capture the time correlation details between events which make it more suitable for telecommunications systems. Dynamic simulators are most attractive when analysis of mobility and handover is required. The drawback of such simulators is that a lot of information is generated from each run and therefore the storage requirement is too big and the analysis is time consuming.

Simulators are also of two types: link level simulators and system level simulators. Link level simulators allow the analysis of the physical feature of the link path loss and the quality of service based on certain bit error rate or block error rate. On the other hand, system level simulator analyzes the system behavior and system performance. A system level simulator simulates actual access network procedures like CAC, handover and mobility. Link level simulators are used as input parameters for system level simulators.

The system level simulator that has been designed in the context of this research is a Discrete Event Simulator (DES), more details are provided in next Chapter (the system) and in [86]. DES is a simulation method where each event is scheduled with time instance in the future and the outcome of the event execution will influence the system behavior.

In [87] the authors proposed a discrete event simulator that covers E-UMTS.

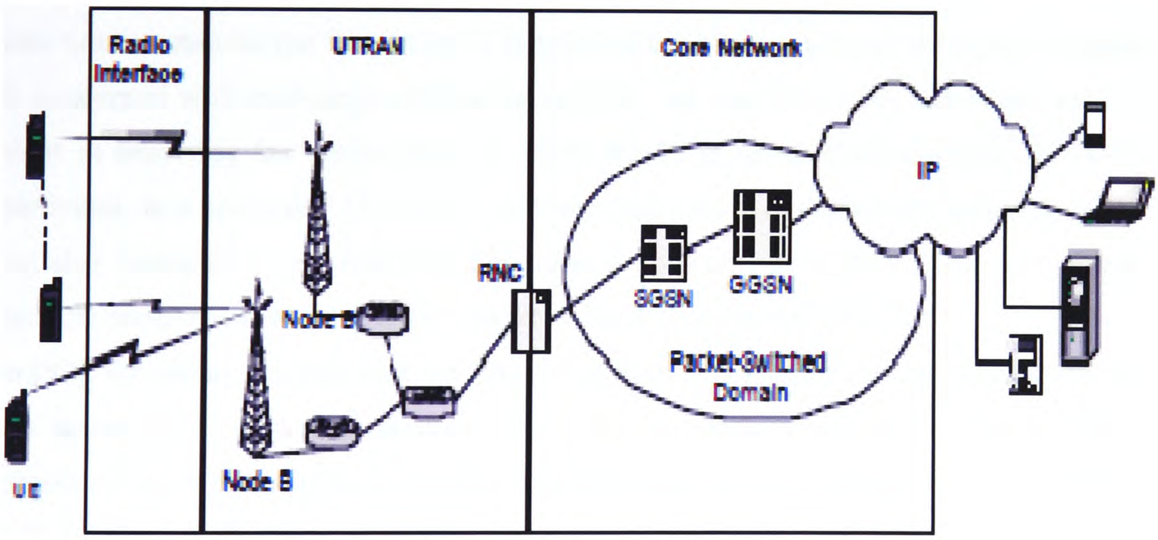

Figure 3.2 Enhanced UMTS network topology [98]

The authors in their approach considered the UTRAN technology and at the same time considered the packet switched domain of the UTRAN. The generation of users and traffic is not done in real time and only three types of nodes; Pico-cellular, micro-cellular and macro-cellular are simulated. The distance between the nodes is set in advance. Mobility and handover are considered in the same cell type which implies only horizontal handover is studied but not vertical handover. However, as discussed in Chapter 2, UTRAN has two Iu interfaces, one connected to SGSN and the second is the Iu-CS which is connected to MSC for circuit switched services. Radio resources are not clearly defined or explained. This shows clearly the need to design a modular scalable simulation tool as in Chapter 5.

Modularity enables the study of interaction between different systems (GERAN, UTRAN and LTE). In [88], the authors extended their work of [87] as the simulator is an extension of ns-2 (the academically widely used simulation tool). Their investigation of the RRM is in the packet switched domain part of the UTRAN. RRM admission control mechanism is split between RT and NRT traffic. Based on the proposed algorithm the calls are checked for SIR, if the user SIR is below a specified threshold values then the call is blocked, else if the call is RT, check if resources are available then accept the call, else check if NRT resources can be free, if not, then the request is blocked. Packet scheduling is part of the simulator in the RRM functionality which allocates the resources for all traffic. Power control was introduced as a load factor and a scenario was run to estimate the frequency of power commands used. Even though the simulator is concerned with studying multimedia applications and their QoS requirements, it falls short in depicting the architecture of 3GPP RATs in general and certainly UTRAN in particular, in a sense that UTRAN is not only delivering multimedia application services but also dedicated voice channels. The proposed simulators in [88-89] are an example of specific simulators to address one aspect of the technology or services.

In [90], the authors designed a new universal discrete event simulator to answer some of the issues raised above, as the current simulators are too specific or not scalable. This simulator is not specific to 3GPP technologies as it is designed to address the deficiencies of ns-2, ns-3 and other general simulators available commercially or otherwise.

In [90, 91], the momentum projects and its deliverables included an outline for UMTS simulation procedures that included short term dynamic process (more discussion in the Chapter 5). However, their design was based on extending the static simulation to study the effects of mobility on the statistics of the simulation tool.

In Chapters 1 and 2, the different services and interfaces that 3GPP standards roll out have been discussed. Any proposed simulators have to reflect these services and radio access technologies and capabilities but not be limited to one scope of these services and technologies. This stresses the need to design a scalable comprehensive modular and scalable simulation tool as presented in Chapter 5.

3.5 Chapter Summary

In this Chapter an overview of the literature concerning CAC and handover algorithms was presented. Several algorithms to provide solutions to the CAC problem have been proposed. These algorithms considered two traffic types, namely voice and data. Service continuity and vertical handover algorithms have been proposed as well. However, these algorithms have been confined to two traffic types while the services, as standardized by 3GPP, have several traffic types and different requirements as specified in [36].

When the proposed algorithms in the literature are evaluated by the authors simulation tools, the complexity of such algorithms were scaled down to one 3GPP radio access technology or one 3GPP access technology and a WLAN. Such studies do not provide a full and comprehensive evaluation of the proposed algorithms. As an example, to evaluate vertical handover algorithms, the environment of evaluation was confined to WLAN node and GPRS node. However, this is not a realistic approach since the area coverage of WLAN is limited and a mobile user will be outside the coverage area of WLAN node before the connection is transferred. Several other numerical evaluations were done for vertical handover algorithms, which basically highlight the need for a detailed simulation tool to study the proposed algorithms.

In Chapter 4, a detailed CAC algorithm that considers the existence of several different traffic types as per 3GPP standards in the presence of multi radio access technologies deployment has been presented. Another complicated horizontal and vertical handover to provide service continuity in a multi-radio and multi-services environment has been presented as well.

Chapter 4

Devised Multi Parameter CAC and Service Continuity System Algorithms

4.1 Introduction

As discussed in Chapter 3, the algorithms proposed in [38, 39, 44] are not suitable for the current and future deployment of mobile radio access technologies. Deployment of legacy 3G to enhance circuit switched voice capacity for GSM and to provide additional packet switched bandwidth (better than GPRS) has opened the door wide to internetworking and management of the two radio access technologies. The enhancements and evolution of 3G to 3.5G through HSDPA further complicates the management of these resources and technologies, even when HSDPA is designed to satisfy packet switched services, it has been enhanced to provide conversational like packet switched as in VoIP. Other proposed algorithms as in [40, 68], considered the availability of two technologies (i.e. GPRS and WLAN or EDGE and UMTS) which is not suitable for the current and future deployment of mobile networks. At the moment, there are two radio access points of attachment in GSM and 3G with four different technologies GSM, GPRS, 3G and HSDPA.

Adding to the diversity in technologies, the requirement to support multi services as standardized in the 3GPP namely conversational, real time streaming, non real time streaming, interactive and best effort further complicates internetworking.

Several technical issues need to be answered to serve this deployment scenario and others; some of these issues are radio resource management of the different radio access technologies in the presence of multi service requests as standardized by 3GPP. Another issue is mobility and handover management, in other words, how the network reacts when the need for changing point of attachment is needed because of user mobility. This last issue of handover is rather complex, because the user should not feel any noticeable degradation in service if mobility is within the same UTRAN/GERAN. However, this might not be available throughout the operators' coverage area and some sort of service degradation is expected and accepted.

The algorithms presented in this Chapter will shed light on the core network behavior in different topology deployment and the effect of these algorithms on the existing mobile networks in general.

In the following two sections, common radio resource management and handover algorithms are presented.

4.2 Common Radio Resource Management (CRRM)

The following three issues should be considered when discussing CRRM. First, the existence of two radio networks GERAN and UTRAN at the same site complicates Radio Resource managements (RRM) as each radio technology has its own RRM scheme. This is referred in the literature as multi-access, multi-radio networks. There are UEs capable of accessing all radio technologies however some are still limited in their access capabilities. Usually the UEs are backward compatible with technologies (i.e. the UMTS UE can access GSM/GPRS technology but not vis versa). Second, several services are competing for the available radio resources, such services are standardized by 3GPP [3, 61]. Third, the existence of HSDPA service can add to the resource pool but can also complicate management. As radio resources are far from being homogenous, it is expected that some resources are more suitable to service specific requests than others, thus radio resources managements is of significant importance in such deployments. Therefore, Common Radio Resource Management (CRRM) has been standardized by 3GPP to resolve the above mentioned issues. One of the requirements of CRRM algorithms is to enhance the truncking gain for real time services and lower the delay for non real time services [2]. Based on the service requested by the customer, the network operators need to envision policies to maximize active customers' satisfaction and minimize blocking probability of new customers through the efficient use of radio resources. One of the main objectives of CRRM is to manage the network resources in a way that insures the minimum Quality of Service (QoS) for the existing calls or sessions while accommodating new call arrivals through coordinating the pool resources of all the Radio Access Technologies (RATs) involved. As mentioned above, each radio access technology has its own Radio Resource Management (RRM) functionality[2].

Several research papers have been published to study the effects of different CRRM algorithms [58-60] on the multi access networks. As mentioned in Section 3.2, the only two services the authors considered were voice and interactive services.

In the following, a complete CRRM algorithm which includes the set of services as standardized by 3GPP in the presence of GSM/GPRS, 3G and 3.5G (HSDPA) will be presented.

4.2.1 Devised Common Radio Resources Management Algorithm

The initial cell selection process and CRRM policy go hand in hand. The proposed CRRM algorithm can serve as a blue print for other algorithms; in other words the algorithm can be modified, enhanced or replaced.

If the service requested is a voice call, the algorithm will check three attributes, the UE category, the user location and the path loss of the user. The first decision making criterion is the UE. If the UE is of 3GUMTS category or better then the UE scans all 3G nodes for service and store the potential serving nodes in a list. A process of eliminations takes place on this potential list to determine the best node for the service based on the other two criterions, if the user is outside the 3G coverage area and/or if the path loss calculations is outside the maximum allowable propagation loss (MAPL) for the voice service then the node is eliminated from the potential node serving list. The potential serving node list is sorted based on the best MAPL value. Another process of elimination takes place based on the node capacity; the first node in the list that satisfies the request will be serving the user. If 3G nodes are not available then the admission immediately transferred to GSM nodes. The same process described above is repeated for GSM nodes. However, the system now checks the capacity of the nodes based on available time slots.

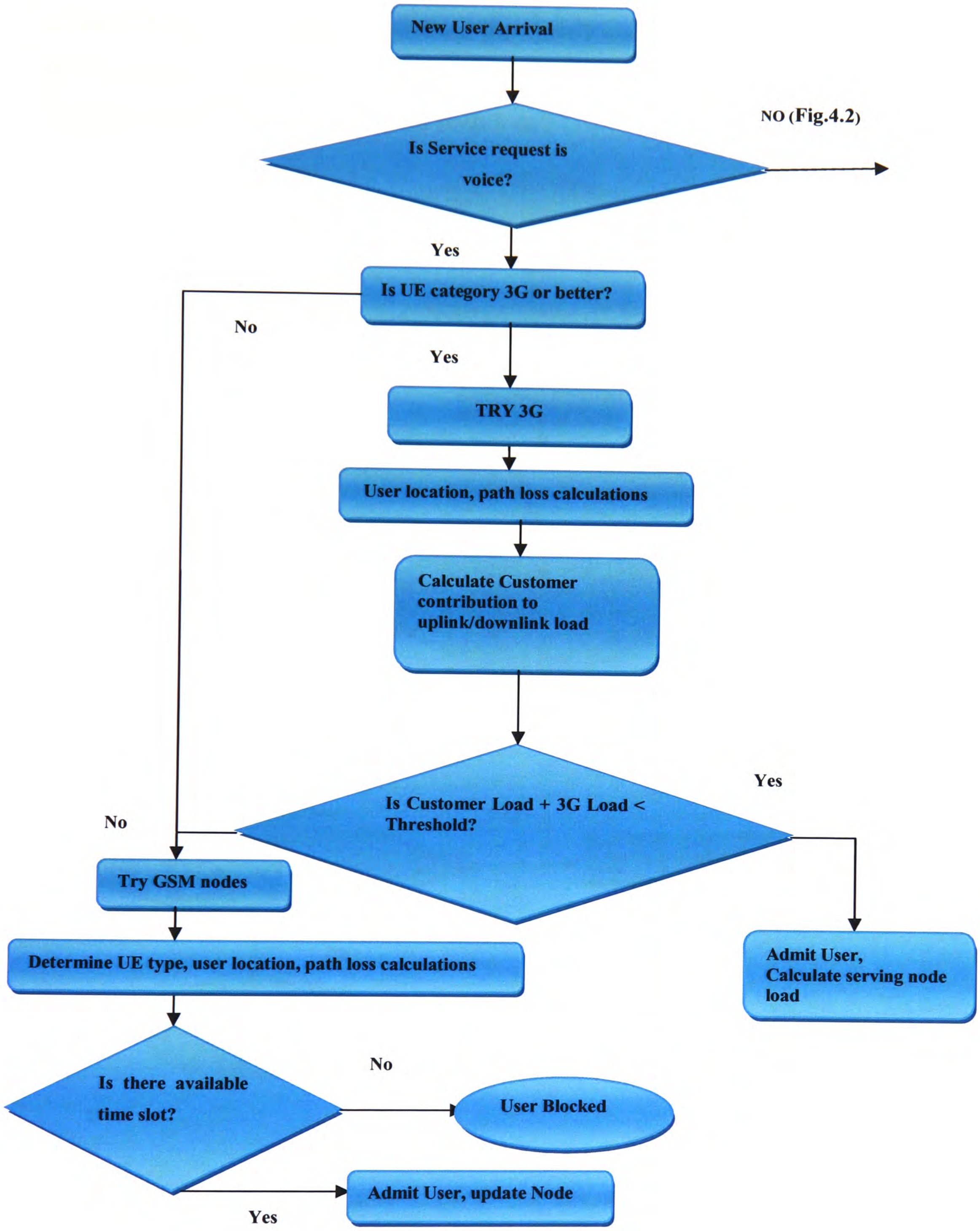

Figure 4.1 CRRM for Voice Services

If the UE is GSM only capable, then the initial selection policy is GSM nodes, if there is no available time slot then reject from the node/s and block from the system.

If the service request is streaming, the algorithm will check three criterions, UE category, the user location and the path loss.

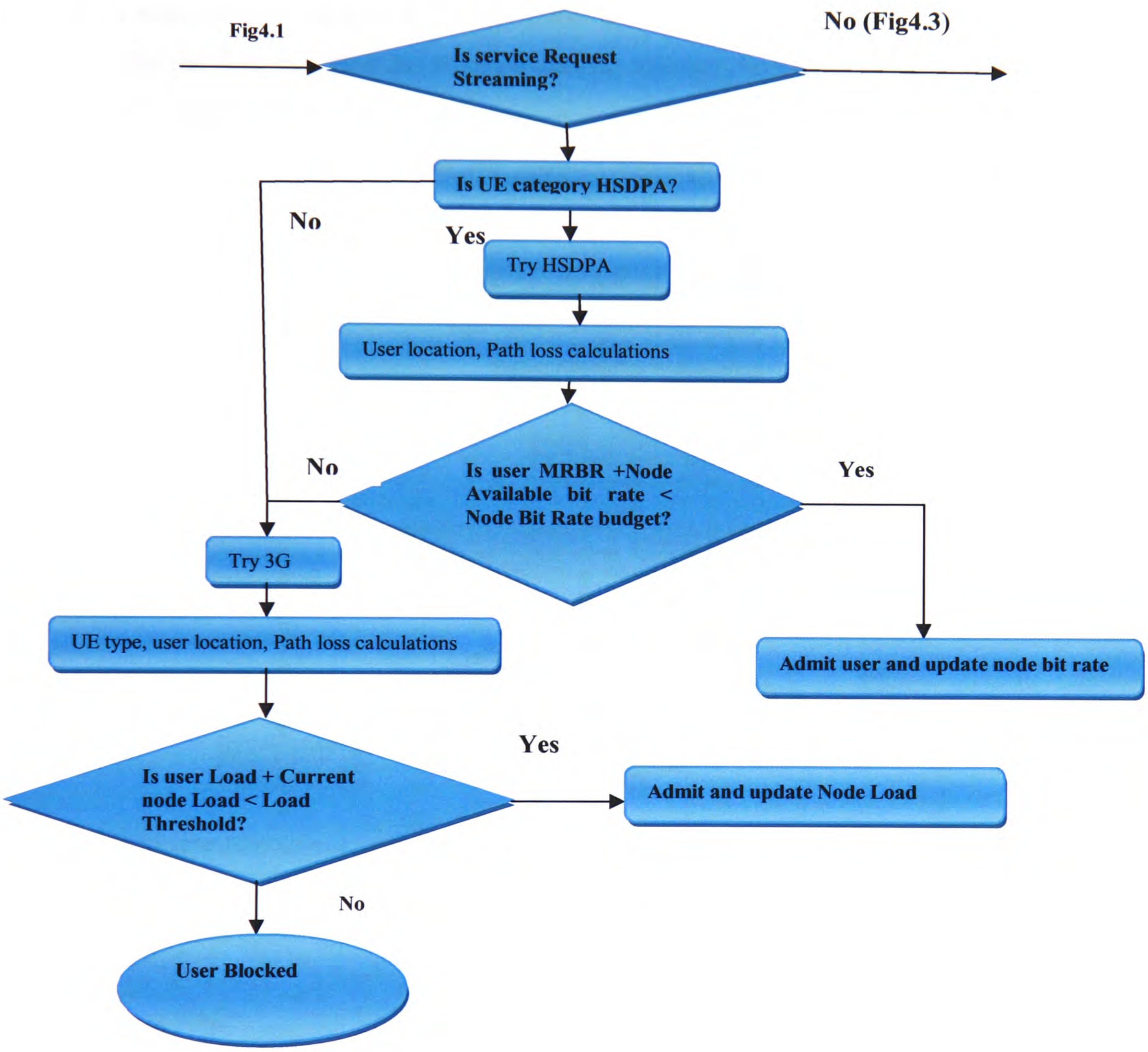

Figure 4.2 CRRM for Streaming Services

The first decision making criterion is the UE. If the UE is of HSDPA category or better then the UE scans all HSDPA nodes for service and store the potential serving nodes in a list. A process of eliminations takes place on this potential list to determine the best node for the service based on the other two criterions, if the user is outside the HSDPA coverage area and/or if the path loss calculation is outside the MAPL for the streaming service then the node is eliminated from the potential node serving list. The potential serving node list is sorted based on the best MAPL value. Another process of elimination takes place based on the node capacity; each user requesting a streaming service is assigned a minimum requested bit rate (MRBR) based on the type of streaming, the first node in the list that satisfies the request will be serving the user. If HSDPA nodes are not available to serve the user then the process is repeated on 3G nodes, however the system now check for available load on 3G nodes, otherwise the service is blocked from the system.

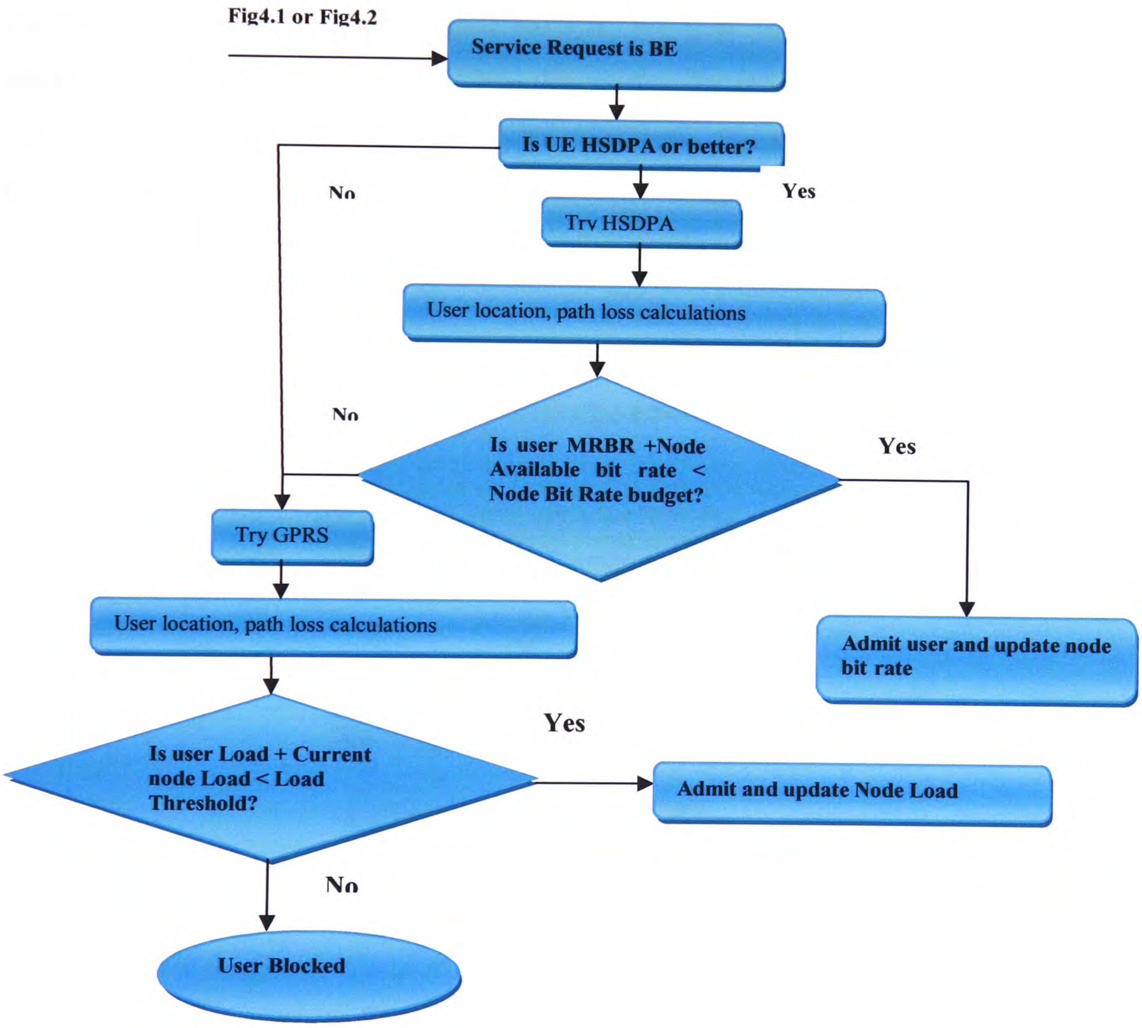

Figure 4.3 CRRM for BE Services

The final case is if the service request is best effort such as WEB browsing, the first decision making criterion is the UE. If the UE is of HSDPA category, then the UE scans all HSDPA nodes for service and store the potential serving nodes in a list. A process of eliminations takes place on this potential list to determine the best node for the service based on the other two criterions, (i.e. node coverage and path loss), if the user is outside the HSDPA coverage area and/or if the path loss calculation is outside the MAPL for the BE service then the node is eliminated from the potential node serving list. The potential serving node list is sorted based on the best MAPL value. Another process of elimination takes place based on the node capacity; each user requesting a BE service is either FTP user or WEB user, each is assigned MRBR, the first node in the list that satisfies the request will be serving the user. If HSDPA nodes are not available to serve the user then the process is repeated on GPRS nodes. However, the simulator now will calculate a new bit rate based on the geometry distribution which assigns a much lower bit rate to the users because of GPRS nodes data rate throughput, otherwise the service is blocked from the system.

