ADDIS ABABA UNIVERSITY

ADDIS ABABA INSTITUTE OF TECHNOLOGY

SCHOOL OF ELECTRICAL AND COMPUTER ENGINEERING

Quality of Service Evaluation of Voice over UMTS Network: The case of Addis Ababa

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Declaration

I, the undersigned, declare that this thesis is my original work, has not been presented for a degree in this or any other university, and all sources of materials used for the thesis have been fully acknowledged.

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This thesis has been submitted for examination with my approval as a university advisor.

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Abstract

In mobile communication systems the demand for real-time and high quality services, such as voice and video, is on the rise. Universal Mobile Telecommunication System (UMTS) is one candidate technology to fulfill the above demand. Quality of service (QoS) in UMTS network is the capability of UMTS service providers to provide satisfactory services to their customers. On the other hand, quality of experience (QoE) is how a user perceives the usability of a service when in use.

QoS can be measured from a network perspective (via the network management system) and from a user perspective (via drive test system). In UMTS network, the former QoS measurement is called control plane system while the latter is called the user plane system. The control plane system generates QoS indicator parameters such as network geographic observation; network coverage and quality analysis; and network performance analysis parameters. The user plane system on the other hand generates QoS indicator parameters such as coverage analysis; quality analysis; accessibility; retainability and mobility parameters.

The current UMTS network infrastructure deployed in the Addis Ababa city, which is solely managed by ethio telecom, is undergoing major expansions in the last 4 years and resulted in a tangible improvement of coverage and quality. However, there are complaints from subscribers from various parts of the city. This thesis intends to evaluate the voice transmission QoS of existing UMTS network in the city of Addis Ababa.

The evaluation is made by analyzing collected real data from both control plane and user plane systems, by using industry-standard analysis tools such as Element Management System (M2000), Performance Surveillance (PRS), Nastar, Nemo handy, Nemo outdoor and Actix Analyzer.

The analysis results show that, in general, there are some disparities between the ethio telecom targets and analysis results, which indicate the need to further improve the voice QoS. To improve the quality of voice transmission the recommended interventions include: implementation of QoS manager in different levels of network, appropriate resource allocation in the network, check if the inter node B distance or inter system distance (ISD) meet the required separation and finally, organizing the city's UMTS coverage in a hierarchical ways.

Key Words: UMTS; Coverage; QoE and QoS

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This work is fully dedicated to my mother Ms. Barite Gelgelo.

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Acronyms and Abbreviations

1G	1st Generation
2G	2nd Generations
3G	3rd Generations
3GPP	Third Generation Partnership Project
4G	4th Generations AMPS Advanced Mobile Phone Service
AF	Assured Forwarding
ALCAP	Access Link Control Application Part
AMPS	Advanced Mobile Phone Service
AMR	Adaptive Multi-Rate
ATM	Asynchronous Transfer Mode
AuC	Authentication Center
BCH	Broadcast Channel
BLER	Block Error Rate
BS	Base Station
BSC	Base Station Controller
BTS	Base Transceiver Station
CAC	Call Admission Control
CDMA	Code Division Multiple Access
CE	Channel Element
CN	Core Network
CPICH RSCP	Common Pilot Channel Received Signal Code Power
CS	Circuit Switch
D-AMPS	Digital-Advanced Mobile Phone Service

DCCH	Dedicated Control Channels
EDGE	Enhanced Data Rates for Global Evolution
EF	Expedited Forwarding
EIR	Equipment Identity Register
EMS	Element Management System
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FER	Frame Erasure Rates
FPLMTS	Future Public Land Mobile Telecommunication System
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GIS	Geographic Information System
GMSC	Gateway Mobile Switching Centre
GPRS	General Packet Radio Service
GPS	Global Positioning System
GSM	Global System for Mobile communication
GUI	graphical user interface
HLR	Home Location Register
HSDPA	High Speed Downlink Packet Access
HSUPA	High Speed Uplink Packet Access
HTTP	Hypertext Markup Language
IP	Internet Protocol
IRAT HO	Inter Radio Access Technology Handover
IS-136	Interim Standard 136
IS-95	Interim Standard 95
ISDN	Integrated Services Digital Network

LTE	Long Term Evolution
MAC	Medium Access Control
MSC	Mobile Switching Centre
MSE	Mobile Switching Equipment
MT	Mobile Terminating
MTS	Mobile Telephone Service
NBAP	Node B Application Part
NE	Network Element
NMS	Network Management Subsystem
OSS	Operations Support Systems
OVSF	Orthogonal Variable Spreading Factor
PCCH	Paging Control Channel
PCH	Paging Channel
P-CPICH	Primary Common Pilot Channel
PDC	Personal Digital Cellular
PDCP	Packet Data Convergence Protocol
PDN	Packet Data Network
PDP	Packet Data Protocol
PDU	Protocol Data Unit
P-GW	Packet Data Network Gateway
PHS	Personal Handy phone System
PRI	Primary Rate Interface
PRS	Performance Report System
PS	Packet Switch
PSTN	Public Switch Telephone Network
QAM	Quadrature Amplitude Modulation

QCI	QoS Class Identifier
QoE	Quality of Experience
QPSK	Quadrature phase shift Keying
RAB	Radio Access Bearer
RACH	Random Access Channel
RAN	Radio Access Network
RANAP	Radio Access Network Application Part
RB	Radio Bearer
RL	Radio Link
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RNSAP	Radio Network Subsystem Application Part
RRC	Radio Resource Control
RSVP	Resource Reservation Protocol
SCTP	Stream Control Transmission Protocol
SDF	Service Data Flow
SDU	Service Data Unit
SGSN	Serving GPRS Support Node
S-GW	Serving Gateway
SIM	Subscriber Identity Module
SLA	Service Level Agreement
SNR	Signal to Noise Ratio
SRB	Signaling Radio Bearer
TACS	Total Access Communication System
TCP	Transmission Control Protocol
TDD	Time Division Duplex

TD-SCDMA	Time Division Synchronize CDMA
TFCS	Traffic of Circuit Switch
TFP	Traffic Forwarding Policy
TNL	Transport Network Layer
TPC	Transmission Power Control
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
UTRA	Universal Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VPN	Virtual Private Network
WCDMA	Wideband Code Division Multiple Access
WFQ	Weighted Fair Queuing
WIMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network

Chapter 1

1. Introduction

Telecommunication has been in existence for ages and has undergone numerous changes in the recent years, specially in the development of mobile communication systems. Starting from the first generation of mobile system, which is analog communication