In Figure 4.1, Figure 4.2 and Figure 4.3 the box referred to as 'Try 3G', 'Try HSDPA', 'Try GS' or 'Try GPRS', executes several operations as described below.

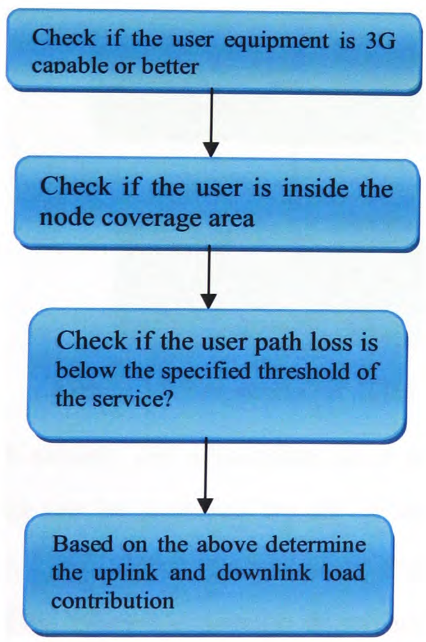

Figure 4.4 Try Access Function for 3G Nodes

For HSDPA nodes the 'Try access' is depicted in Figure 4.5 below

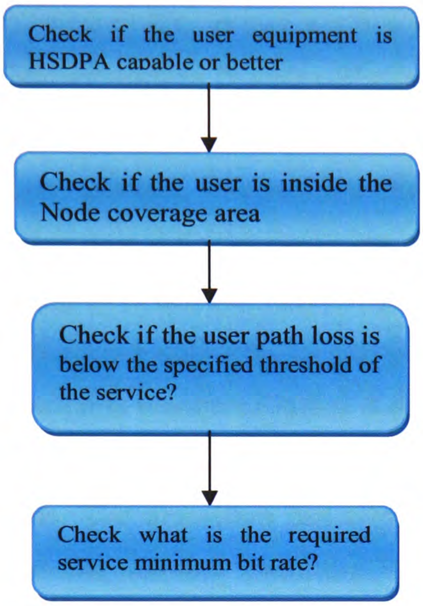

Figure 4.5 Try Access for HSDPA Nodes

For GSM/GPRS Nodes the 'Try access' is depicted in Figure 4.6 below

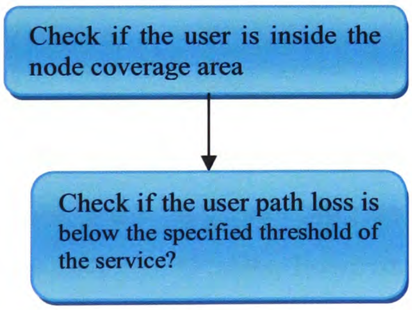

Figure 4.6 Try access for GSM Nodes

As noted in GSM/GPRS nodes, the algorithm does not check for the user equipment capabilities since this node can be accessed by all UE categories.

With each new customer, CAC algorithm based on the service requested starts with the try access functionality. The following section describes the handover algorithm.

4.3 Vertical and Horizontal Handover

Handover is the process of maintaining session Quality of Service (QoS) while the customer is changing radio access network point of attachment. Handover due to mobility, while seamlessly maintaining service continuity and QoS is one of the 3GPP standard requirements. Handover with in 2G/3G cellular networks has been successfully and widely deployed. However, deployment of new radio technologies in the mobile cellular networks has been gaining momentum. Early experience with HSDPA installations have shown improved data rate offering and opened a new era of video communications and streaming over cellular networks. The diversity of the services and the availability of two radio access networks to choose from to deliver these services make the management of available resources rather complex. In this research thesis the proposed algorithm efficiently utilizes the radio resources available based on the services supported in 3GPP standards. The algorithm has two goals: provide highest available resource to the request and maintaining minimum QoS when networks availability changes due to mobility.

In the following a complete algorithm to cover vertical and horizontal handover will be presented and studied while keeping in mind that this algorithm can be enhanced and modified to support new services as well as new radio access technologies.

4.3.1 Devised Handover Algorithm

Handover is changing of mobile serving node; this can be due to mobility or to network load balance. Handover from one cell to another requires certain measurements and thresholds set by the system, usually the received chip rate to total noise (E_c/I_0) is used as a measuring factor [8, 19] in case of 3G cells, however in GSM cells, the received signal strength is used as a measuring criterion.

There are two types of handover, horizontal handover and vertical handover. Horizontal handover is when the serving node is of the same technology of the target node, while vertical handover is when the serving node is of different technology of the target node.

The handover Algorithm has three main stages, first, the Algorithm check for all active customers in the database if the active customer is outside the serving cell coverage area, if the cell is UMTS or HSDPA, for other cell types GSM/GPRS it check if the customer distance is within the set coverage area.

If this is true, then the path loss is calculated for all cells that satisfy the customer path propagation loss threshold as depicted in Figure 4.7.

If there are no potential target cells available then the call will be dropped and the radio resources of the serving cell are updated. The potential serving cells then are updated through the second stage.

This second stage is the elimination process of the algorithm, if the cell is of the same type as the serving cell, the system checks the potential cells' capacity. If there is enough capacity for the required service, then keep the cell in the potential target list, if not then remove the cell and update the potential serving target list.

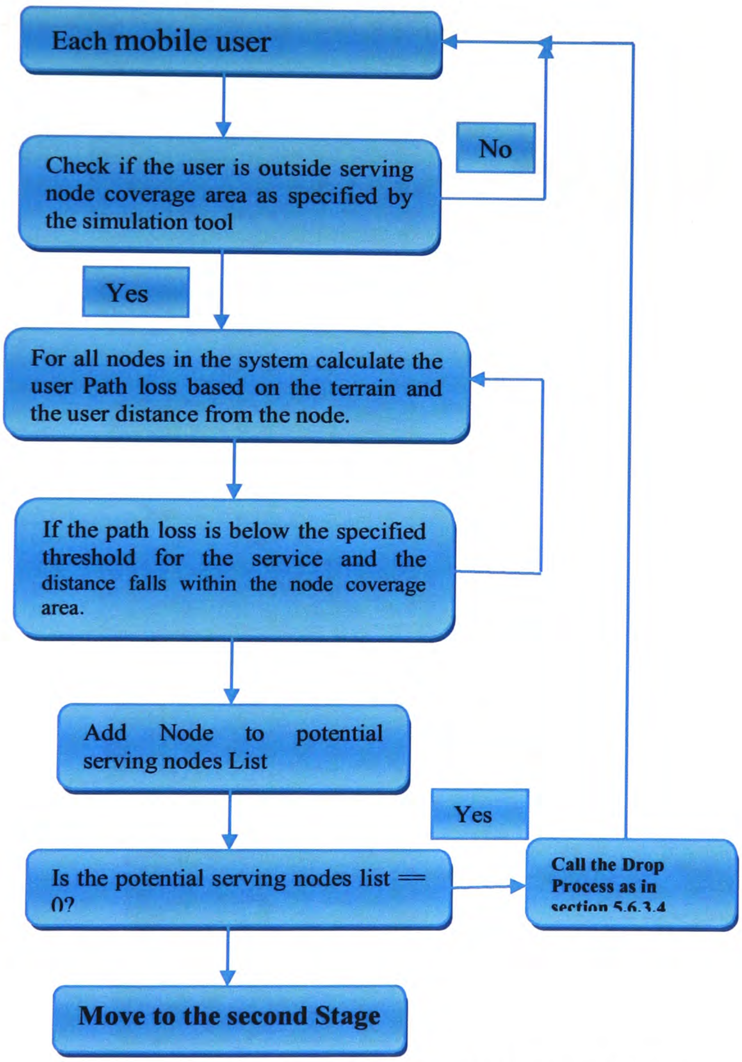

Figure 4.7 The first stage in the handover algorithm

If the cell is not of the same type as the serving cell then check if the target cell can support the customer UE category, if yes then check the cell's capacity. If both criterions are satisfied then keep in the list, otherwise remove and update the cell target list. If the cell target list is empty then the call is dropped, the customer is removed from active users and the radio resources of the serving cell are updated.

Two lists are devised, the same type cell potential target list and different type cell potential target list. Both lists are sorted based on the propagation path loss measurement of the user.

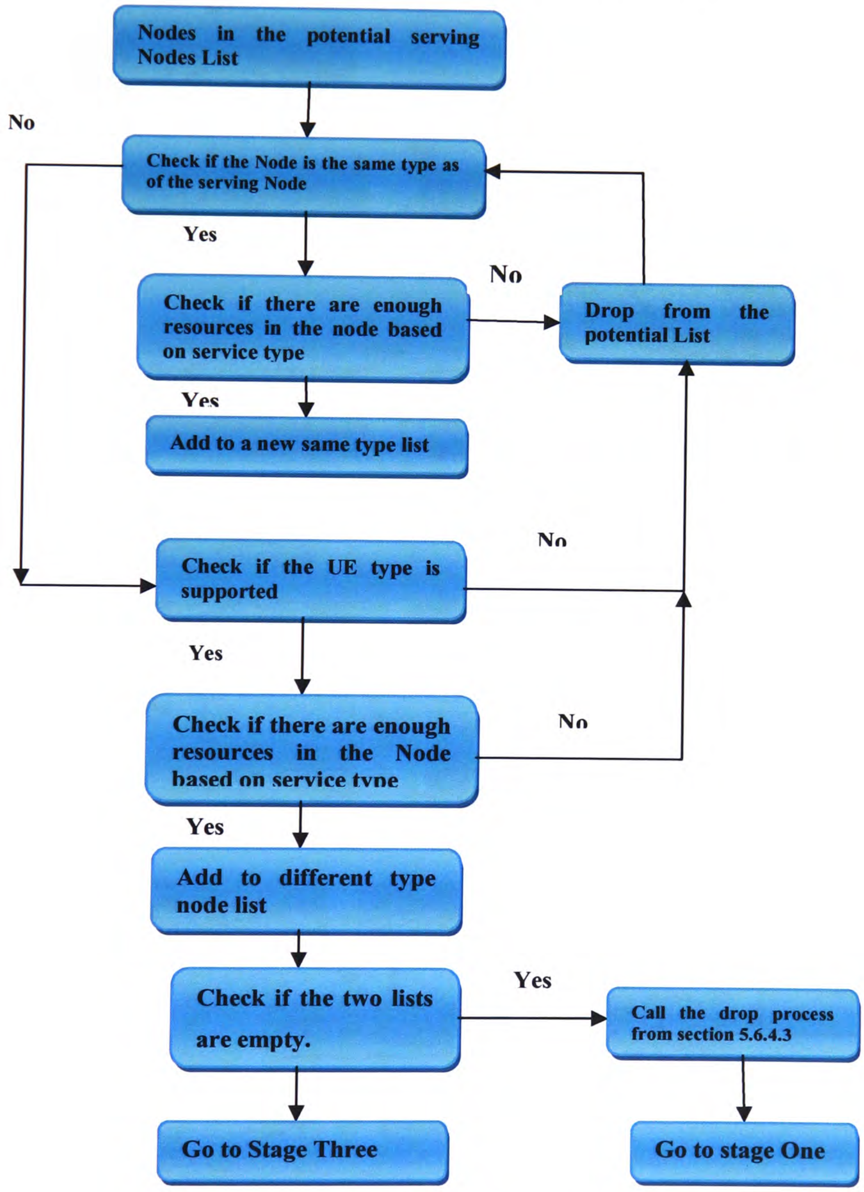

Figure 4.8 The second stage in the handover algorithm

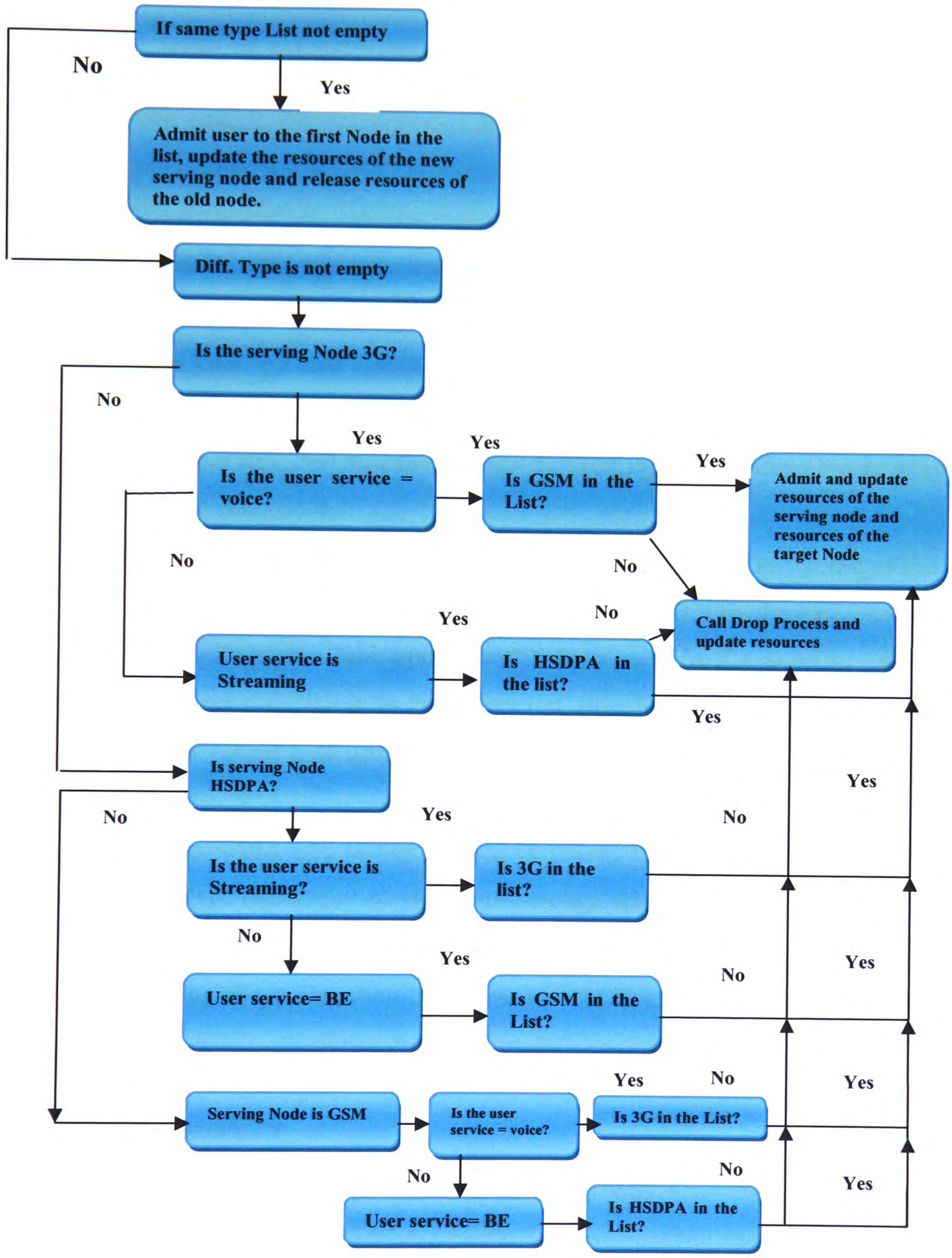

Figure 4.9 The third stage

The third stage is separated into two parts, horizontal handover execution and vertical handover execution such that horizontal handover is given priority over vertical handover.

The algorithm starts by checking if the same type list is not empty, if this is the case, admit the customer, update the resources of the new serving cell, release the resources of the old serving cell and terminate the algorithm, this concludes the handover algorithm. Otherwise the following vertical handover algorithm is executed based on service type.

If the serving cell is 3G and the user service is voice, if GSM is in the target list then handover to GSM and update the resources in the new serving cell and the old serving cell, otherwise drop the call and update resources in the serving cell. If the service is streaming and HSDPA is in the target list, then handover to HSDPA and update resources in the new serving cell and the old serving list, otherwise drop the session and update the serving cell resources.

If the serving cell is HSDPA and if the service is streaming, if 3G is in the target list then handover to 3G, update the resources in the new serving cell and the old serving cell, otherwise drop the call/session and update radio resources in the serving cell. If the service is best effort and if GPRS is in the target list, then handover to GPRS, update the resources of the new target cell and old serving cell, otherwise drop the session and update the serving cell.

If the serving node is GSM and if the service is voice, if 3G is in the target list then handover to 3G, update resources of the new serving node and the old serving node, otherwise drop the call and update resources of the serving cell. If the service is BE and if HSDPA cell is in the target list, then handover to HSDPA, update the new serving and old serving radio resources otherwise drop and update resources in the serving cell.

4.4 Chapter Summary

In this Chapter an outline of the proposed algorithms was presented. Common Radio Resource management (CRRM) algorithm has a decision making capability upon arrival of the call/session to the system; basically the decision is based on initial arrival of the requested service. The algorithm is based on the best fit radio access technology to service the call/session based on network availability and congestion. Radio access technology availability can be interpreted as; first either the technology exists or does not exist, second; are there available resources to serve the call/session.

Designing the algorithm in such a way provides a comprehensive scalable approach depending on the investigated scenario and at the same time including the criterion of best fit radio access technology to serve the request.

Service continuity is presented in the form of a complete and comprehensive handover algorithm. This algorithm also feeds from service type and subtype of the customer and at the same time the available radio access technology.

The decision making process of the above algorithms includes network availability, network load and UE capabilities.

The algorithms presented in this Chapter are modular. If the need arise to scale down or scale up the algorithm it shall be done without any difficulty. Also these algorithms are designed with scalability in mind to accommodate any new technology that will be deployed in the future on a wide scale (i.e. LTE) as discussed in Chapter 7.

This Chapter dealt with the proposed algorithms for call admission control and handover in multi radio access and multi service environment taking into consideration network availability, network load, user mobility and user equipment capabilities.

Chapter 5 will discuss the design of the proposed simulator in general and its different entities and functionalities.

Chapter 5

The System Simulator

5.1 Introduction

There are two main types of system simulation tools in general, static and dynamic. Static simulators generate snapshots based on random traffic intensity, customers are generated at random then the serving base station is assigned based on signal strength [93]. The power transmission is adjusted based on path loss calculations then the radio resources are assigned to the customers. This method is repeated hundreds of times until the results reach statistical confidence. Static simulators fall short when the need to study dynamic events like handover due to user mobility is required.

Dynamic simulators have a basic property of time correlation between events [89]. Mobility and handover have the triggering event in the past and dictate behavior in the future. This modeling of the events in the system can produce more accurate results in communications systems. Dynamic simulators can be of two types, time driven and event driven simulators, in time driven simulators the steps of the time are chosen based on known criteria for example TDMA frame time in GSM. Event driven simulator is based on the events triggered by mobility or any other event produced by the behavior of the users. An example of these events can be loss of coverage for user, arrival of new user or departure of a user among other events.

Simulators can also be categorized as link level simulators and system level simulators. Link Level Simulators used to analyze the physical radio link like channel coding, modulation and bit error rates. Link level simulations are essential in determining the quality of service requirements for a particular traffic type. It is also used as an input for system level simulators. System level simulators used to give an overview of the network behavior. System level simulators simulate the network procedures like admission control, handover and power control. The developed simulator is a system level discrete event simulator (DES) [95-97], meaning that time is only considered when there is an event to be executed. The events could be new customer arrival, customer departure, customer handover or customer dropped from the node or system.

5.2 Simulator General Description

The simulator consists of several modules, each module has it is own properties and description. There is a module for the environment, to determine the area of investigation, the propagation module to be used, coverage and capacity of cells and the terrain of the investigated area. There is a module for the user, which determines the user requested service and data rate, mobility, speed, location and user direction. This design has proven to be efficient for any investigation scenario. Any module can be disabled with minimal effects on the simulator core. The design is also scalable; any technology can be added with ease. In the following the simulator designed is explained in more details.

The basic building blocks of the simulator are shown in Figure 5.1. The simulator starts by downloading the topology and the environment (will be discussed later in details). The simulation parameters are chosen to reflect the investigated scenario as it will be discussed later. The simulation starts by generating the first event which is generating an initial arrival event at time zero. This event is generated to kick off the simulation run and subsequent arrival events are generated randomly. The random arrivals model used is Poisson arrival process where inter arrival time is exponentially distributed with mean of $1/\lambda$ time units while λ is the arrival rate of customers in a time unit. The arrival event is stored in the scheduler. The simulation fetches the first event from the scheduler, since the event is an arrival event, the arrival process is called. The arrival process (more details later) basically generates the departure event for the user based on the average required service time using exponential random distribution with service mean time of $(1/\mu)$. The customer is stored in the records database, and a departure event is created and stored in the scheduler. The simulator consults the scheduler for the next event, if the next event is arrival then new customer is created, an arrival event is stored in the scheduler; a departure event is created and stored in the scheduler. If the next event is a departure event then the departure process (details later in this Chapter) is called. Different statistics are collected and the customer is removed from the records database and the departure event is removed from the scheduler.

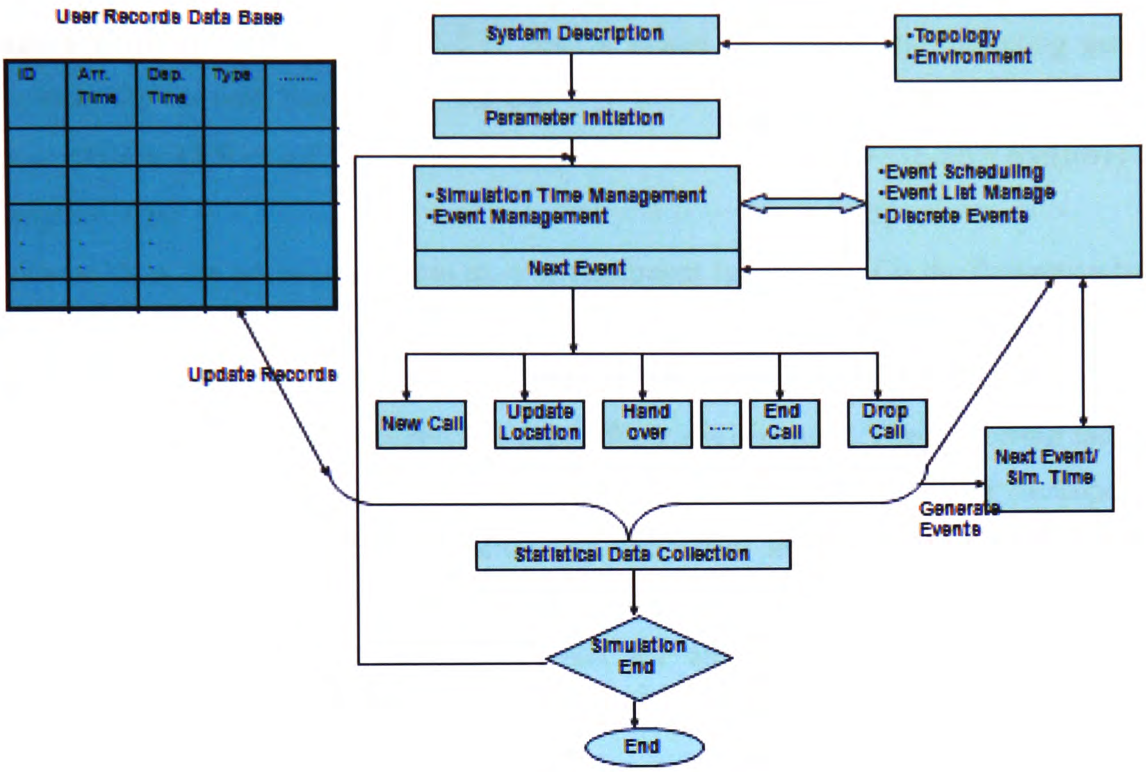

Figure 5.1 The simulator general process

The simulation checks if total simulation time has finished, if yes the simulation will terminate, if not the simulation checks with the scheduler for the next event to execute. The two main events are departure and arrival. The scheduler contains the two main events and it is the entity that manages the simulation time i.e. the next event. Many other events have been added and the simulator has to execute these events as they occur. Such events are Handover, loss of coverage, mobility and drop call/session. The other events are executed between these two events times.

5.3 The Environment

Several propagation models are used for cellular mobile networks, however the European model that is adopted for frequencies in the range of 1500 to 2000 MHz is the extended COST 231 HATA model [98] for macro cells as illustrated in Equation (1) below.

There are five different environments modeled in the simulator namely; free space, urban, suburban, dense urban and rural. The propagation loss for these environments is calculated using Extended COST-231 HATA model. In [98-99] the following general Equation for the path loss in dB is defined

$$Lx = 46.3 + 33.9 * \log(\mathrm{f}) - 13.82 * \log(\mathrm{hb}) - \mathrm{a(hm)} + \left(44.9 - 6.55 * \log(\mathrm{hb})\right) * \log(\mathrm{d}) + \mathrm{CF}x \qquad (1)$$

Where Lx is the propagation loss in, x environment type in dB, f is the frequency band in MHz, hb is the height of the base station in meters, hm is the height of the mobile receiver in meters, d is distance in kilometers, a(hm) is the mobile antenna correction factor and CFx which is the correction factor for x environment. The correction factor is equal to 0 dB for medium size cities and suburban centers and 3 dB for metropolitan areas; however as a general Equation the correction factor is calculated based on Equation (8) below. The authors of [99] derived an Equation for the capacity bounds and maximum distance range for UMTS cells for all propagation environments. The maximum distance in the uplink due to the limited transmitted power of the user equipment as in the following Equation:

$$\mathbf{d} = \left(\frac{Ps*Lpce*Kx(hm,hb,f)}{Pr}\right)^{\frac{1}{Cx(hb)}} \qquad (2)$$

Where d is the coverage distance of the cell, Ps is the transmitted power, Pr is the received power, K_x is a function of the base station height, the mobile height and the frequency as depicted in Equations (4)-(7) below, x is the environment and C_x is the correction factor for the environment as mentioned above and L_{pce} is the power correction for shadowing or slow fading. In the same paper [99], an Equation for the maximum distance has been defined as,

$$\mathrm{dmax} = \left(\frac{Psmax*Lpce*Kx(hm,hb,f)}{Nt}\left(S - (n-1) * \frac{\varepsilon*V}{F}\right)\right)^{\frac{1}{Cx(hb)}} \qquad (3)$$

where S is the service factor, which is defined in [100] as the spreading factor of the service divided by the (E_b/N_0) of the service requested, n is the number of users, έ is the interference factor and F is the inter cell interference factor and v is the service activity factor. From this Equation it is apparent that the maximum coverage distance of the cell depends on the service factor and the number of users in the cell. In other words this Equation combines the two capacity limitations of UMTS cells, limitations due to interference and limitations due to UE Uplink power. Because of imperfect power control Lpce term is included to compensate for the difference between the estimated path loss and the actual path loss. Equation (3) is used in calculation of cell maximum coverage distance in the simulator which will be calculated for each UMTS cell on the fly especially when a handover decision is needed. Kx has the following value for urban environment

$$Ku(hm, hb, f) = \frac{\mathrm{hb}^{1.382}}{\mathrm{f}^{3.546-0.11\mathrm{hm}} * 10^{4.55+0.07\mathrm{hm}}} \tag{4}$$

For suburban environment the following K is used

$$Ks(hm, hb, f) = \frac{\mathrm{hb}^{1.382}}{f^{3.546-0.11hm} * 10^{4.01+0.07hm-.2\left(log\left(\frac{f}{28}\right)\right)^2}} \tag{5}$$

For dense urban environment the K value is

$$Kd(hm, hb, f) = \frac{\mathrm{hb}^{1.382}}{f^{3.39} * 10^{5.427-.32(log(11.75*hm)^2)}} \tag{6}$$

For rural environment the K value is

$$Kr(hm, hb, f) = \frac{\mathrm{hb}^{1.382}}{f^{5.379-0.11hm} * 10^{0.456+0.07hm-.478(log(f)^2)}} \tag{7}$$

As far as the environment correction factor is concerned the propagation model specify the following Equation for Cx to be used in Equation (2) and (3) above

$$Cx(hb) = 4.49 - 0.655 * \log(hb) \tag{8}$$

For free space environment Cx = 2, and K = 1.

The area modeling is divided into grids; each grid is assigned a height (terrain), population density and allowable speed. Three categories of speed are modeled; stationary, slow and vehicle speed. Each grid can be assigned to one of the five different environments mentioned above while having its own height, population and speed. In

UMTS, frequency reuse is a fundamental concept, thus deployment of cells can lead to performance degradations and co-channel interference. Careful site planning is needed to balance between the number of active users to serve and area coverage as demonstrated in Equation (3). This imperfection in UMTS coverage can lead to coverage holes or what is known as cell breath. This is very important when designing the network as it influences the system handover decisions. Careful planning is also needed to prevent the ping pong behavior effect from taking place where users are handed over between nodes unnecessarily as this will generate a lot of signaling messages. This is also important factor when service continuity is required at the edge of the cell coverage. The developed simulator is capable of calculating the cell coverage with each active user admitted to the cell, as the service contribution to cell load and interference is calculated on the fly and the decision to admit the user or to reject the user is influenced by these calculations. In UMTS active user is allocated a code (spreading codes discussed in the Chapter 2), the user signal will be spread on the whole bandwidth at all times for the duration of the call. Each call/session is distinguished by the allocated spreading codes which are mutually orthogonal. As far as GSM/GPRS nodes are concerned, usually the coverage and capacity are determined first then the associated power and path loss requirements are calculated and dimensioned. In GSM TDMA the call is allocated a time slot that is reserved for that call using the whole bandwidth of that specific time slot for the duration of the call. The same thing can be done with LTE nodes except the radio resources are modeled in a different way as LTE is all IP broadband packet switched technology based on OFDMA.

5.4 Topology

Node coverage depends on the environment of the grids, in UMTS the simulator calculates the coverage area based on the environment and terrain, the type and location of the node is determined. For 100 points around the location of the UMTS node and for each point the average environment on that line between the node location and the end of the investigated area is calculated. Calculating the average environment depends on the terrain, the height of the grids and the height of the base station tower. Once the average environment is determined for each of the 100 points then these points are connected together to form the coverage area of the node. For GSM/GPRS nodes, an input parameter of the radius of the node is used and the simulator will draw the coverage area for the node. The difference between UMTS and GSM/GPRS nodes is that coverage in UMTS is heavily influenced by interference from users in the same node and users in other nodes. In GSM/GPRS coverage planning is assumed to be constant and coverage threshold is assumed to be the same for each node [19, 52]. In the simulator, coverage is dependent on the number of users and service required as per Equation (3) above and on the interference as explained in Chapter 2 (the noise rise). Before admitting a user/service the simulator calculate the distance between the user and the base station, if it is within the maximum distance calculated for that node then the path loss is calculated, if the path loss is below the maximum value specified for the service then the node resources is checked as explained in the arrival process below. The simulator is capable of creating as many cells as needed to satisfy the required investigated scenarios but cell deployment have to overlap in case of investigating handover algorithms.

5.5 Terrain

Coverage area in wireless networks depends on the geographic terrain of the site.

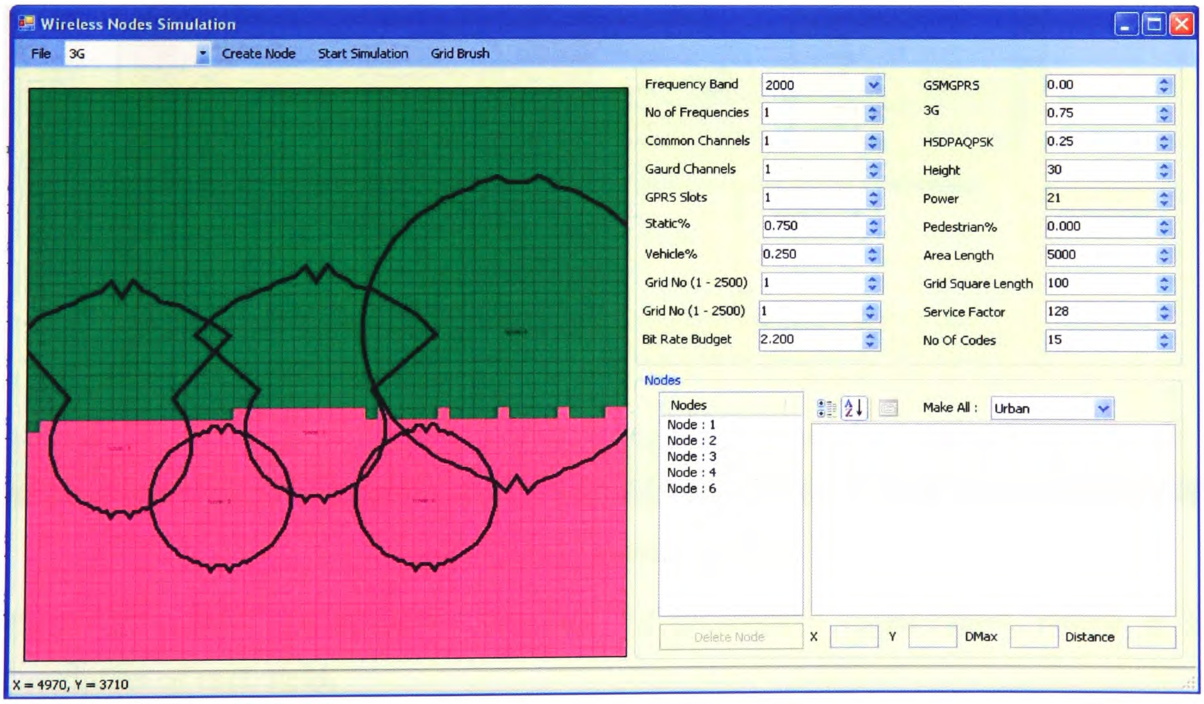

Figure 5.2 Node Coverage in Different Environments

As discussed in Section 5.2 the simulator was designed in modules, one of which is the terrain. The terrain was designed to reflect the coverage in a homogenous environment where height of the terrain was not a factor. The reason behind this decision is that, a preplanning site company Keima based in Cardiff has designed a site selection simulation tool that incorporates the geographic terrain. However, the tool does not include the services that the operators will offer to customers in line with 3GPP standards. As traffic type and intensity will affect the preplanning site location.

The main purpose of the simulator was to carry investigations with regards to CRRM and service continuity and we can interface Keima simulation tool to provide the digital terrain data to the designed simulator. This has also influenced our decision to use Visual Studio environment and C# as a programming language.

5.6 Parameters Initialization

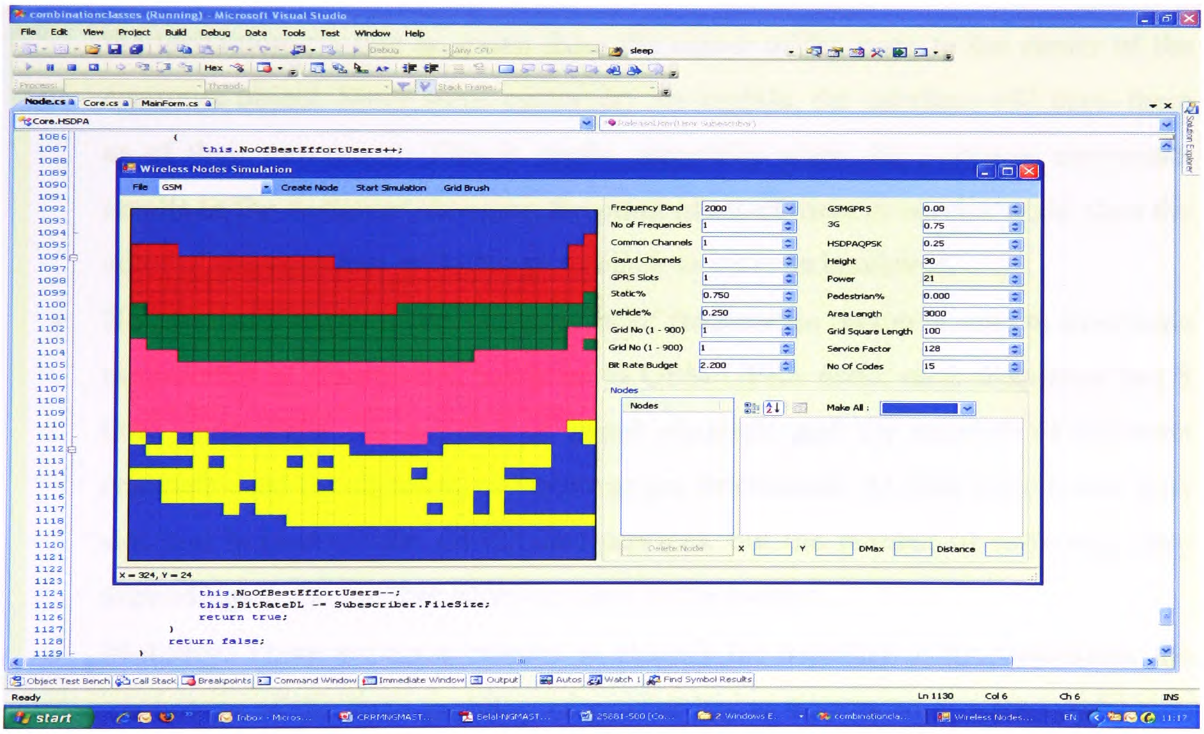

Figure 5.3 The Simulator Interface

Figure 5.3 shows the simulator interface for parameters initialization. Below is a description of each field.

- **The Environment**: The five different colors represent the environments that are included in the simulator as mentioned in Section 5.3.
- **Generate node**: Four different technologies are available to choose from, GSM/GPRS, 3G, HSDPA and LTE.
- **Frequency Band**: The frequency button has a drop down menu to choose what frequency band is investigated, the frequency bands range from 150 MHz to 2000 MHz, this is essential for path loss calculations as per the Equations in Section 5.3.
- **Transmit power**: The cells transmit power ranges between 10W to 40W.
- **Tower Height**: The tower height buttons are included in the interface, tower heights range from -300 m to 300 m.
- **Node Type**: Each node of different technology is created with different color to help identify the technology. The nodes are drawn with distinct colors based on the technology; a line is drawn from the center of the node to the center of the customer object. Since some customers are mobile, the interface will draw them as of their movement. This is useful especially when the customer movement results in the customer changing the point of attachment or service node, then the color of the customer will change to black to indicate handover.
- **Number of Frequencies**: The number of frequencies button is used to determine the number of frequencies assigned to GSM/GPRS node, each frequency has 8 time slots while the number of guard channels and the number of common channels used for signaling or handover are determined. At least there is one time slot that is reserved for GPRS data services, but the number of slots may vary depending on the number of voice users in the system.
- **Mobility**: There are three buttons to classify the mobility of the customers; the percentage of the three mobility categories has to sum up to 1, static, pedestrian and vehicle mobility buttons are included in the interface.
- **User equipment (UE)**: UE categories are also included in the interface; four different categories are GSM/GPRS, UMTS, HSDPAQPSK, and HSDPAQAM. The UEs are backward compatible, which means that the newer technology can access the old network but not vis versa, this is an essential property of the

requested service. The percentages of the UEs are determined and have to sum to unity.

- **Investigated Area**: The area of investigation has a button that is set to vary from hundreds of meters to 100 kilometers.
- **Grids:** The area is divided into square grids. Each grid can be in one environment.
- **Make All:** In this drop down menu, each grid is assigned a height, environment and allowable speed.
- **Spreading Factor**: As far as HSDPA technology is concerned, the number of SP ranges from 5 to 15 codes maximum.
- **Bit Rate Budget**: cell data rate throughput depends on the scenario and if MIMO is used or not, the values range from .5Mbps to 40Mbps.
- **Service Factor**: The service factor can be determined for 3G/3.5G to calculate the maximum coverage distance in the uplink as per Equation (3) above.
- **Start Simulation**: Basically the 'Start Simulation' button is linked to the simulation time and to inter arrival time which can be changed based on the simulation scenarios. The simulator is automated to run for several inter arrival times as many runs as needed to raise the confidence level in the results. The simulator can be paused at any time if needed to check out results by a click of a button, it can also be continued from the point that it has been paused at.
- **Customer Type**: Based on a percentage, the customer type can be either voice in which case a blue color will appear on the interface. If the customer type is data then a red color is to appear on the interface. However, this does not prevent data customers from accessing voice services, but a scenario might arise where voice customers will be given priority to voice services while data customers are given priority to data services and charges scheme will be different if the customer decides to violate user service level agreement (SLA). A line from the service node to the customer is drawn, such that by just looking at the graphical interface, a feeling for the simulator is developed. If the customer service has been terminated for any reason, either service time finish, out of area coverage drop or handover drop, then the graphical interface no longer shows the

customer. The graphical interface draws each user in the database that is still active or has just been admitted to the system for starting the service.

5.7 The Simulator Modules

The simulator has been designed as a modular and scalable simulation tool. Modularity enables the study of interaction between systems GERAN, UTRAN and LTE with minor changes in the core of the simulator. The idea is that if one module is not needed or a need arise to modify the module then there is no need to change other simulator modules. From programming point of view, the simulator has been designed in classes and methods. The class will not be called unless it is needed for that particular execution, an abstraction of the real mobile world into a simulation by abstracting nodes, wireless channels, users, services, radio propagation, event scheduling and events. The nodes have been abstracted into location, radio resources management, movements of users with respect to service node and coverage and capacity. This will help a great deal in planning an area user distribution, user traffic class and user mobility. As mentioned above this is a system level simulator with link level simulation tools used as input parameters. The following is a detailed description of each of the simulator classes and methods.

5.7.1 The User Class

The user main components are shown in the Figure 5.4 below.

- **Generate users:** The customer is generated with many attributes, each generated customer has customer ID that is incremented with each new user generated and used for identifications. The customer profile is built from the attributes such as, customer type as in voice or data which is randomly chosen by the simulator based on specified percentages. Customer type can be exploited by the operators to build a customer profile to determine the pattern of the customer behavior.

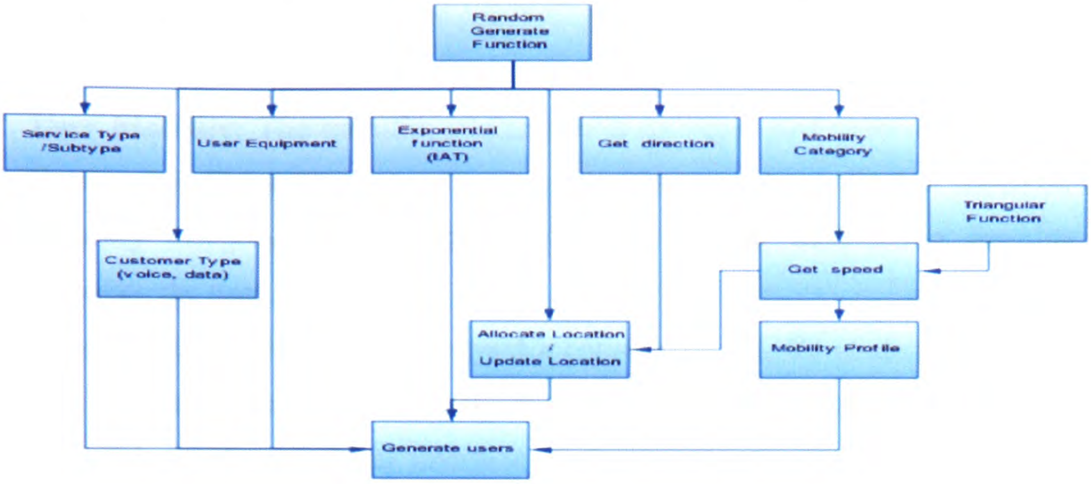

Figure 5.4 Generate Customer

- **User Equipment:** Each Customer is using one type of equipment, the percentages of the UE are an initial input parameter (refer to Figure 5.3).
- **Services:** The three major service types in accordance with [3] are Voice, Streaming and Best Effort services as shown in Table 5.1.

Table 5.1 Services considered

Service Type	Service Subtype
Voice	Voice Call
Streaming RT	VOIP
	Video Streaming
	Mobile TV
Best Effort	HTTP_WWW
	FTP

- **Service Sub Type:** The service sub type is generated based on percentages. Voice service has one subtype which is voice. The streaming service has three subtypes which are VOIP, Video Streaming (144 kbps) real time service and Mobile TV (384kbps) non- real time service. Best Effort has two subtypes which are Web browsing and FTP services. If the customer has a best effort service then what type of service bit rate is requesting, Web browsing is 80Kbps and FTP is 200Kbps.

- **User Mobility**: The mobility percentage is an input parameter as explained in Section 5.5. The average speed is an input parameter that is fetched based on the mobility category which is fed into the triangular distribution function to calculate the user speed.
- **User location**: The user is assigned a coordinates (X, Y) randomly on the grid and based on the mobility category the user starts changing location. The location is visible on the graphical interface as the user is changing location. The user location on the grid is updated by multiplying the speed of the user with the simulation time that has passed since the last user location update.
- **User Direction**: The user is assigned any direction in 360 degrees circle randomly and location starts changing based on location, direction and speed.
- **User Speed**: The user average speed is fed to the triangular function to determine the actual user speed, and the user location on the grid is calculated by multiplying the speed of the user with the simulation time that has passed since the last user location update.

5.7.2 Node Class

The node class share common attributes, when creating a node of any type the following attributes are included:

- **Frequency Band:** The operating frequency of the node is assigned by the drop down menu as described in section 5.5. The path loss calculations depend on the frequency band.
- **Node ID:** Each node created has a unique node ID. This will help in managing the simulator statistics and it is a requirement in the standards to supply this information to the UE in the cell coverage area. The core network supplies this information to the customer through monitoring of the common broadcast channel.
- **Node Power and height:** The height and power of each node is determined as shown in Section 5.5 above when initializing the simulator parameters. For

3G/3.5G nodes this can be a major factor in limiting and determining nodes capacity.

- **Node Location:** Location of the node in the investigated area is also specified. The X and Y coordinates are known and attached to the node. Node location is used for distance calculations between the user and the node.
- **Node Coverage area:** The grid average environment is used by the node to calculate coverage area especially when the node is coverage is spanning more than one environment. Maximum distance and frequency band are used as input parameters in Equations (1), (4–8). The coverage for UTRAN nodes is presented in Section 5.3. Each node keeps track of the number of users it is serving. Following are specific attributes of each technology type.

5.7.2.1 GSM/GPRS Node

The specific attributes for GSM/GPRS nodes include:

- **Radio Resources**: The number of frequencies the node is carrying. In TDMA/FDMA system as it is the case in 2G/2.5G systems, the frequency of 200 KHz is a frame. Each frame is divided into 8 time slots (as detailed in Chapter 2). The number of frequencies in each node determines the number of servers/time in the node by multiplying the number of frequencies by 8.
- **Channel Reservations:** Channel reservations like common channels, guard channels are reserved for signaling.
- **Data Channels:** Channels reserved for GPRS data services can be specified. The maximum number of data users in GSM/GPRS nodes is 32 which are consistent with the message header in GPRS. At least one time slot is reserved for GPRS.
- **Cell Coverage:** Depends on the maximum allowed path loss calculation, the area is drawn by the simulator.
- **User Equipment Accessibility:** Depending on the service sub type, all UEs can access this node type.

5.7.2.2 UMTS Node

In addition to the common attributes mentioned above in Section 5.6.2, each node has its own radio resources.

- **Radio Resource:** UMTS 3G nodes resources are determined based on Equations (3-7) in Chapter 2. The node keeps track of the number of users, their load and interference contribution to the serving node, to the neighboring nodes (as discussed in details in the Chapter 2). Uplink and downlink loads are also input parameters and can be modified depending on the investigated scenario.
- **Cell area Coverage:** The maximum area coverage is calculated as per Equation (1-8) above and is changing continuously by the number of users served by the cell. An input parameter of hundred points around the centre of the cell is taken to calculate the cell area coverage. The maximum coverage distance from the node centre depends on the environment by applying Equation 3 based on the following Table 5.2 of input parameters. The default parameters in Table 5.2 are based on [100] and can be changed in the simulator with ease. Thus, these parameters are used for an initial coverage area determination. While the simulator is running the maximum coverage area is changing dynamically.

Table 5.2 Initial system parameters for UMTS maximum coverage

Service Factor	2,4,8,……,128
Maximum UE transmission power	125mW
Other cell to own cell interference	55%
Thermal noise	-103dBm
Interference factor	0.2,0.4,0.5
Log normal Distribution	8dB
Frequency	2000MHz
Height of a mobile	1.5m
Height of base station	50m

- **User Equipment Accessibility:** UMTS and HSDPA UEs can access 3G nodes and as mentioned above the node can only serve streaming and voice services. The system considers 3G channels as virtual circuit switched channels and therefore it can be accessed by reservations. Thus, best effort services are not allowed to access to 3G nodes. This is not to say that the algorithm cannot be changed depending on the scenario. Antenna height and power are also attributes for each node and power required by each user can be calculated in the simulator if needed.
- **User Speed:** The user speed influences the radio resources allocation. The standard in [20] has identified the propagation channel conditions, user speed that ranges from 3Km/h to 250 Km/h and the required QoS.

5.7.2.3 HSDPA Node

A few more attributes are proprietary to 3G nodes offering HSDPA technology.

- **Spreading Codes**: If the node is to offer dedicated channels for voice as well as streaming and best effort services, then there is a need to allocate a number of spreading codes to be specified for streaming and best effort services. If the node is used totally for HSDPA then the power is divided among the 16 spreading codes (channels) of the node equally.
- **Power Allocation**: If the node is shared between 3G/HSDPA then the power is allocated dynamically HSDPA as discussed in Chapter 2.
- **Coding and Modulation**: HSDPA technology supports 16 Quadrature Amplitude Modulation (16 QAM) [25]. As a matter of fact the cell throughput can be doubled in theory. Two factors will influence this in the simulator, first, the measurements of the Signal to Interference plus Noise Ratio (SINR). If the instantaneous SINR value is 5 dB or better then the 16QAM is used, otherwise QPSK is used. However this has to be coupled with the second factor, which is the capability of the UEs.
- **User Equipment**: HSDPAQPSK and HSDPAQAM UEs are capable of accessing HSDPA node type. Some UE classes are not capable of taking

advantage of better SINR; in this case even in better propagation conditions the throughput remains constant.

- **Services:** The minimum required bit rate (MRBR) is depicted in Table 5.3.

Table 5.3 Minimum Required Bit Rate

Service Subtype	Minimum Required Bit Rate (Kbps)
VOIP	16
Video Streaming	144
Mobile TV	384
FTP	~200
WWW	~80

- **Cell Bit Rate Budget**: The node bit rate budget is an input parameter as described in Chapter 2 and as described in Table 2.4.

5.7.2.4 LTE Node

LTE is an all IP broadband technology based on OFDMA, below are a list of parameters specific to LTE:

- **Services:** Two more Non Real Time (NRT) services have been modeled, a minimum required bit rate of 512 Kbps and 1024 Kbps. The Mobile TV service is also NRT service and will be moved to be scheduled on LTE nodes.
- **User Equipments**: one more UE capable of accessing LTE has been modeled and will take advantage of the presence of LTE node.
- **Cell bit Rate Budget**: The bit rate budget has been modeled in accordance with Table 2.5.

5.7.3 The Core Simulator Process

The core of the simulator links all the above modules together. The core itself has several processes. The initial input from the environment and the topology are deployed to the core. The parameters are initialized as discussed above and the core is ready to be launched.

The simulation time and Inter Arrival Time (IAT) is pre-determined in the simulation process, the simulator reads the IAT while the output is saved in a text file for later analysis. This is repeated for all values in IAT file automatically, all of the text files that are created for each run are saved in a folder stamped with a time and a date of execution. To increase the confidence level in the simulation results, the runs can be repeated as many times just by specifying the number of runs.

The simulation process starts by creating the first arrival event and store it in the scheduler.

In the core simulator, several events are executed; two types of events are available. First, events that are related to specific time and have to be executed according to their designated time; the arrival and departure events are two examples. Second, events that are not tagged with time; handover and drop events are two examples. However the latter events are executed every time that the simulator is fetching an event from the scheduler an example are handover process and drop process.

Following is a description of each block.

- **The Arrival process:** This process has several functionalities; when the event is arrival this process is executed. It determines the customer profile (i.e. voice or data, service requested, service subtype, mobility and UE capabilities). It generates the user with its different attributes as described in Section 5.6.1. The call admission control algorithm is called by the arrival process as depicted in Section 5.6.3.1. Based on service subtype, an expected departure time is generated and a departure event is stored in the scheduler using the random access generator based on average service time. It also creates the next arrival event and feed that back to the scheduler based on the inter arrival time. The scheduler sorts the events based on the events time associated with each event.
- **The Departure Process:** The departure process is executed if the next event in the scheduler is departure. The counters are updated and the statistics are collected.
- **Handover Process**: If the handover is required then executes the handover process. The simulator fetches the next event from the scheduler; if the event is an arrival event then the handover process is called and executed. This process has three stages and explained in details in Section 5.6.3.5.

- **Drop Process:** The drop process is executed between the time events. The drop process drops users for many reasons, as example, out of cell coverage area, handover failure and loss of connection due to propagation path loss exceeding the threshold value.

The Scheduler: Acts as time manager; it stores the arrival and departure events. If the first event in the scheduler is arrival, then the simulator executes the arrival process as explained in Section 5.6.3.1. However, if the next event is departure then the simulator executes the departure process where statistics are collected and the user is released.

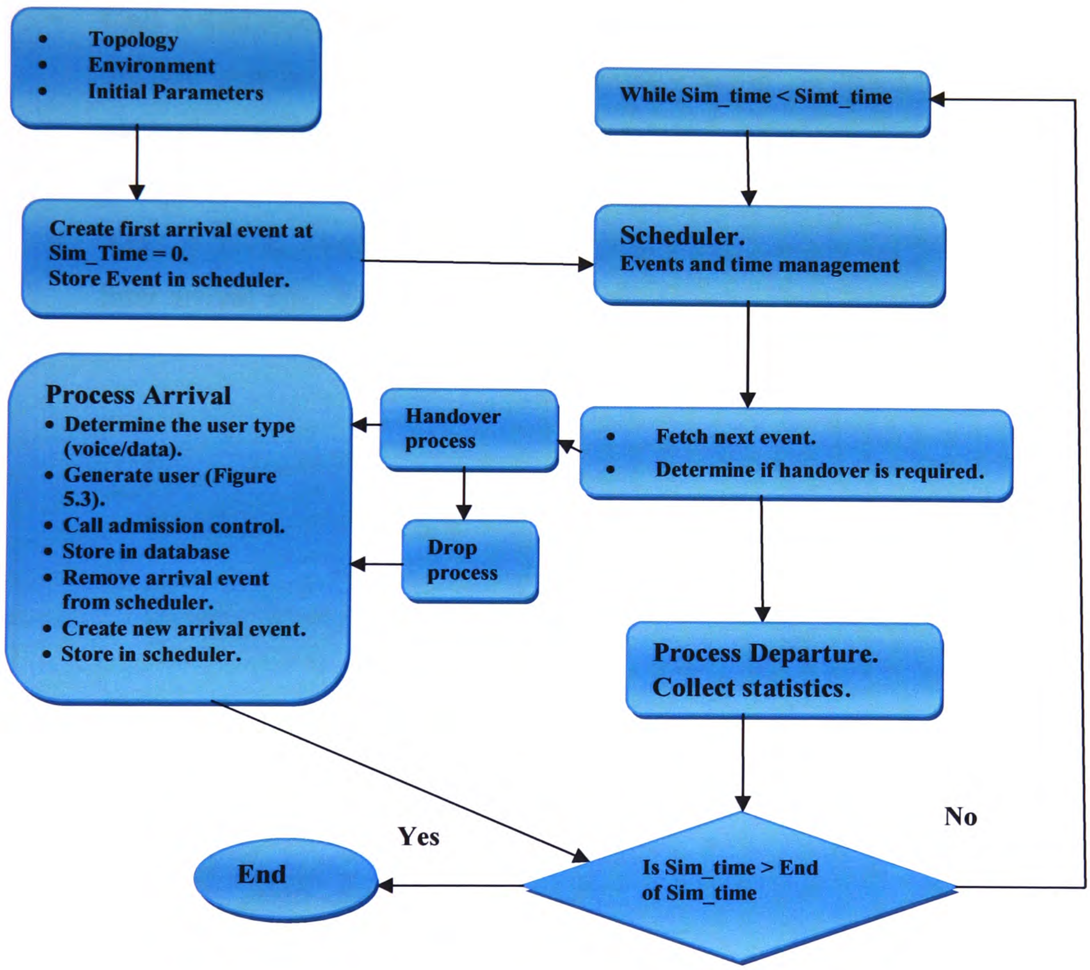

Figure 5.5 The Simulator Core Process

5.7.3.1 The Arrival Process

The arrival process has several tasks as depicted in the following Figure 5.6.

- **Inter Arrival Time (IAT):** The customer arrival rate is modeled using Poisson exponential distribution with one inter arrival time for all user types. However, based on predefined percentages the user type is generated. This could be used as part of user profile in future studies. If the user is assigned as a data user then the behavior and services requested are different than if the user is assigned as only a voice user.
- **User Generator:** The user is created as illustrated in Figure 5.4, with all attributes associated with it. The user profile is formulated based on the percentages for each attributes as illustrates in Figure 5.3.
- **Call Admission Control:** This is the decision making entity to admit or reject the new generated users. Based on the user profile and the available radio access network resources the decision is formulated. More details on this entity when we discuss the CAC algorithm in Section 5.6.3.2.
- **Next Arrival Event Generation:** Next arrival event is generated based on the inter arrival rate. An arrival event is created and stored in the scheduler entity.

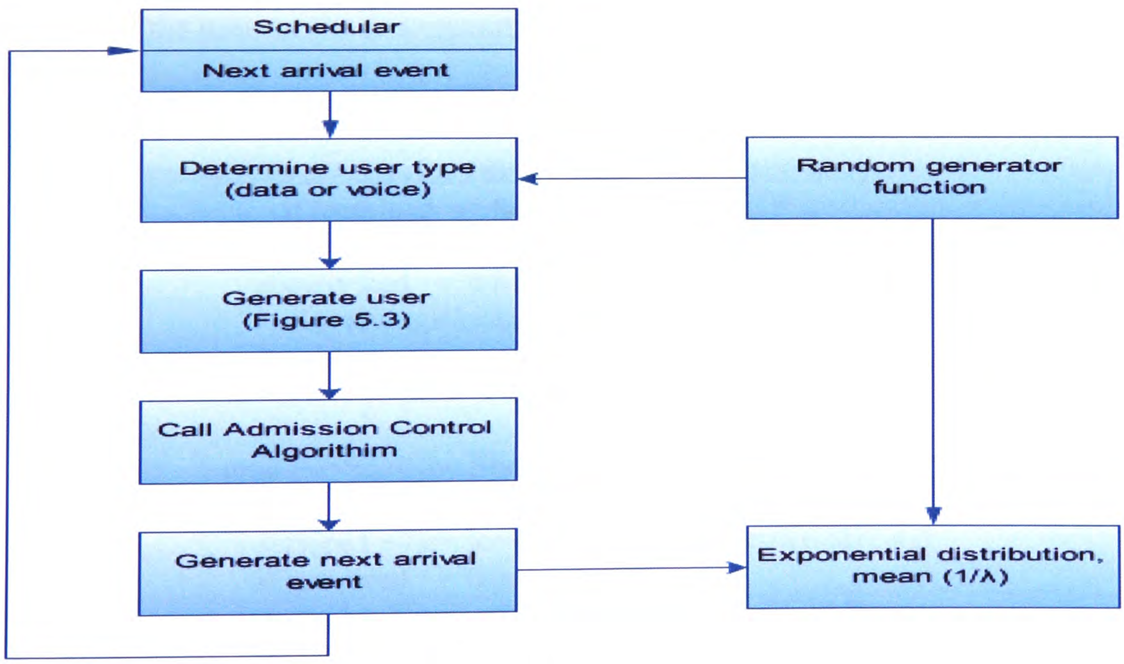

Figure 5.6 The Simulator Arrival Process

5.7.3.2 Call Admission Control Entity

Call Admission Control (CAC) entity has several blocks that help in executing the Algorithm. Figure 5.7 below depicts each of these components.

- **Try Access:** Based on the customer profile that has been created when the arrival process is called by the simulator and based on service requested by the customer. The initial cell selection node type is determined by executing the "Try Access" method as follows:

```
void TryAccess(User Subescriber)
        {
///based on user service type the user is directed to the proper RAT.
            bool serviced = false;
            switch (Subescriber.ServiceType)
            {
/// If the service is voice try 3G nodes first if rejected then try GSM.
                case ServiceType.VoiceCall:

                    serviced = Services(Subescriber, typeof(ThreeG));
                    if (serviced == false)
                        serviced Services(Subescriber, typeof(GSM));
                    if (serviced == false)
                        VoiceBlocked++;
                    break;
///If the user service is streaming then try HSDPA first, if rejected then Try 3G

                case ServiceType.Streaming:

                serviced = Services(Subescriber, typeof(HSDPA));
                        if (serviced == false)
                        serviced Services(Subescriber, typeof(ThreeG));
                    if (serviced == false)
                        StreamingBlocked++;
                    break;
///If the user service request is best effort then try HSDPA first, if rejected try
GPRS
                case ServiceType.BestEffort:

                    serviced = Services(Subescriber, typeof(HSDPA));
                    if (serviced == false)
                   serviced = Services(Subescriber, typeof(GSM));
                    if (serviced == false)
                        BestEffortBlocked++;
                break;
```

```
/// If user service request is NRT service then LTE if rejected then try HSDPA
                    case ServiceType.NRTLTE:
                    serviced = Services(Subescriber, typeof(LTE));
                        if (serviced == false)
                            serviced = Services(Subescriber, typeof(HSDPA));
                        if (serviced == false)
                            NRTBlocked++;
                     break;
}
```

- **Path Loss and Coverage Area:** After determining the radio access technology that is better suited to service the user request as described above. The CAC algorithm is executing the path loss method as follows:

```
public void CallAdmissionControl(TypeNodeType)
    {
      Blocked = false;
      ServiceNodes.Clear();

SortedDictionary<double, Node> NodesArray = new SortedDictionary<double,Node>();
               foreach (Node node in Nodes)
             {
              for (int i = 0; i < Nodes.Count; i++)
              {
               if (node.GetType() == NodeType)
              {
               double powerloss;

//// if the potential serving node type is GSM then the algorithm will execute the
following program by first check if the User is within the coverage of the node and
calculate the pathloss for the user. The propagation loss is calculated based on the
environment and Equations 4-8 as mentioned in Section 5.3///
              if (node.GetType() == typeof(GSM))
                        {
      if (node.CustomerDistance(this.location.X,this.location.Y) > distance)
                         powerloss = -1;
      else powerloss = node.PathLossPower(this.Location.X, this.Location.Y);

/// However if the node is either 3G or HSDPA then the algorithm calculates the path
loss while considering the maximumdistance calculations. Coverage is related to
capacity as explained in Section 2.3.4.2 and [98-99] for UMTS nodes. ///

else if (node.GetType() == typeof(ThreeG) || node.GetType() == typeof(HSDPA))
                    {
                       powerloss = node.GetPathLossConsideringDmax(this);
                    }
          else
               powerloss = node.PathLossPower(this.location.X, this.location.Y);
                  if (powerloss != -1)
```

/// Each node satisfies the above conditions are added to the potential serving list in an array based on the calculated power loss.///

```
                    NodesArray.Add(powerloss, node);
                for (int i = 0; i < NodesArray.Count && i < 5; i++)
                {
                    ServiceNodes.Add(NodesArray.Values.ElementAt(i));
                }
                if (NodesArray.Count == 0)
                {
                    Blocked = true;
                }
    }
```

- **Voice Calls**: If the UE category is 3GUMTS or better then try all 3G nodes in the list. If there is enough capacity in one of the 3G nodes then admit the user and update the node counters. Otherwise, try GSM nodes available to serve, if there is available time slots then admit the user and update counters else the customer is blocked. If the UE category is GSM/GPRS then try GSM nodes only, if there is available time slot then admit. Otherwise, the service is blocked. Voice calls are admitted first on 3G nodes based on UE category, the rationale behind this approach is that 3G nodes have better spectral efficiency than GSM nodes.
- **Streaming Services**: If the UE is HSDPA capable, then the CAC check if HSDPA node/nodes available, if the node/nodes can serve the customer, admit the user to one HSDPA node and update node counters. Otherwise, check if 3G node/nodes are available and the node can serve the customer (capacity available), then admit the user and update counters, else block the customer.
- **Best Effort Services**: If the UE is HSDPA capable then try admission to HSDPA node types. Otherwise, if there is capacity in GPRS nodes then calculate a new service time and admit the user, else block the session.

CAC Algorithm can be enhanced and tailored to be consistent with the deployment scenario; it can be changed and modified easily to fit the operator envisioned traffic types and area coverage.

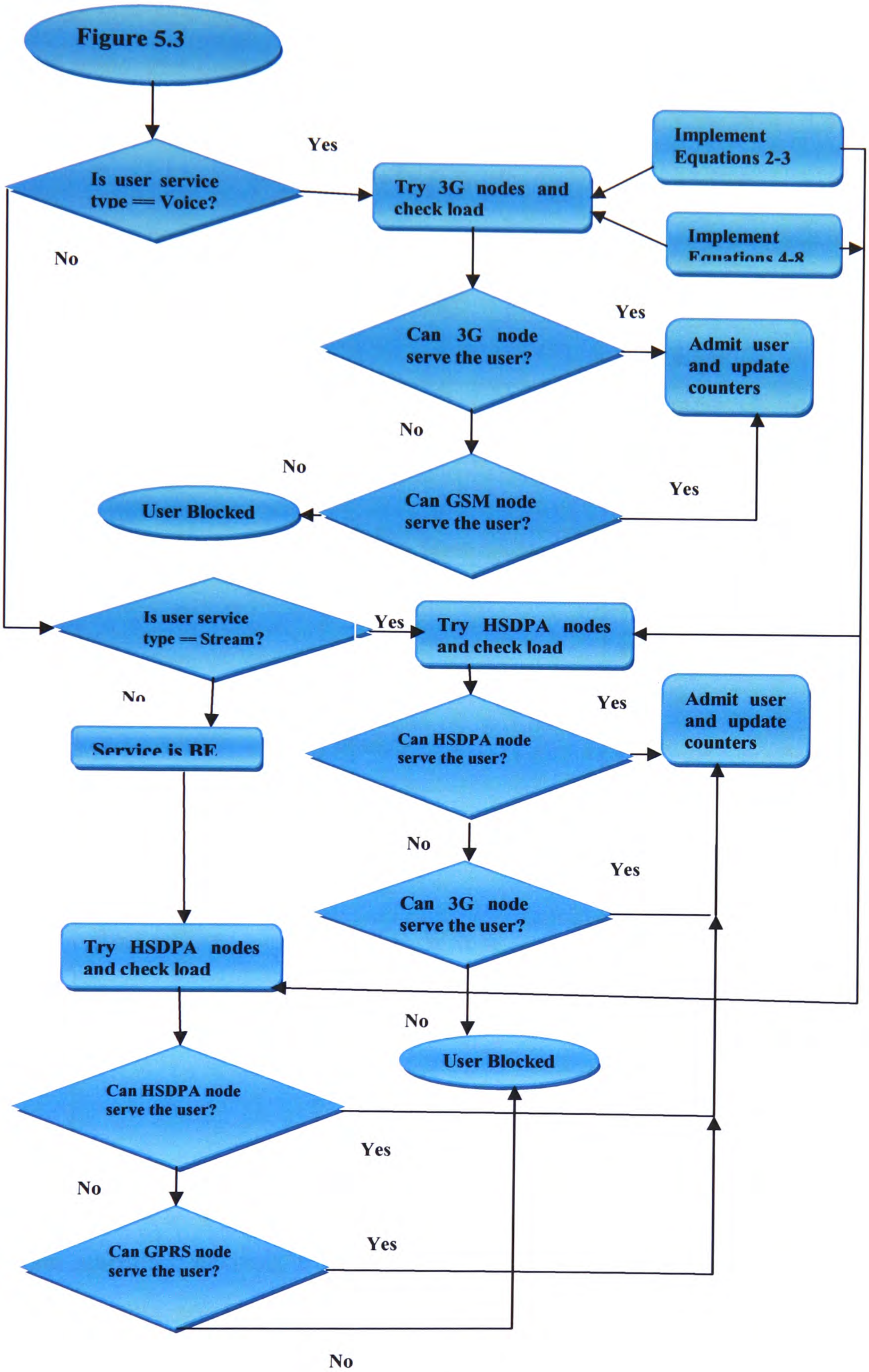

Figure 5.7 CAC Algorithm

If the service node has been determined based on node capacity, coverage, the service requested and service capabilities of the user UE as specified above then the user and service node are coupled together; the average service time is randomly generated according to exponential probability distribution as depicted in Table 5.4.

Table 5.4 Average service time for each service

Service Type	Service Subtype	Parameters
Voice	Voice Call	Average call duration 120 seconds
Interactive	VOIP	Average call duration 120s, and minimum required bit rate 16kbps
	Video Streaming	Average video session duration is 200s, and minimum required bit rate is 144kbps
	Mobile TV.	Average session duration is 300s, and the minimum required bit rate is 384kbps
Best Effort	HTTP_WWW	WWW 20 packets – Average _ Packet size : 500Bytes
	FTP	WWW 50 packets – Average _ Packet size : 500Bytes

The user starts the service, a departure event is created, stored in the scheduler and the scheduler events are sorted. GSM/GPRS Nodes can serve two types of users, circuit switched voice users and Best Effort packet switched users. The voice call average service time is 120 seconds; the call is retained in the system until it is terminated. For best effort services, the user packet size is generated based on the geometry distribution.

The user data rate depends on the number of slots available for GPRS and the number of best effort users requesting the service. Each time slot is assumed to output a 20Kbps, the packet size is divided by the user bit rate and the service time is calculated.

5.7.3.3 The Departure Process

Once the customer service time is finished based on random session time according to parameters presented in Table 5.4, the departure process is called. The departure process based on the customer service node and service rendered. The statistics are collected and the customer is released from the simulator. The departure event is deleted from the scheduler and the user is released from the database. The service node is updated and the load is adjusted.

5.7.3.4 Drop Process

The drop process is invoked by other events. If the user is outside the cell coverage area and the handover is not successful for any reason then the drop process is called, the user also can be dropped if the MAPL reading for the service is above certain service threshold. Once the user is dropped, it is deleted from the database records. The departure event in the scheduler associated with the user is also deleted. Statistics are collected and the service node regains the radio resources accordingly.

5.7.3.5 The Handover Process

The handover process is a key entity in the simulator core. 3GPP standard [62] outlines the service requirements for handover. If handover is due to UE mobility, then there should be no measurable impact on the quality of service. The standard does not imply that all handover should achieve this ideal requirement, but this should be achieved based on the UE speed not exceeding the standard requirements and that the UE stays within a single UTRAN coverage area. In case of the handover to a cell of different radio conditions and the quality of service for streaming cannot be maintained then the quality of service might be modified to a lower quality to maintain the continuity of the session. Handover can also be due to a different radio access technology (RAT) better suited to service the session. This makes it possible for the core network to recommend to the access network to handover the UE to another RAT. Considerations must be given to cell capacity especially when multimedia sessions are involved. As the target cell might not be able to support all the bearers, handover to another cell should not be precluded.

The UE should have access rights to any cell that the handover needs to take place. If the UE does not have access rights based on SLA then the handover should be prevented.

In general, it is always preferable that the UE is not involved in mobility management in order to make the UE battery last longer. The core network mobility management has the capability and information to fulfil the requirements of the mobility without heavy involvement from the UE.

The UE should have access rights to any cell that the handover needs to take place. If the UE does not have access rights based on SLA then the handover should be prevented.

In general, it is always preferable that the UE is not involved in mobility management in order to make the UE battery last longer [16]. The core network mobility management has the capability and information to fulfil the requirements of the mobility without heavy involvement from the UE.

Handover can be classified into two operations, first, inter-system operation where the handover will be to a different radio access technology as the case when the UE is moving from a cell in GERAN to a cell in UTRAN or UTRAN to GERAN. In this case the handover is hard handover and the UE is involved in the operation, this type of handover is referred to as vertical handover. Second, intra-system handover, where the handover is to the same type radio access technology and called horizontal. In this case the UE is moving from GERAN cell to another GERAN cell or when moving from UTRAN cell to another UTRAN cell [101]. The operators have an extensive experience with this type of handover because mobile networks used to be one type of technology until recent days.

The handover algorithms and the events that triggers the handover process are not standardized [5, 65, 102] by 3GPP, they are operator specific.

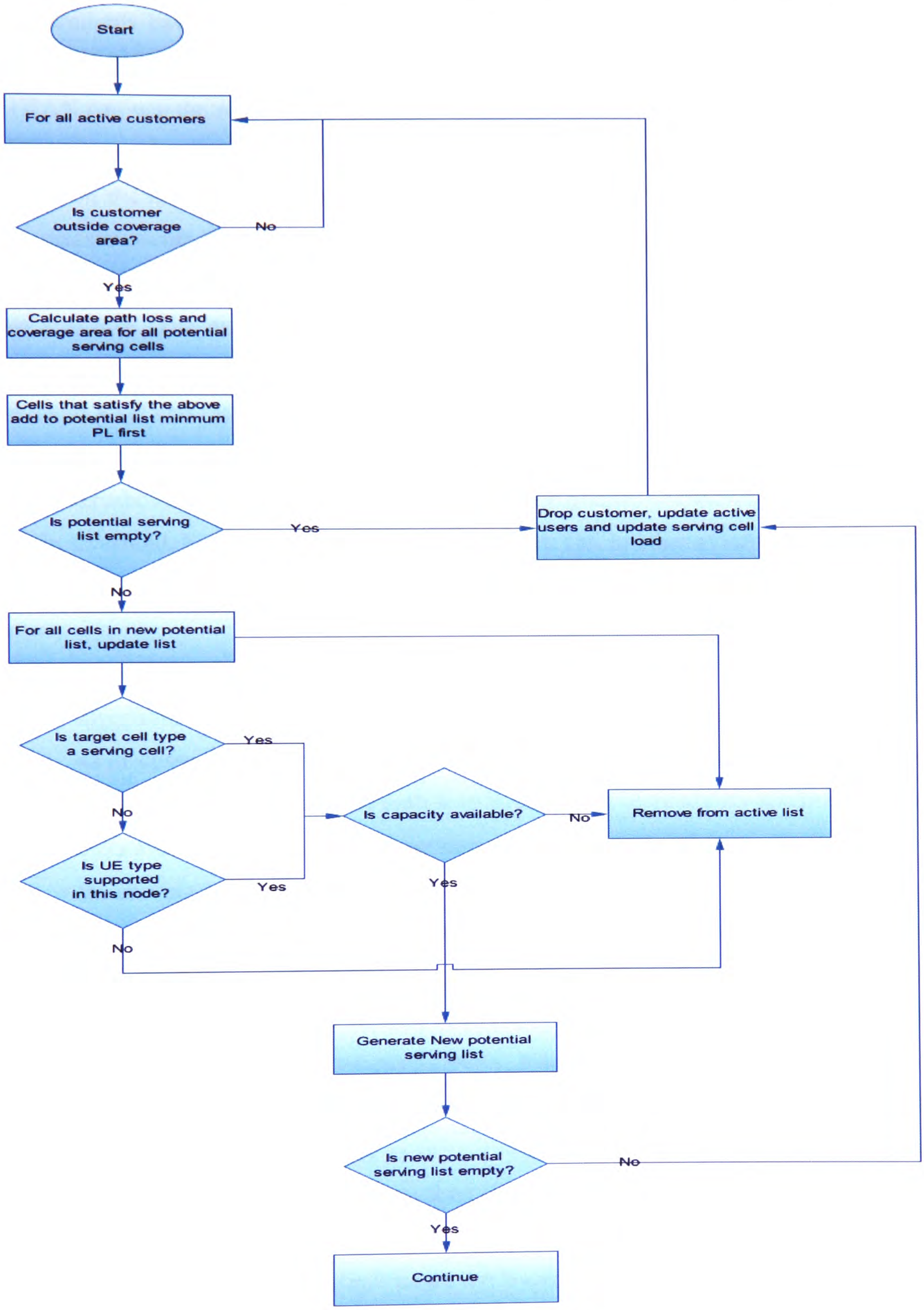

Figure 5.8 Handover Algorithm

The standards have defined measurement reporting criteria in the measurement control message. UTRAN notifies the UE which events should trigger the reporting events. The UTRAN can use the reporting events to evaluate if a handover is needed [65, 102]. The UTRAN can choose the reporting events to be triggered by the UE. Such events are carried on the monitored primary common pilot channel (CPICH) of the cell specified. The measurements include path loss measurements criterion which was the dominant criterion because horizontal handover was the only required handover as only one RAT was available. The events are numbered 3A, 3B, 3C and 3D [65]. Event 3A used to estimate the quality of the currently used system if the quality falls below a certain threshold level and the estimated quality of the other system is above a certain threshold level. Event 3B is used to report if the estimated quality of other system is below a certain threshold. Event 3C is used to estimate if the quality of other system is above a certain threshold. Event 3D is used to change the best cell in other system [65].

The handover Algorithm proposed in the simulator has three main stages as illustrated below:

1) **The First Stage (measurement Process)**: As depicted in Figure 5.8, the algorithm check if the active customer is outside the serving cell coverage area as discussed in Section 5.3. If this is true, then the Path loss is calculated for all cells that satisfy the service path loss threshold. If there are no potential serving cells available then the call or session will be dropped and the radio resources of the serving cell are updated.
2) **The Second Stage (Node Elimination Process)**: The potential serving cells list is updated through the second stage, this stage deals with capacity. If the cell is of the same type of the serving cell, check the capacity of the cell. If there is enough capacity for the required service, then keep the cell in the potential target list, if not then remove the cell and update the potential serving target list. If the cell is not of the same type as the serving cell then check if the target cell can support the customer UE, if yes then check the cell's capacity. If both criterions are satisfied then keep in the list. Otherwise, remove from the potential target list and update the list. If the cell target list is empty then the call/session is dropped.

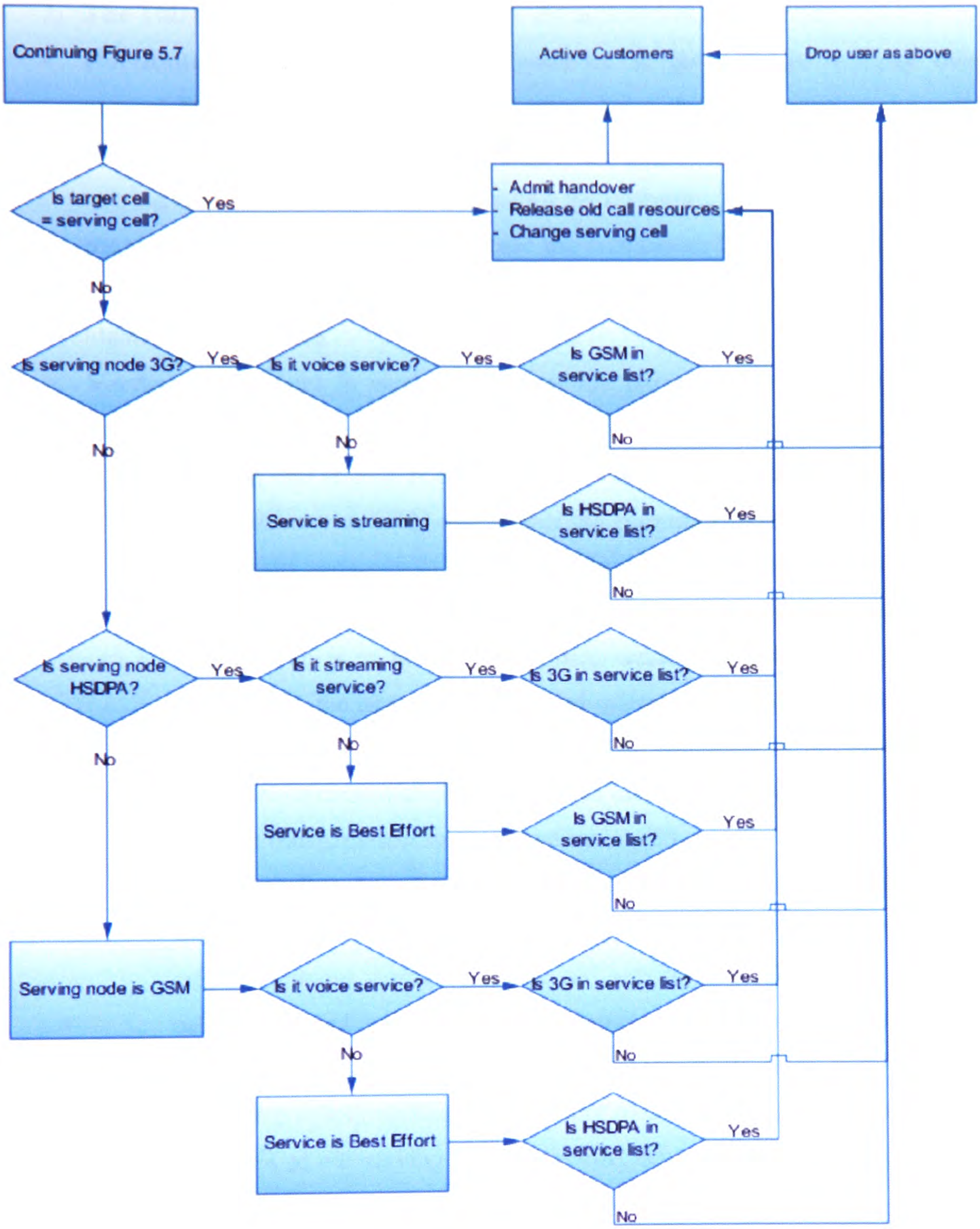

Figure 5.9 Handover Algorithm continue

The customer is removed from active users and the Radio Resources of the serving cell are updated.

3) **The Third Stage (Execution Process)**: As depicted in Figure 5.9:

- **Horizontal Handover**: Horizontal handover is given higher priority over vertical handover. The algorithm will try to keep the call/session in the same system. The algorithm starts by checking if the target cell is of equal type to the serving cell. If this is the case then admit the customer, update the resources of the new serving cell and finally release the resources of the old serving cell. If the target cell is not of the same type as the serving cell then the following algorithm is followed.
- **Vertical Handover:** In vertical handover the process depends on the serving node and on the service sub type as follows:
  - **Serving cell is 3G:** If the service is voice, if GSM is in the target list then handover to GSM and update the resources in the new serving cell and release resources of the old serving cell. Otherwise, drop the call and update resources in the serving cell. If the service is streaming, if HSDPA cell is in the target list then handover to HSDPA, update the resources in the new serving cell and release resources of the old serving cell. Otherwise, drop the call and update resources in the serving cell. This is consistent with the CAC initial cell selection process as only two types of traffic classes are serviced by UMTS.
  - **Serving cell is HSDPA:** If the service is streaming, if 3G nodes are in the target list, handover to 3G, update the resources in the new serving cell and release resources of the old serving cell. Otherwise, drop the call/session and update the radio resources in the serving cell. Otherwise, the service is best effort, if GSM is in the target list, if yes then handover to GSM (GPRS), update the resources of the new target cell and release resources of old serving cell. Otherwise, drop the session and update the serving cell.
  - **Serving node is GSM**: If the service is voice, is 3G in the target list then handover to 3G; update resources of the new serving node and release resources of the old serving node; otherwise drop the call and update resources of the serving cell. If the service is Best Effort, is HSDPA cell

in the target list, if yes then handover to HSDPA, update the new serving cell and release resources of old serving cell. Otherwise, drop and update the resources in the serving cell.

There are two main properties in the algorithm, first, it covers both horizontal and vertical handover, and second, priority is given to horizontal handover or intra system handover as required by the standard [8, 62].

5.7.3.6 Mobility

Location is updated dynamically at each event of the simulator. When the user is created speed, direction and initial location point in the investigated area is associated with the user at random. As the active customer get serviced, its location is updated based on the distance multiplied by the difference between the simulation time that has passed and the time when the customer service has started. Customer location is updated dynamically. The speed of the user is an input parameter to the simulator, with percentages are determined in line with the scenario of the study. Users are uniformly distributed in the investigated area.

5.7.3.7 Traffic Class Modeling

Four UMTS traffic classes have been defined in the standards [36, 61]. Each traffic class has certain QoS characteristics. Below is a description of each of the traffic class and QoS requirements.

- **Conversational traffic:** This class is very delay sensitive class. The traffic is described to be of Constant Bit Rate (CBR) in nature as in circuit switched networks. Some internet applications also have the same characteristics like VOIP and video conferencing calls. This class usually is full duplex and if the delay is more than 150 ms the quality of the service will drop below the quality expected by the user. For video conferencing synchronization between video and audio is essential. The lip movement of the speaker should be synchronized with audio delivery.
- **Streaming Traffic:** Streaming can be audio or video real time streams. The time relations between the transmission entities should be preserved. In other

words, the packets should arrive in order. The startup delay is 10 seconds and 2 seconds for transport delay variations. Usually it is one way service [4]. This service usually is a half duplex or unidirectional service.

- **Interactive traffic:** This traffic is when end user is online sending and receiving data. This interaction can be with another server or machine as in telemetry services. This class can tolerate up to 4 seconds in delay but with zero loss of information. This class usually is a full duplex traffic but with some delay tolerant applications.
- **Background traffic:** This traffic is sent between computers. There is no time constraint on the delivery of the data. The content of the data should be transferred transparently with minimum error rate.

The type of traffic determines the QoS requirements. It is not practical to start listing applications and map them to the traffic classes above [4, 61]. Applications using the interactive or real time conversational communication schemes can also be described according to their possibilities for adapting to different environmental conditions. There are three main categories of applications. First, rigid applications like voice service using full rate speech is referred to as voice call. Second, other applications that require network supported negotiation to adapt to the network conditions are referred to as streaming services such as VoIP, Streaming video and Mobile TV. Third, elastic applications that adapt totally to the network environment and do not require service negotiations, referred to as Best Effort applications.

In the simulator, several services have been defined as illustrated in Table 5.1.

The voice call and the interactive service are of high priority and the QoS is none negotiable. The Best effort class is considered elastic application and does not require service negotiation. However, for the interactive services this might change and the approach might change to network negotiating the QoS in case of congestion or mobility. The voice channels have been modeled based on the node type as explained in Section 2.2 and Section 2.3.4.3.

5.8 Simulation Environment

Visual C# has been used for this simulator. C# has a rich object oriented capabilities and functionality which made it easier to model the different entities as separate objects and the interaction between the objects can be modeled with straight forward approach. The interface that was designed for the simulator made it very easy to interact and design the scenarios in an efficient manner. The DOT NET (.NET) frame work offered by Microsoft includes a huge library classes to tap on. The base class library has helped a lot in modeling the node class [107]. The simulator results have been validated by using confidence intervals.

5.9 Chapter Summary

Simulation and modeling tools can be of a great value to the operators; they can provide extensive statistical data about proposed algorithms, system behavior and network devices prior to actual deployment or field trials. With the deployment of several radio access technologies, CAC is becoming increasingly complicated. Each RAT has its RRM entity to manage the radio resources; the operator can tap into all resources but needs to devise efficient sophisticated algorithms that can evaluate several parameters at the same time. The exponential flux of new portable equipments under the disposal of the end users and the expected connectivity anywhere at any time complicates the operators' objective of providing and maintaining connectivity to end users. Adding to that is the new set of services the operator has to offer as in conversational, RT streaming, NRT streaming and interactive services further complicates the network system. All of the above research will be better enhanced by the availability of a suitable simulation tool that can incorporate all of the above variables in one engine.

This Chapter dealt with the simulator design. The entities and classes are designed separately and the functionality of each class is called only when needed. This design is modular and scalable; Modular in the essence that each entity can be modified, enhanced or not called at all and the simulator will function properly; Scalable in the essence that any new technology can be added or enhance the existing ones with no measurable effects on the total design or programming efforts. The simulator is designed as a Discrete Event system level Simulator (DES) which means that time correlations between events are considered when an event needs to execute. This has saved lots of storage and run time. An event scheduler was designed to help in tracking the time events and to execute them in the order that they are supposed to be executed.

The design of the geographical area and the segmentation of that area into grid is visually presented. The traffic classes are designed based on the recommendations as in [92, 103], with the QoS requirements as specified in [3, 21]. Finally the main algorithms are also defined more specifically as in the case of call admission control and in case of providing service continuity in the case of the handover algorithm.

In Chapter 6 next, several investigations are carried out to provide proof of concept of the simulation system and of the algorithms that are presented in this Chapter and in the previous one.

Chapter 6

Investigated scenarios based on simulator

6.1 Introduction

Based on the algorithms that are presented in Chapter 4 and Chapter 5; namely the CAC algorithm and the handover algorithm, a number of scenarios based on the DES system that was presented in Chapter 5 were investigated and evaluated.
Below is a description of the scenarios studied and the basic system assumptions that was considered when before running the simulation.

6.2 Basic System Assumptions

The algorithm as described in Chapter 4 and detailed in Chapter 5 was considered in the following studies.

Table 6.1 System Parameters used in simulation

Parameters	GSM/GPRS	3G	HSDPA
Capacity	8 Frequencies/ 8time slots, 1 slot reserved for Data	Deduced from Equation 5, 6, 7 and 8 of Chapter 2.	2.2 Mbps
Load Factor		Uplink 0.75/ Downlink 0.8	
Noise rise		6 d B	
Chip rate		3.84Mchip/s	3.84Mchip/s
Spreading factor		Variable	16 Codes, 15 used for data
Orthogonal factor		0.5	
Chip rate		3.84Mchip/s	
Orthogonal factor		0.5	
Carrier Frequency	900 MHz	2000MHz	2000MHz
Propagation Environment	Free Space		
Traffic	Voice/ BE	Voice/ Streaming	Streaming/ BE

Table 6.1 depicts the system parameters that was used in the investigation, below is a brief description on each parameter:

- **Capacity**: In GSM/GPRS nodes the capacity has been applied to FDMA/TDMA with ease as the number of time slots available in each node. When considering packet switched services through GPRS it has been considered that each time slot can have 20Kbps throughput. In WCDMA this same principle cannot be applied because the resources are in two domains, the circuit switched domain and the packet switched domain. One of the assumptions in our simulator is to use Equations 5, 6, 7 and 8 from Chapter 2 to quantify the number of simultaneous users based on service, propagation models and interference. In HSDPA, the total budget bit rate was used, each user adding to the total bit rate budget.
- **Load Factor**: The load factor is considered for UMTS nodes in the uplink to be 75% and the downlink load is 80% of the total node capacity.
- **Propagation environment**: For the propagation environment, the following parameters have been considered, the path loss is defined in the following Equation as in [98]:

$$Lp(d) = 10 * log(PS / PR) \qquad (1)$$

d is the distance in kilometres, Ps and Pr are the transmitted and received power. The propagation loss in free space propagation environment in d B [98] is given by:

$$Lp(d) = 32.44 + 20\, log(f) + 20\, log(d) \qquad (2)$$

f is the frequency in Mega Hertz. From Equation (2) the maximum distance of cell coverage can be deduced as in [100].

- **Traffic Classes**: Three traffic classes are used in the scenario, Voice, Streaming and best effort as depicted in Table 6.2.

6.3 Measure of Investigation

The arrival intensity was used as an input parameter for the simulation investigations. Several inter arrival time was stored in a file, with each simulation run the run is saved for further analysis.

Traffic distribution was also used as an input parameter; the percentage of the traffic is changed between the voice services, streaming and best effort services. The percentages have also been changed among the same traffic service as in streaming services. Streaming services have three subtypes, VoIP, Video streaming and MTV streaming and can be investigated with different percentages.

The performance measures that have been recorded in the investigations are as follows:

- **The blocking probability**: The blocking probability is defined as

$$Bi = \frac{(Number\ of\ Blocked)i}{Total\ Arrival} \qquad (1)$$

  For each service type the blocking probability is calculated. The operators usually like to consider a blocking probability of 2% or less in the network [8]. For data services the blocking probability is controlled by the MRBR for the required streaming service and also defined as in Equation 1 above.

- **Trunking gain**: Less blocking for real time services, higher throughput and lower delay for non real-time services is defined as the trunking gain. To provide optimal resource management and to optimise network performance in multi-radio environment [3].

6.4 Envisioned Scenarios

The scenarios that can be investigated through the developed simulation tool as discussed in Chapter 5 are enormous. However, in the following a small number of scenarios have been used to illustrate the simulation tool capabilities.

In Table 6.2 below two scenarios are presented with different traffic distribution and percentages.

The simulator has been set up to run with traffic intensity that span from 0.1 to 6, the average holding time is exponentially distributed and depends on the service requested and as explained in Section 5.6.3.2. The simulator was run for the traffic distribution scenarios' in Table 6.2. The simulator was set for eight GSM/GPRS frequencies, the 3G uplink and down link load is at .75 and .8 respectively and HSDPA node has a 2.2Mbps bit rate budget.

Table 6.2 Traffic and UE distribution for Scenario 1 &2

Scenario 1		**Percentages %**		**UE Category**
Traffic Distribution	**Voice 75%**	75%		75% 3G
	Streaming 25%	VOIP	12.5%	25% HSDPAQPSK
		Video Stream	7.5%	
		Mobile TV	5%	
Scenario 2				
Traffic Distribution	**Voice 85%**	85%		85% 3G
	Streaming 15%	VOIP	7.5%	15% HSDPAQPSK
		Video Stream	4.5%	
		Mobile TV	3%	
Scenario 3				
Traffic Distribution	**Voice 65%**	65%		65% 3G
	Streaming 35%			
		VOIP	17.5%	35% HSDPAQPSK
		Video Stream	10.5%	
		Mobile TV	7%	

Figure 6.1 below shows the blocking probability of the three voice traffic distributions, it is noted that for 85% voice traffic, the 2% blocking probability of the system is reached when the traffic intensity is approximately 0.9 and for 75% voice traffic, the system reaches the 2% blocking probability when the traffic intensity reaches .9375. However, when the voice traffic percentage is at 65% the 2% blocking probability was reached when the traffic intensity is at 0.858. This is an indication that the system efficiency is best when the voice traffic percentage is at 75% of the total traffic.

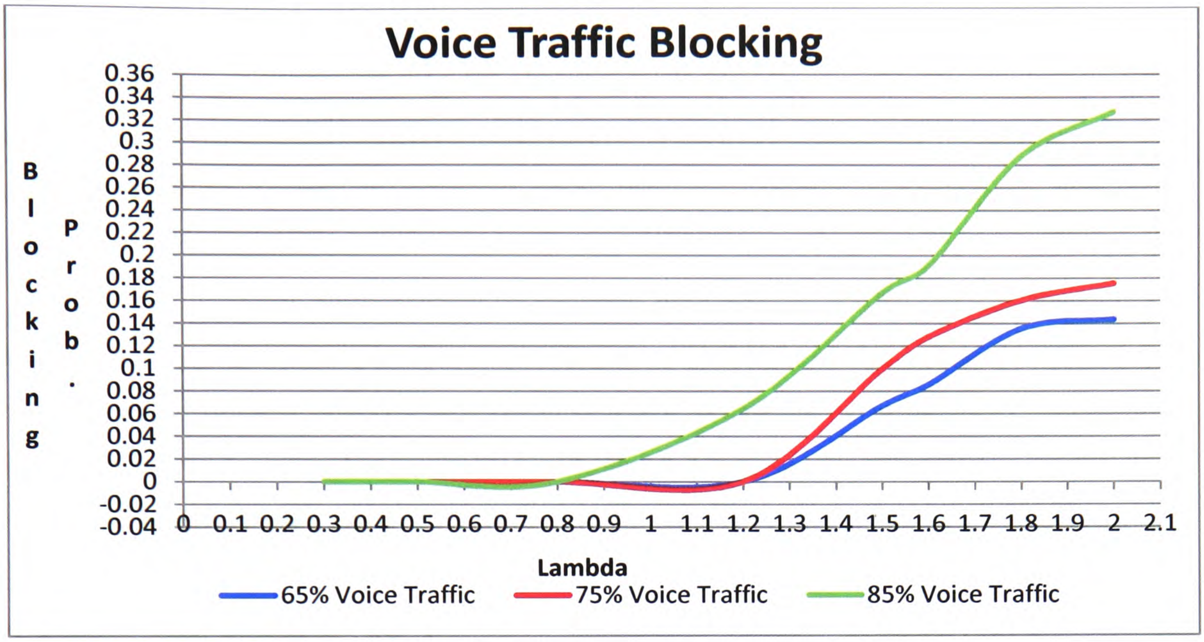

Figure 6.1 Blocking Probability for Voice Traffic

However for the streaming services, for 15 % traffic percentage as in scenario 2, the 2% blocking probability is reached when traffic intensity reaches 0.24 and for 25% streaming traffic percentage the 2% blocking probability is reached when traffic intensity is close to 0.3125 as depicted in Figure 6.2.

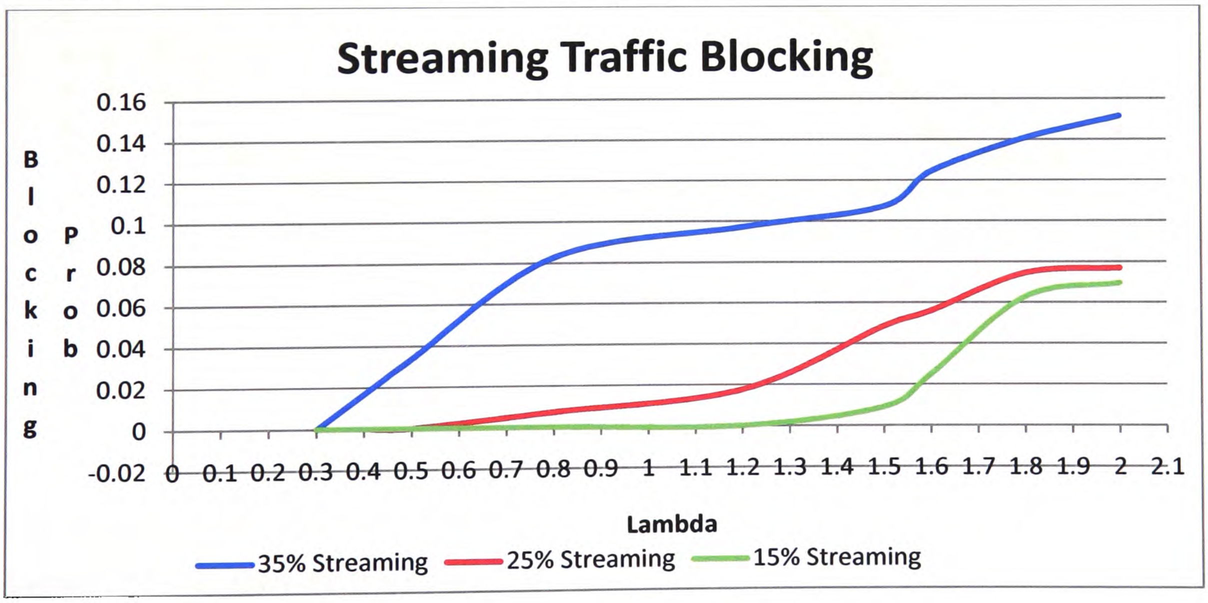

Figure 6.2 Blocking Probability for Streaming Services

However, for 35% streaming services, the blocking probability at 2% will drop down to 0.14.

From the above, it is clear that traffic distribution will influence the blocking probability and it has an inverse proportional relationship with traffic intensity. However, the best utilization of the system is achieved when the voice traffic percentage is 75% as in scenario 2 in Table 6.2

The trunking gain has been defined as the ratio between the traffic classes blocked calls if only one system exists (i.e. GSM/GPRS, 3G or HSDPA) in the site to the traffic calls that has actually been blocked as a result of the deployment of other radio technologies in the same node site. As depicted in Figure 6.3 below, the trunking gain has been improved by the deployment of additional radio access technologies in the same cell site. The deployment of different radio technologies in the same cell site increases the radio resources and enhances the customer perception of the services provided by the operators. On the other hand, the operator can deliver diverse services and will have the flexibility to distribute traffic by utilizing other radio resources.

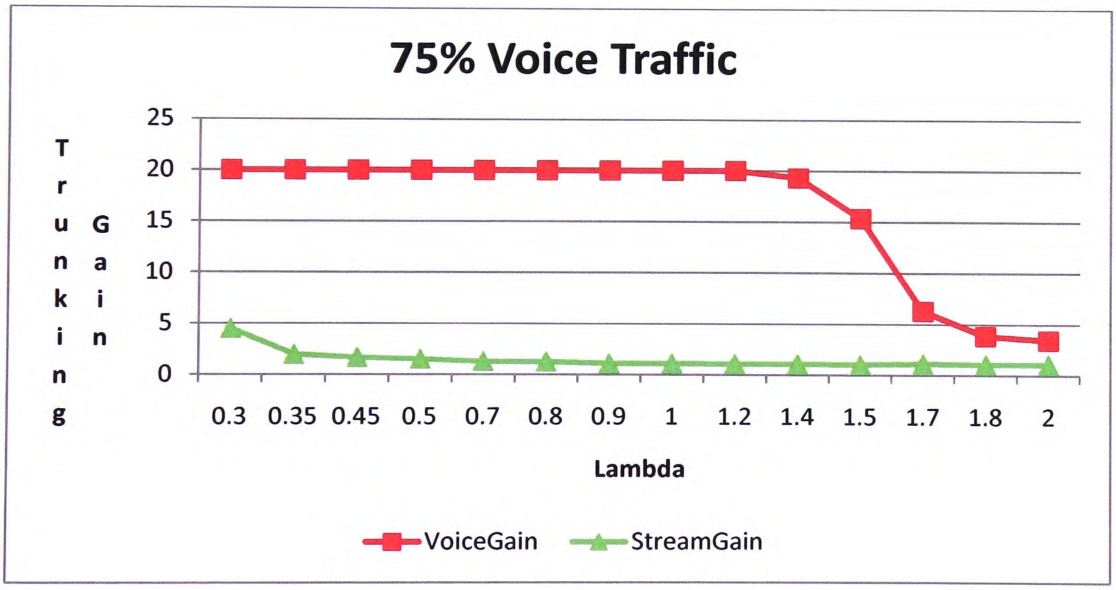

Figure 6.3 Trunking Gain for Scenario 1 Traffic mix

The trunking gain was over 100% until the system started to saturate at the traffic intensity of 1.25 then the trunking gain starts to drop. It has been noticed that even when the system is saturated, it still shows a trunking gain of 45% or better. The same has been noticed for streaming services. However, in streaming service the drop was steeper than the voice services. This is an indication of the effect of the data services on the total system when streaming services QoS requirements are stringent. The same has been noticed for scenario 2 and 3.

These scenarios show clearly that the traffic distribution affects the blocking probability and the saturation of the system. The operator can choose the traffic distribution based on statistical data available from their existing networks and from the SLA that they have with the customers.

In the next section the scenarios for the handover algorithm are discussed and results are presented.

6.5 Vertical and Horizontal Handover algorithm

Following are the basic system simulator assumption that has been considered.

6.5.1 Basic System Assumptions

To investigate the handover algorithm as described in Chapter 4 and Chapter5, the following system set up has been designed. Coverage estimation in WCDMA is different than of GSM/GPRS. In GSM/GPRS TDMA/FDMA system coverage is assumed to be constant in each base station. However, Interference has a major effect on coverage of WCDMA. Each user admitted to the cell contributes a percentage of interference dependants on user density and user required bit rate. This in turn affects the coverage of the cell as well as capacity. The quality of a signal for a required service for a certain bit error rate (BER) is measured by Eb/N0. The spreading factor represents the number of chips that is needed to carry the required bit rate; at times this is referred to as the spreading code. In [99] the authors defined a service factor as the following

$$Si = \quad SFi/(Eb/N0) \tag{1}$$

where Si is the service factor, SFi is the spreading factor for that particular service and Eb/N0 is the minimum QoS requirement for the service. As discussed in Section 5.3, five different environments have been considered, Free space, Urban, Suburban, Dense urban and Rural.

An investigated area of suburban environment as shown in Figure 6.4, 5Km square area covered by 14 nodes, 7 nodes are 3G and 7 nodes HSDPA, the nodes are overlapping and one 3G and one HSDPA node is covering the same cell area.

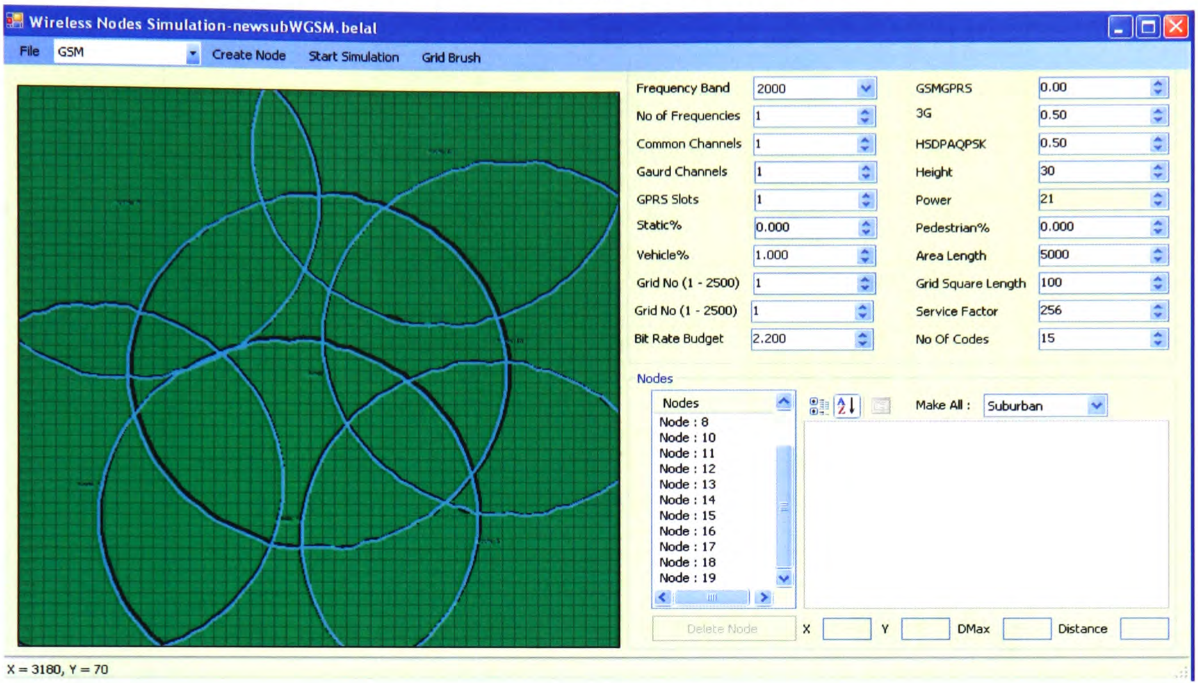

Figure 6.4 The Simulation Environment

Three scenarios have been setup to investigate the handover algorithm as described above. Traffic and user equipment distribution is depicted in Table 6.3 below.

The services in scenario 1 are limited to voice calls and VOIP streaming only. This scenario the vertical handover between 3G nodes and HSDPA nodes was enabled, in other words, there is call continuity mechanism for VOIP sessions. In all of these scenarios only HSDPA and 3G nodes are present.

Table 6.3 Traffic and UE equipment distribution

Scenarios	Traffic Service distribution	User Equipment	Traffic Percentages	GSM Present
Scenario 1	Voice 50% Load	3G/UMTS 50%	100% voice call	NO
	Streaming 50% Load	HSDPA 50%	100% VOIP	NO
Scenario 2	Voice Load 25%	3G/UMTS 25%	100% Voice	NO
	Streaming 75%Load	HSDPA 75%	100% VOIP	NO
Scenario 3	Voice 50% Load	3G/UMTS 50%	100% voice call	NO
	Streaming 50% Load	HSDPA 50%	50% VOIP	NO
			30% Video Str.	
			20% MTV	

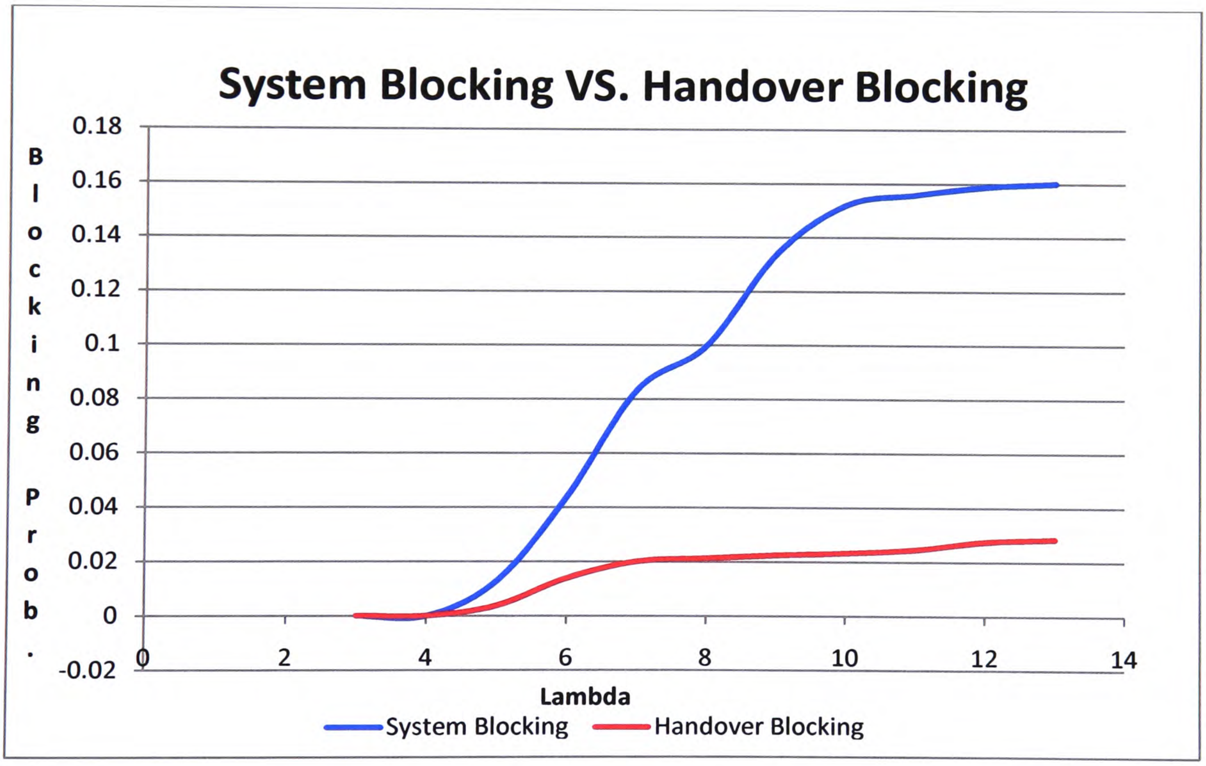

Figure 6.5 System and Handover Blocking Probability Scenario 1

The system blocking probability reaches 0.01 when traffic intensity was around 4.75 as it is clear from Figure 6.5. However, when the traffic intensity is around 4.25 the system handover blocking probability is reached 0.001.

From the results it can be concluded that the system will reach an acceptable balance between blocking probability and handover blocking probability when traffic intensity is around 4.5.

The second scenario, the streaming percentage has been changed to 75% of the total arrival traffic. The system and handover blocking probability is depicted in Figure 6.6 below. The system blocking probability and the handover blocking probability have improved compared with Figure 6.5.

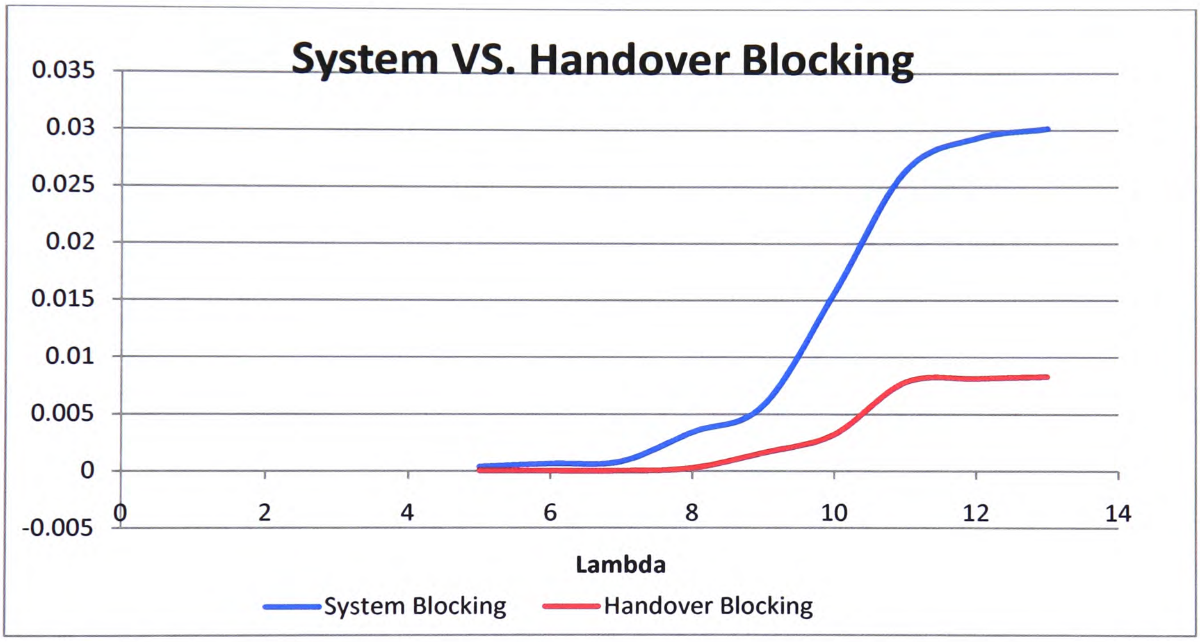

Figure 6.6 System and Handover Blocking Probability Scenario 2

The reason for the improvement is that HSDPA cells have better spectral efficiency than 3G cells. The blocking probability for the system reached 1% when the traffic intensity is around 9.25 as depicted in Figure 6.6. As far as the handover blocking probability, the percentage of .001 was reached when the traffic intensity was around 8.7.

In both cases the system gain when the 75% of the user equipments can access HSDPA has been doubled. This is an incentive for the mobile operators to speed up replacing the user equipments of their customers. It is apparent that the network utilization when

HSDPA is fully deployed will be improved dramatically by replacing the customer's user equipments.

In the third scenario, 50% of the arrivals are streaming customers with the following traffic distribution; 50% of the streaming service is VOIP, 30% is Video Streaming (144kbps) and 20% is Mobile TV (384KBPS). The effect of the streaming load on the system as a whole could not be more apparent. Figure 6.7 depicts the system blocking probability for this scenario. When the traffic intensity is around 1.25, the blocking probability is 1%. Comparing with Figure 6.5 where there were no other streaming services but VOIP, the blocking probability is almost one third. Streaming services other than VOIP can dramatically decrease the system and handover blocking probability.

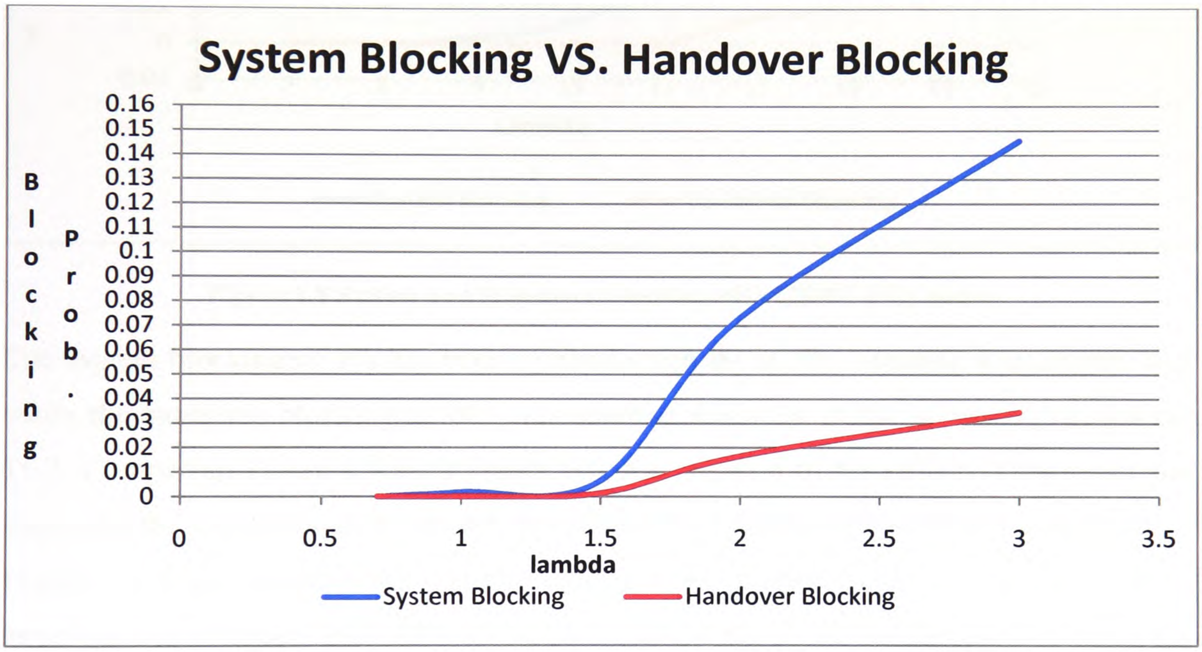

Figure 6.7 System Blocking and Handover Probability Scenario 3

The handover blocking probability reached .001 when the traffic intensity was around 1.2.

Furthermore, the same above scenarios in Table 6.3 have been studied under the assumption that no vertical handover is present. All studies have shown that the handover blocking probability regardless of the traffic mix is reached when the traffic intensity is well below 1.

In Table 6.3, scenario 2 has been repeated in the presence of two GSM/GPRS nodes. Each node was assigned 8 frequencies or 64 time slots are used for the node capacity.

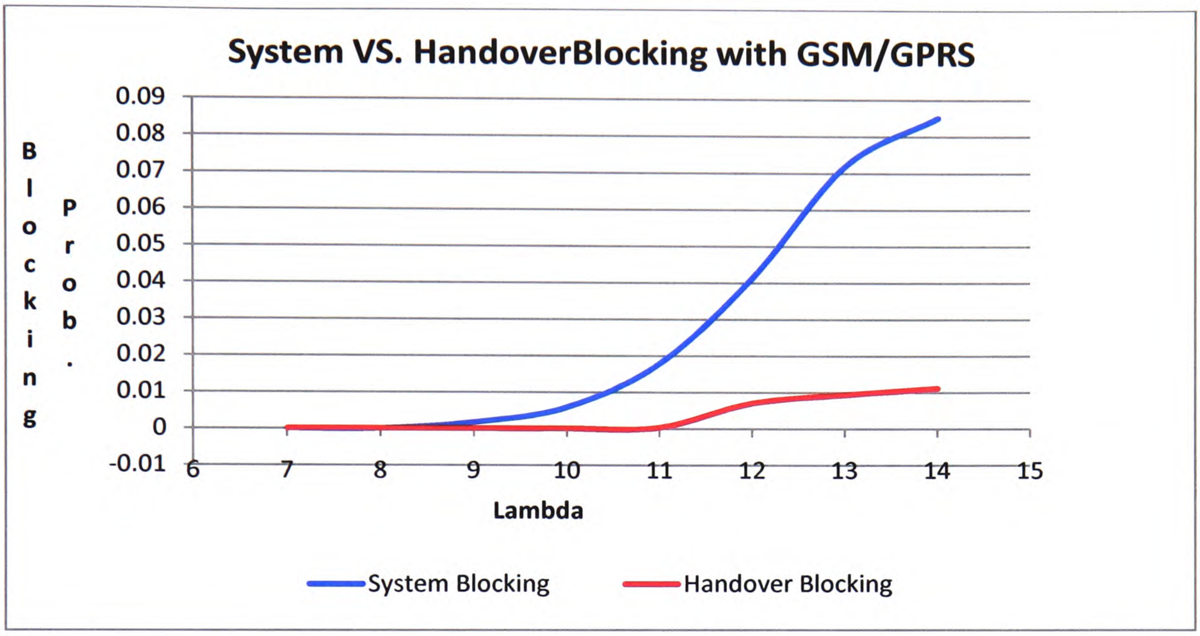

Figure 6.8 System and Handover Blocking with GSM/GPRS nodes

The system blocking of 1% has been reached when the traffic intensity was around 10.5 while the handover blocking at .001 was reached when the traffic intensity was around 11.2. Comparing Figure 6.8 with Figure 6.6, the addition of the GSM/GPRS nodes has improved the system capacity as well as enhanced the handover blocking probability.

Figure 6.9 below depicts the trunking gain for the system voice traffic. There is an improvement of more than 10% of the traffic intensity in the presence of GSM/GPRS nodes. There is also more than 22% more enhancement in the system handover blocking probability.

It can be concluded that the CRRM algorithm and the service continuity handover algorithm as proposed in this chapter can satisfy the requirements of the 3GPP standards as described in Section 6.3 and improve the efficiency of service continuity.

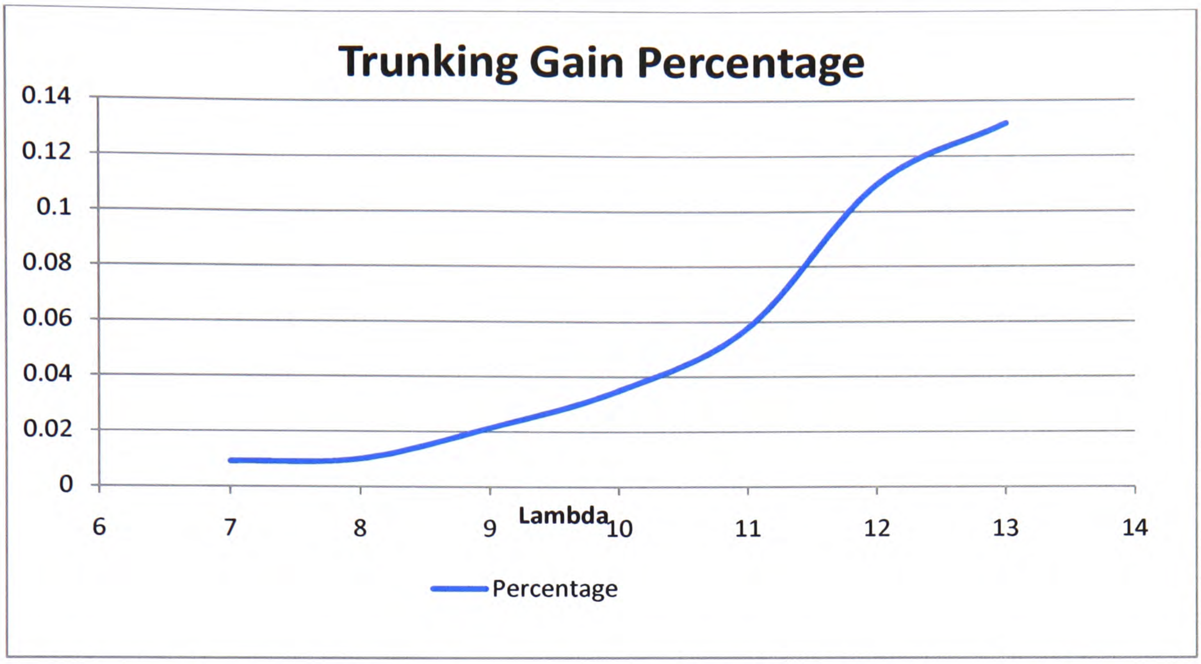

Figure 6.9 Trunking Gain for Voice Traffic

6.6 Chapter Summary

In this Chapter a CRRM Algorithm results have been presented for heterogeneous networks that enhances the trunking gain for voice and data customers. The scenario was envisioned based on total area coverage by deploying WCDMA and HSDPA resources on top of GSM to share the same cell sites. Using GSM resources as a fallback for the admission of voice and data services makes both financial and technological sense from operators' point of view. The results indicate that for voice services the system throughput or trunking gain was enhanced between 19 times. For streaming services the system throughput enhanced between 6 folds. Our approach is deterministic if the service offered is voice or streaming. However; when considering Best Effort services the Algorithm can be considered as dynamic.

Mobility is at the core of our simulator, vertical and horizontal handover have also been designed and studied. A complete handover algorithm which enables service continuity in line with 3GPP standards has been presented. It has been shown that vertical handover can support service continuity which leads to enhanced handover blocking probability of the system. A balance between handover blocking probability and system blocking has been determined. It can also be concluded that for the operators to have a better utilization of their new deployed technologies, the operators need to speed up the replacement of the existing user equipments with the new enabled HSDPA user equipments.

It has been shown that the traffic distribution can impact the system and handover blocking probabilities. At the same time the traffic distribution within a single traffic can also impact the results as shown with streaming service.

As LTE deployment has started on trial basis on Norway the above CRRM algorithm will be enhanced to include LTE nodes in the next Chapter.

Chapter 7

Long Term Evolution

7.1 Introduction

3GPP standards release 8 and 9 has introduced Long Term Evolution (LTE) Radio Access Technology (RAT) to provide broadband services to end users. Operators are likely to deploy the new radio technologies in the same geographical site as of GSM/EDGE Radio Access Technology (GERAN) and / or UMTS Terrestrial Radio Access Networks (UTRAN). LTE at the beginning will share the same frequency band with GERAN and UTRAN as no dedicated frequency has been assigned yet.

In this Chapter an extended CRRM algorithm to efficiently utilize the additional available network radio resources in multi radio environment based on service requests is presented. The proposed algorithm efficiently distributes the radio resources available based on the services standardized by 3GPP.

7.2 LTE Performance

Using a large bandwidth up to 20MHz while utilizing high order 64QAM modulation and multi stream transmission allowed LTE to provide high peak data rates. QPSK modulation carries 2 bits per signaling element, while 16QAM carries 4 bits and 64QAM carries 6 bits. Using multi streaming can further double the peak data rate up to 12 bits per signaling element. The theoretical peak data rate can reach 325Mbps in the downlink.

Since the deployment of LTE will be in different bandwidth sizes that range from 1.4MHz up to 20MHz and not fixed to 5MHz, the peak data rate scale back with the bandwidth. Thus, this shall be an incentive to the operators to utilize their bandwidth in a more efficient manner.

The evolution towards LTE for some operators is essential because of the reduced capital expenditure compared to 3G networks. This in turn will be a cost effective measure for the operators to offer large bandwidth at a reasonable price.

Internetworking with GSM/GPRS technologies can be done through the optimization of IP Multimedia Subsystem (IMS) control system. Thus, the evolution path is not necessarily had to go through 3G/3.5G technologies.

As it was the case with the deployment of HSDPA, the demand for data and video services has increased; the same will happen with the deployment of LTE. The operators can elevate the pressure of providing higher data rates from the 3G/3.5G and use LTE as an overlay network. This can also be true for the GSM/GPRS only technology operators.

In the following, a discussion on extending the algorithms proposed in Chapter 4 and Chapter 5 as well as several scenarios envisioned to study the implications of LTE deployment.

7.3 Enhanced CRRM - LTE

The CRRM algorithm in Chapter 4 and Chapter 5 has been extended to include the new standardized technology LTE. Below as depicted in Figure 7.1 is a description of the modified algorithm and the simulation assumptions and results.

With the inclusion of LTE in the algorithm design; two more services have been envisioned for inclusion, non real time (NRT1) at 512kbps and non real time (NRT2) at 1024kbps. These two services will be allowed access only on LTE because they will consume a considerable portion of the bandwidth if they are allowed access to other technologies. However, as the operators are gaining more insight into the LTE technology this might change. Another reason for such decision is that, some operators have not deployed 3G/3.5G technologies and would like to deploy LTE alongside GSM/GPRS, in which case bandwidth consuming services like NRT1 and NRT2 would exert a bandwidth burden on GSM/GPRS technology.

- **Voice Calls**: As described in Section 5.6.3.2.
- **Real time streaming Services**: As described in Section 5.6.3.2.
- **Non Real Time Services**: LTE Node is tried first, if capacity available then the customer is admitted else if the traffic is MTV then, try HSDPA and if capacity

is available then admit else block customer. If the requested service is NRT1 (512kbps) or NRT2 (1024kbps) then it is blocked without attempting on HSDPA.

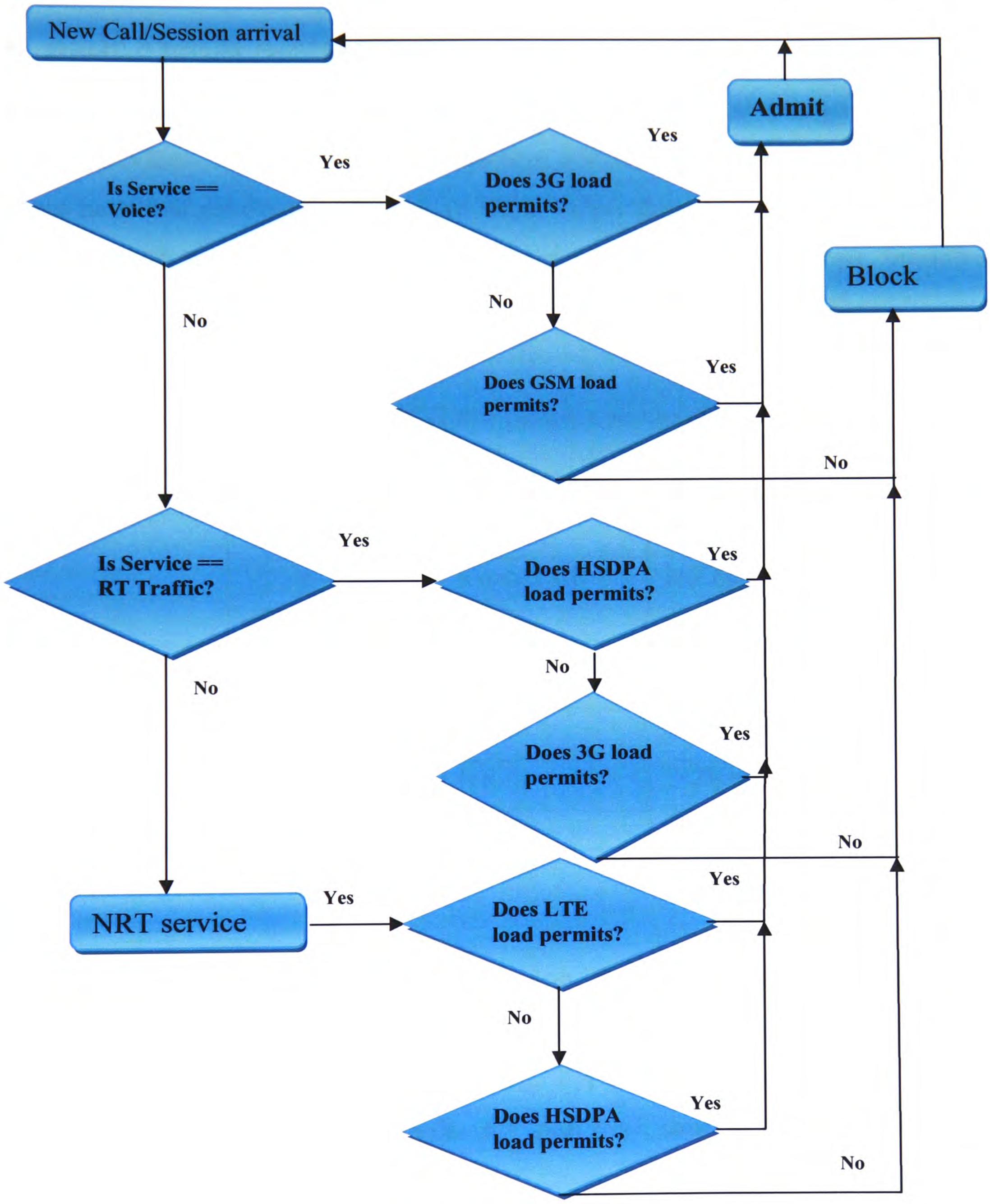

Figure 7.1 CRRM LTE Algorithm

- **Best Effort Services (NRT)**: If UE is LTE capable then LTE is tried first, if LTE capacity or UE is not advanced then tried on HSDPA. Otherwise, tried on GPRS as per section 5.6.3.2.

7.4 Basic System Assumptions

The services assumed for this extended algorithm are voice, real time streaming and non real time services. The voice has one subclass which is voice call, real time streaming services have two subclasses VOIP at 16 kbps, Video streaming at 144kbps and non real time broadband services have three subclasses; mobile TV at 384kbps, broadband at 512kbps and 1024 Kbps. The traffic classes are depicted in Table 7.1 below.

Table 7.1 Traffic Classes Considered

Traffic Class	Sub Class	Data Rate
Voice	Voice call	16Kbps
Streaming real time	VoIP	16Kbps
	Video Str.	144Kbps
Streaming Non real time	MTV	384Kbps
	NRT1	512Kbps
	NRT2	1024Kbps

The voice and VoIP call is modelled through Poisson exponential distribution and assumed to have an average service time of 120 seconds where video streaming have an average service time of 200 seconds [103]. For non real time services mobile TV average service time is 300 seconds, for non real time is 400 and 600 seconds respectively.

The inter arrival time is also using Poisson exponential distribution with varying average inter arrival values. It is also assumed that once the call is admitted to the system it will be retained until the call is finished. All customers are static and arrival is uniform over the investigated area. Four nodes are used with 100 percent coverage. GSM is assumed to have 9 frequencies, 72 time slots, 62 for voice and at least one reserved for GPRS, the rest are used for signalling. 3G node is used and load is quantified at 75% load factor is used corresponding to 6 dB noise rise in the uplink for 3G. HSDPA total bit rate budget is assumed to be 2.2 Mbps [25]. LTE bit rate budget is assumed to be 35Mbps using single stream. Other basic system assumptions included as described in Section 6.2.1. If the user is in favourable link conditions then node data throughput is reflected by the simulator as modulation and coding technique improves to double or treble the data throughput by using 16QAM or 64QAAM respectively.

7.5 Simulation Setup

The simulator is used to run two scenarios as in Table 7.2 below.

Table 7.2 Simulation Scenarios

Scenario	Traffic Type	Traffic subtype	Traffic Percentage
Scenario 1	Voice	50% of arrival	100% Voice
	RT stream	25% of arrival	100% VOIP
	NRT stream	25% of arrival	50% MTV
			30% NRT1
			20% NRT2
Scenario 2	Voice	25% of arrival	100% Voice
	RT stream	50% of arrival	100% VOIP
	NRT stream	25% of arrival	50% MTV
			30% NRT1
			20% NRT2
Scenario 3	Voice	25% of arrival	100% Voice
	RT stream	25% of arrival	100% VOIP
	NRT stream	50% of arrival	50% MTV
			30% NRT1
			20% NRT2

For LTE, one terminal is modelled capable of downloading NRT1 and NRT2 traffic with traffic distributions as depicted in Table 7.2.

Figure 7.2 depicts the blocking probability of voice traffic for scenario 1 and scenario 2. From the figure, it is noted that when the voice traffic percentage increased to 50%, the blocking probability reached 2% when the traffic intensity is around 1.22. However, when the voice percentage has been decreased to 25% the blocking probability mark of 2% has reached when the traffic intensity was around 1.38. This is clear evidence that traffic distribution does affect the blocking probability of the system and that the operators need to conduct traffic surveys for the sites before RAT deployment.

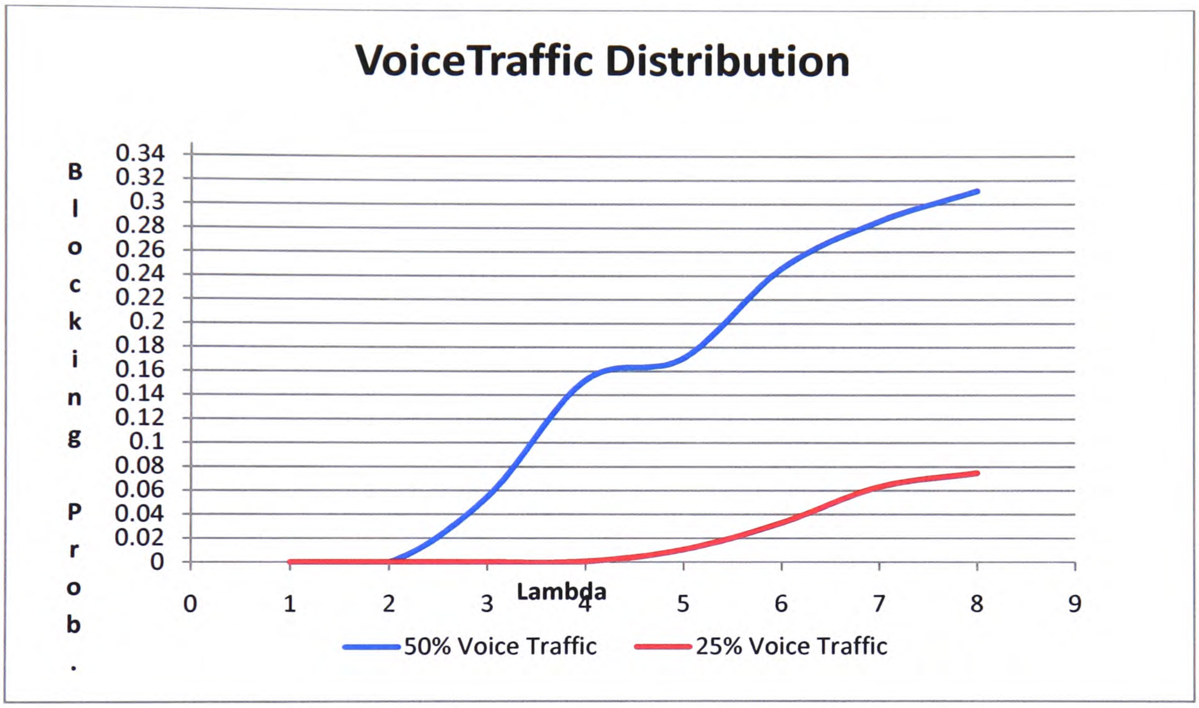

Figure 7.2 Voice Traffic Blocking Probability S1 and S2

Figure 7.3 depicts the blocking probability for VoIP scenario 1 and scenario 2:

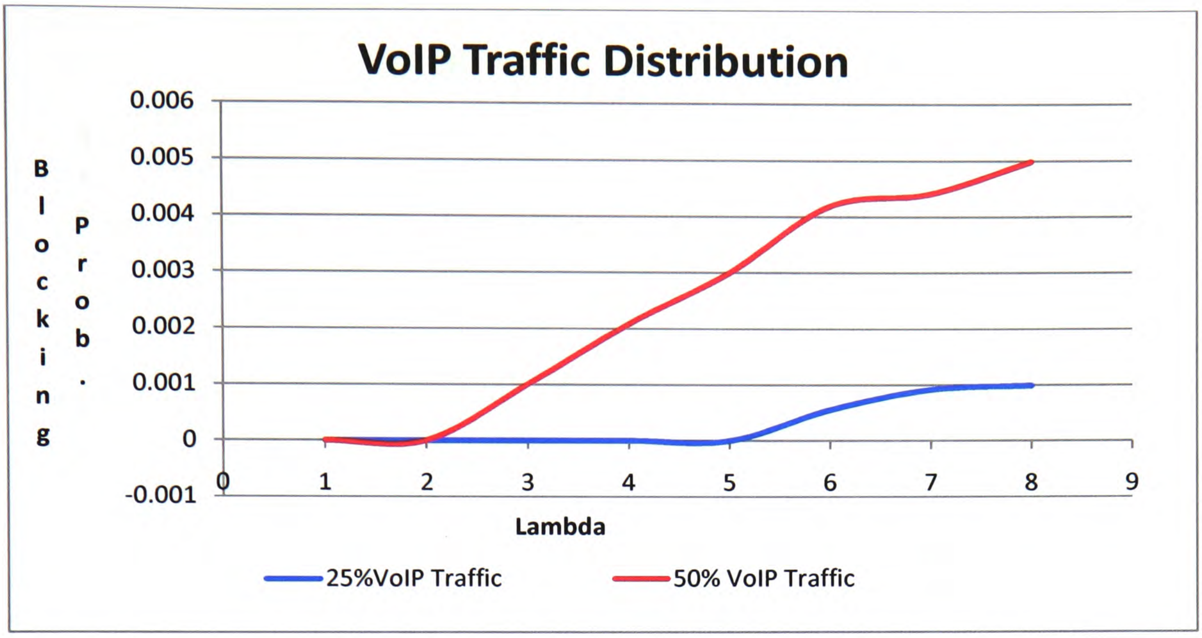

Figure 7.3 VoIP Blocking Probability S1 & S2

The blocking gain has improved from Figure 7.2 as to Figure 7.3 because of change in the traffic class. Figure 7.3 shows clearly the benefits of deploying new radio access technologies capable of delivering real time streaming services as VoIP. This also clearly shows that HSDPA has a better spectral efficiency than GSM and 3G and at the same time provide an incentive for the operators to replace the customers UEs to be capable of accessing HSDPA.

Figure 7.4 depicts the blocking probability of the same percentage of voice traffic class and VoIP traffic class which confirms the benefits of new RATs deployments.

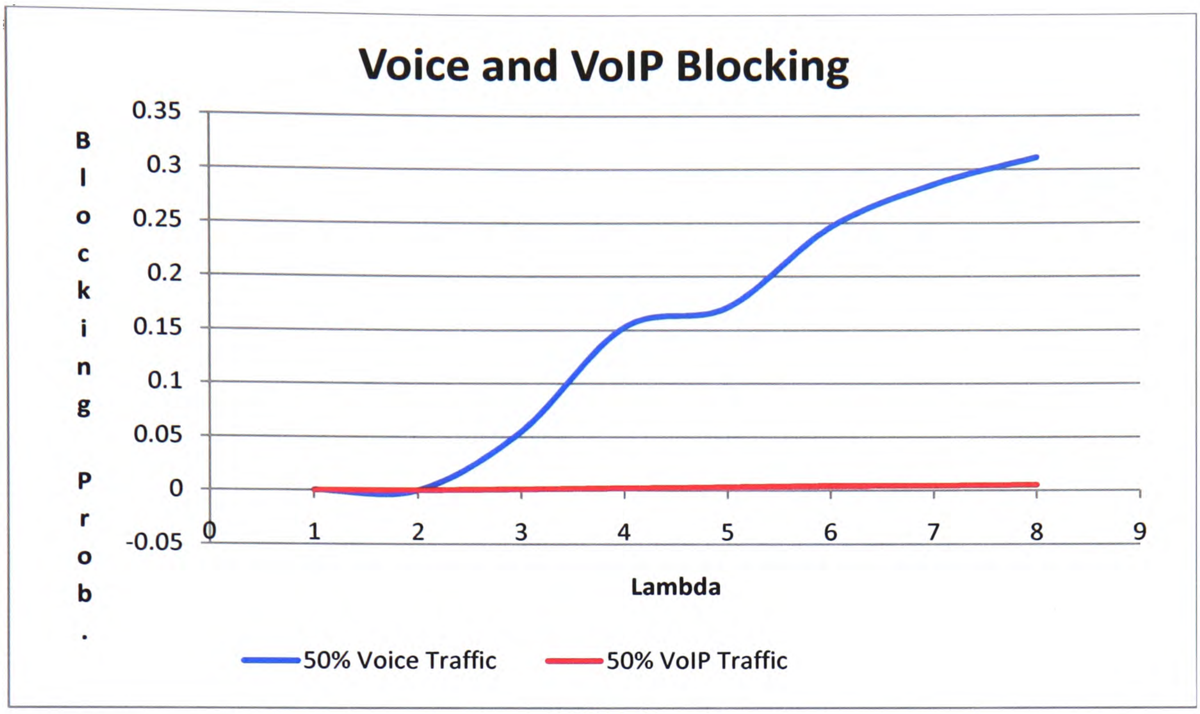

Figure 7.4 Voice and VoIP Blocking Probability

Figure 7.4 shows a comparison of the blocking probability when the percentages of the traffic distributions are equal. From the above it can be shown that the blocking probability of voice class traffic reached the 2% mark when the traffic intensity is around 1.22. However, the blocking probability for VoIP traffic class stayed flat for up to and including when the traffic intensity reached 8.

The above results demonstrate clearly two issues to the operators. First, the benefits of the deployment of HSDPA for real time services especially VoIP. Second, the benefits of directing all non real time traffic towards LTE as this will elevate the bandwidth stress from 3G/3.5G networks because of NRT bandwidth demand. Third, for the operators to benefit from all the above they need to speed up the replacement of the customers UEs to take advantage of the new infrastructure.

In the following scenario 3 as in Table 7.2, the NRT was changed to represent 50% of the traffic while the streaming traffic (VoIP) and the voice traffic are kept at 25% each of the total traffic intensity; Figure 7.5 depicts the Blocking Probability for such scenario.

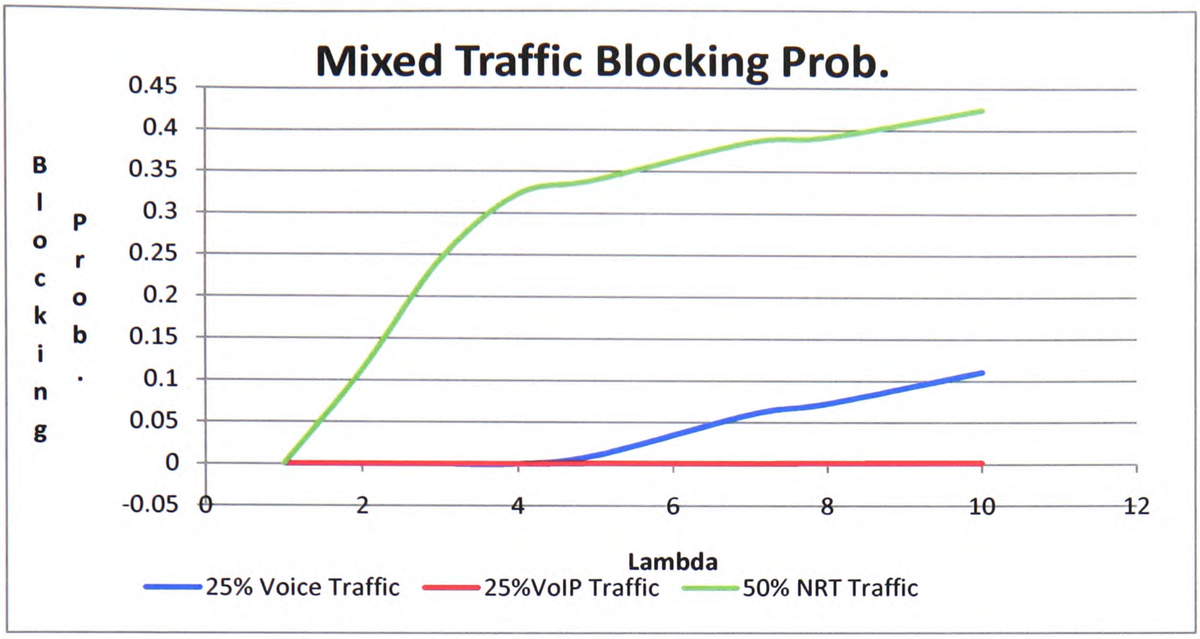

Figure 7.5 Mixed Traffic Blocking Probability

Comparing the results of Figure 7.4 and Figure 7.5, the voice traffic blocking probability has been enhanced because of the reduction in the percentage of the traffic. The same observation can apply to VoIP streaming traffic. However, when the percentage of the NRT traffic has been increased to 50% of the traffic intensity, the NRT blocking probability reached the 2% mark when the traffic intensity was around .75.

It should be mentioned that for NRT traffic, the operators can live with higher than 2% traffic blocking probability. Another issue with this study is that the NRT traffic was confined to LTE technology which might not be a suitable scenario for service continuity.

In all the above scenarios, the NRT traffic as per Table 7.2 was directed towards LTE technology. However, the average service time has been specified at 400 seconds for 512Kbps service and 600 seconds for 1024Kbps service.

To be consistent with the packet switched IP traffic, the bandwidth is shared among all users; the average service time should be calculated based on the instantaneous available bandwidth. All the above scenarios in Table 7.2 have been studied again with this new concept in mind. The CRRM algorithm has been extended to reflect this new requirement. As shown in Figure 7.6 below,

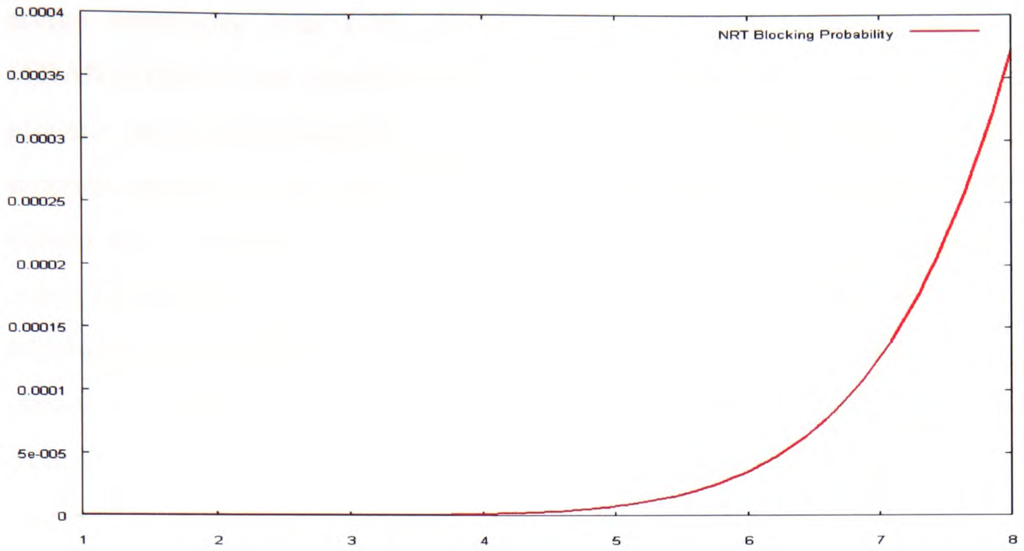

Figure 7.6 NRT Blocking Probability

The blocking probability for the LTE is not noticeable which reflects the capacity of LTE nodes.

LTE capacity was modelled at 35Mbps in the scenarios studied above and the effect of that capacity cannot go unnoticed as it enhanced the system throughput. Thus, LTE capacity is anticipated to answer the data rate issues and elevate the bottleneck problems of the radio technologies. Figure 7.6 shows clearly the advantage of having LTE deployment in the same cell site as of GERAN/UTRAN as the capacity that LTE bring is a clear benefit to all parties involved operators and consumers.

The real time video streaming was not included in this scenario but there is no restrictions in including the service if needed required for further studies.

7.6 VoLTE

The initial deployment of LTE will be to satisfy the demand for high data bit rates where wireless broadband technology is demanded and considered the only viable solution.

Since LTE is packet switched IP network all envisioned services are deployed on compliance with the TCP/IP architecture. However, because of the capacity and coverage of LTE technology, many operators would like to provide seamless voice service continuity over LTE. Service continuity and handover between LTE and UTRAN/GERAN can present many technical challenges to the operators. First, voice calls are carried traditionally in the circuit switched domain, while LTE is all IP packet switched networks. The voice call needs to be transferred from the CS domain to PS domain and vis versa. Second, a new UE has to be of a single radio technology, the reason behind this is that the deployment frequency band of LTE will be initially be of neighbouring proximity of UTRAN/GERAN. Having a terminal that will simultaneously support signalling on both frequencies presents a severe technical challenge to the manufacturers [108]. This is the reason why a SRVCC capable terminal has been proposed to resolve such an issue. Third, transferring a voice call between PS domain and CS domain requires a controlling mechanism for call continuity and handover. It is worth noting that many operators would like to inter network directly from GSM/GPRS to LTE. The proposed controlling mechanism for voice call continuity is IP Multimedia Subsystem (IMS).

7.6.1 IP Multimedia Subsystem (IMS)

Several functional entities comprise the IMS network, below is a brief description of such entities [109]:

- **Call Session Control Function (CSCF)**: This entity is responsible for establishing, monitoring and release of multimedia sessions which uses SIP signalling to register users with IMS.
- **Media Gateway Control Function (MGCF)**: The main responsibility of this server is to control the allocation and de-allocation of the resources and modify the resources if needed. The MGCF communicates with the CS domain.
- **Breakout Gateway Control Function (BGCF)**: The main responsibility of this controller is to choose the network to connect the call or session based on operators policies.

There are more entities to the IMS, the above falls within the interest and the scope of this thesis.

The IMS will provide a service control subsystem to the voice call to enable service continuity functions for call originating, voice call terminations and domain transfers between CS and PS. This will provide an opportunity for the operators to utilize LTE to transfer VoIP calls to and from CS domain through the utilization of IMS domain.

In anticipation of this future work the simulator has been extended one more time as follows:

In keeping with the technology evolution and the inter-networking between different radio access technologies to provide voice call continuity, the traffic class considered was either voice calls or VoIP. One more additional SRVCC capable UE was modelled. This UE is capable of accessing CS domain or PS domain. A single GSM/GPRS node with 8 frequencies along with one legacy 3G node, one HSDPA node with bit rate budget of 2.2Mbps and one LTE node with bit rate budget of 35Mbps in the same site. The simulator was run for extended period of time with different traffic intensities.

It was not possible to collect any statistics in relation to blocking probability, which confirms the expected results when all the above radio resources are deployed together. This is an incentive for the operators to deploy LTE and speed up the replacement of the customers UEs.

This is also a motivation for the operators to deploy LTE even in the absence of 3G/3.5G networks.

Another scenario was carried out to illustrate the performance of LTE technology. One LTE was deployed with cell throughput at 50Mbps. Only one NRT service was modelled for this scenario where the user requested service of 1024Kbps is considered. The user is assumed to have a file size .625MB (5*10^6 bits) for download. The system will consider blocking in the case where it cannot deliver the 1024Kbps service. The service holding time depends on the available users in the system and is calculated by the simulator.

In the following, Figure 7.7 illustrates the average user throughput.

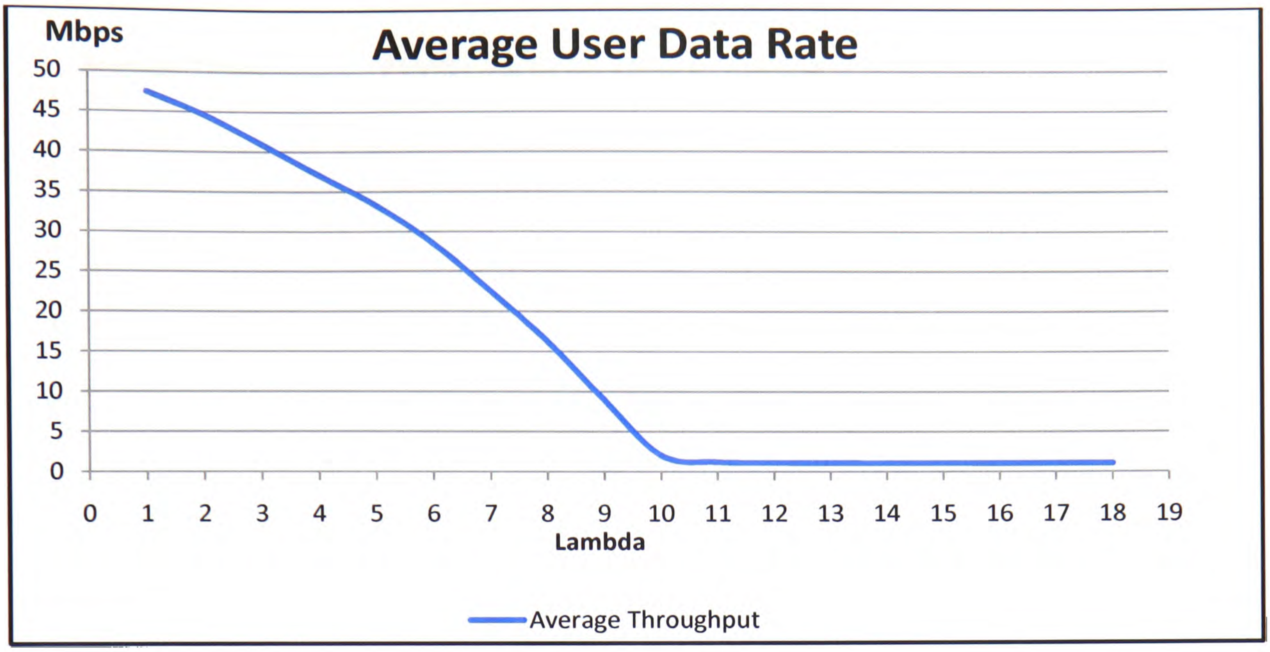

Figure 7.7 LTE User Data Rate

It is clear from the above figure that the average user data rate is decreased as the number of user increase in the system. When the system is saturated, the user minimum data rate was holding at approximately 1024Kbps.

When the user average service time was considered as illustrated in Figure 7.8, as the number of users increases in the system the service time increases as well. This demonstrates that in shared all-IP networks the average service time is inversely proportional to the user bit rate. This is because the system bandwidth is shared among all the users of the system equally.

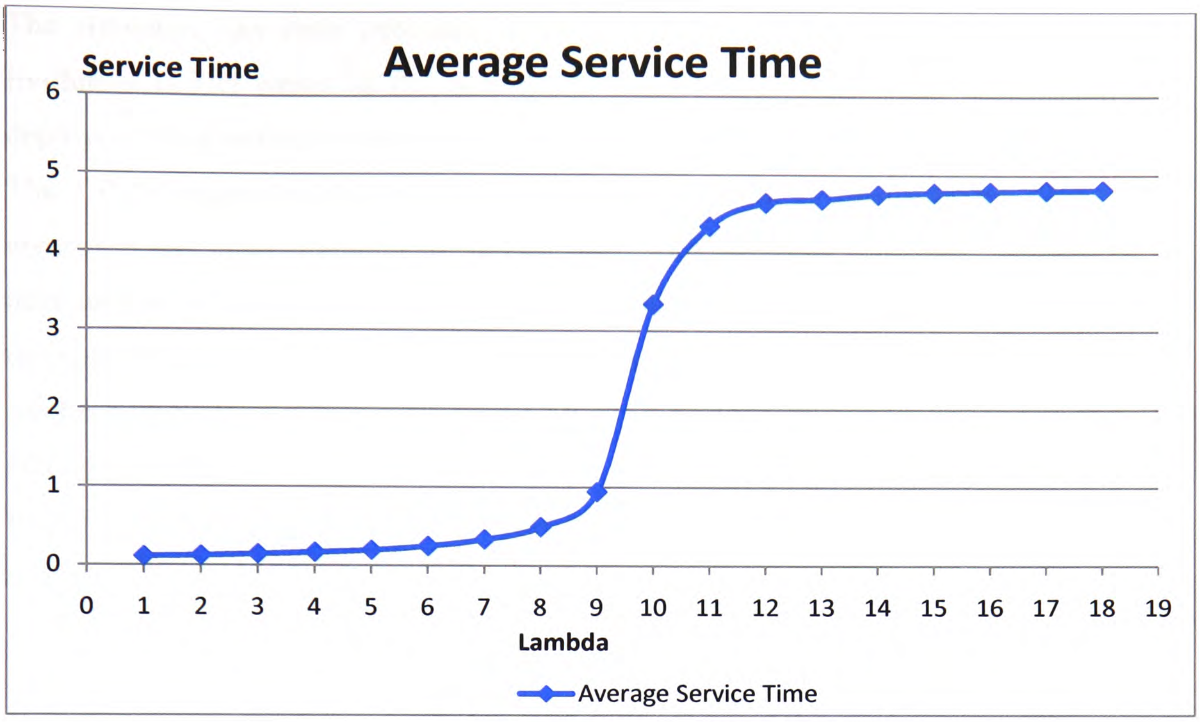

Figure 7.8 User Average Service Time

7.7 Chapter Summary

Deploying WCDMA on the existing GSM networks has its advantages such as there is no need for site acquisition. The operators experience with traffic densities on GSM can help in initial deployment as it makes more economical sense to the operators and it is easy to implement. With the potential introduction of LTE, coexistence of GERAN/UTRAN and LTE become curtail. Therefore, several studies to determine the effect of such deployment have been conducted by 3GPP [12]. The studies showed that the effect of LTE RAN deployment in an existing GERAN/UTRAN site is minimal. The throughput loss in UMTS cell Capacity is around 3 percent when considering the total interference caused by transmitter to the adjacent channel receiver called Adjacent Channel Interference Ratio (ACIR) and the transmitter mean power called Adjacent Channel Leakage Power Ratio (ACLR). ACIR is dominated by ACLR [104-105]. LTE coexistence with GERAN/UTRAN is very important because there is no dedicated frequency band assigned for LTE yet. Therefore, LTE initial deployment frequency will be in the same frequency band as UTRAN and GERAN. From a geographical location deployment, it is obvious the benefits of sharing sites with other RATs.

The simulator has been extended as one more module is included for Long Term Evolution (LTE) which is the new 3GPP radio access technology envisioned to be deployed in the next few years.

The CRRM algorithm has been extended as presented in this Chapter. The CRRM algorithm has been extended to service only NRT services. However, because of the introduction of SRVCC and the envisioned radio access technologies inter-operability, the algorithm has been extended again as explained in Section 7.6.1. Another extension has been implemented by considering LTE as all-IP network and the bandwidth is shared among users.

In the same way, the handover algorithm can be extended with minor changes to include new Radio access technologies.

Chapter 8

Conclusions and Future Work

8.1 Summary of the Thesis

The aim of the Ph.D. project presented in this thesis was to propose and analyze a set of algorithms to address the problem of multi access deployment in the presence of multi service environment. The first algorithm proposed was for initial radio resource management call admission control algorithm that is a multi criterion decision making algorithm which takes into consideration the user service request, the required minimum quality of service, the user equipment capabilities and the available radio access technologies. The second algorithm proposed was for service continuity based on user mobility and services rendered to the user. Call retention is more important for the operators than initial call admission control to provide ubiquitous connectivity for the call as the users' mobile between different nodes coverage areas. The service continuity algorithm decision making criterion is based on available radio access technologies, the network load, users' service and users' equipment capabilities.

To enable the study and analysis of the above algorithms, a simulator system was designed and implemented, based on discrete event simulation method the simulation tool was built in a C# Windows environment. The designed tool is modular and scalable; each class can be split, added or removed based on the environment of the scenario investigated. The simulator is based on 3GPP technologies and is scalable as more radio technologies can be added without any noticeable negative impact on the system. The main findings and contributions of the research project can be summarized as follows.

8.1.1 Common Radio Resource Management Algorithm

The customers and applications have been starving for more data throughput while operators are recognizing that radio access technologies are the bottleneck, they are stepping up RAT deployment to deliver the promised high data rates to customers. Operators are using existing traffic information from GSM/GPRS technologies to deploy WCDMA, HSDPA and LTE systems. The deployment of WCDMA in the existing GSM/GPRS cell sites has proven to enhance the network resources available to the operators while the initial deployment of HSDPA have shown improved data rate offering and the opening of new era of video communications and streaming services over cellular networks. The diversity of the services and the availability of several radio access networks to choose from to deliver such services make the management of available resources rather complex. Therefore, Common Radio Resources Management (CRRM) has been standardized by 3GPP.

In a multi service multi radio access environment, the traditional algorithms as presented in Chapter 3, are not efficient and cannot be utilized. The proposed new CAC algorithm in Chapter 4 is designed to distribute the available network radio resources in multi radio environment based on service requests. The proposed algorithm efficiently utilizes the radio resources available based on the services supported in 3GPP standards; the algorithm decision making process is based on several criterion, like users' equipment capabilities, network load and available RATs.

In Chapter 6 several scenarios have been investigated to provide an insight into the algorithm and what limitations might arise from the mobile core network. The traffic distribution has proven to carry a significant effect on the system capacity. The user equipment has also its effects; it has been found out that if a customer is using advanced mobile equipment, the system will be better equipped to provide the requested service.

In Chapter 7, an extension of the CRRM algorithm to study the performance of LTE technology has been proposed. LTE has been evaluated in the presence of other technologies like GSM/GPRS, 3GUMTS and HSDPA.

8.1.2 Vertical and Horizontal Handover Algorithm

Providing service continuity is a key objective of mobile network operators. Operators are deploying multi radio access technologies to provide customers with services any time anywhere. However, this diversity may cause call drops which has a severe negative impact on customer perception of the service and on operators revenues.

Handover is maintaining service continuity while mobile station is moving from one node coverage area to another coverage area associated with different base station or node.

Enabling seamless handover especially vertical handover is a challenge for the operators. The decision to move the service within the same technology or to a different technology seamlessly while maintaining QoS is becoming increasingly complicated. The main reason is the existence of different RATs in the same cell site. In addition to the radio resources, the diverse services as presented in the standards [20] offered to customers further complicate service continuity issues. Several scenarios have been presented and analyzed to enable service continuity based on a complete algorithm for horizontal and vertical handover.

Moving a call/session from one system to another (Vertical Handover) requires more processing power and several networking entities will be involved than transferring a call/session within the same system. This is the reason why the proposed algorithm is giving priority to the horizontal handover. The initial handover decision is based on cell coverage and path loss calculations. Each cell has an average cell distance that it covers depending on the environment and service parameters. If the subscriber measurements fall outside these two parameters, this will trigger the initiation of the handover process. The call/session will be tried on target cells that are of the same type as the serving cell (intra system/Horizontal) if the target cell has enough capacity. If this fail, then the call/session is transferred to a cell of different type than the serving cell (inter system/Vertical) if there is enough capacity, otherwise the call/session is dropped.

In Chapter 6, a complete handover algorithm has been presented, several factors influences the decision making process as the user mobility, user equipment capabilities, networks load and user service.

The proposed multi criterion handover algorithm is essential to provide service continuity which leads to enhanced blocking probability of the system. The results show that a balance between handover blocking probability and system blocking can be stroked. The results also show that the operators will have a better utilization of their new deployed technologies, if they speed up the replacement of the existing user equipments with technology advanced UEs. It has also been showed that the traffic distribution can impact the system blocking probability. Three scenarios of traffic distribution have been investigated. The impact of enabling mobile TV traffic and video streaming can severely degrade both the blocking probability of the system and the Handover blocking probability which leads the operators to speed up the deployment of LTE field trials.

8.1.3 Simulation System

From the literature review in Chapter 3, it was clear that there is a lack of comprehensive simulation system that comprises 3GPP technologies to evaluate the proposed algorithms in the literature or the algorithms that was proposed in this thesis in Chapter 4. The need for this essential academic simulation resource prompted the design of the simulation system as in Chapter 5.

The simulation system was designed in Windows platform using visual studio C#, the class library that is provided by the Windows environment and its object oriented functionalities make it suitable for modelling the different objects and link the classes as per the logic of operations in accordance with the 3GPP standards and other scholar research publications.

The simulator processes were designed as modular classes or methods where each method or a class is called according to the logic of execution. This will enable for smooth future addition or changing of technologies and functionalities. It also can change the scope of investigations and usage of the simulation tool in a fundamental way without changing the core of the simulator. The simulator is designed as DES at the system level; however the simulator can be expanded to run simulation studies at the layer level without changing the simulator core.

The simulator is scalable to incorporate other new radio access technology based on 3GPP standardizations.

One of the main inputs to the simulator is the topology which consists of the node class and the propagation environment. The node class consist of four entities, GSM/GPRS, 3G/UMTS, HSDPA and LTE. Each node has its own properties and radio resources that are attached to the node upon creation. Each node coverage area depends on a specific predefined requirements and drawn on the fly by the simulator system before the simulation process is activated. The propagation environment is modelled based on extended HATA COST-231 model equations as described in [92, 98]. The investigated area is divided into grids; each grid has its own terrain, population density, speed and height. An interface has been designed in the simulator system where all of these attributes, topology and functionalities can be loaded to the simulator before the execution of the core simulation process.

The core simulator has several entities and functionalities within these entities; one of the main entities is the scheduler which was designed to keep the time correlations between the events of the simulator. Two main events are stored in the scheduler, the arrival event and the departure event. Based on the event time the simulation process fitches the event from the scheduler and executes the process, all other processes are executed within each of the two main events; this keeps the simulation process run until end of simulation time. Mobility is at the core of the simulation process, where each user is assigned an average speed and directions upon the creation of the user object. Mobility is most important when studying and analysing service continuity algorithms in the form of handover whether horizontal or vertical.

Two main algorithms have been incorporated in the simulation core; call admission control and handover algorithms. Several criterions have been incorporated in the two algorithms, like the presence of the network, propagation loss, user mobility, network congestion, user equipment capabilities and service type.

8.2 Suggestions for further research

The objectives for this research thesis as presented in Chapter 1 have been met; however there are still areas of possible further investigations. One idea immediately comes to mind; to reach a complete convergence in accordance with 4^{th} generation networks, Wi-Max technology needs to be simulated and be part of the simulation system.

Additional work in this thesis could be extended in several directions including:

- Optimization of the services by the call admission control algorithm. Some services especially the packet switched services allow for optimization to accept other connections while down grade the QoS of the existing streams.
- Optimization of the services by the handover algorithm. Some services can be also downgraded to a different QoS when changing point of attachment due to mobility, this also applicable to streaming services.
- Expanding the simulator to incorporate other technologies that envisioned to be deployed in the same cell site as of the existing technologies.
- Expand the simulator to investigate scenarios and studies on the packet level as well as the system level.
- Expand the handover algorithm to include LTE technology.
- Performance analysis and studies in the presence of LTE and Wi-Max technologies.

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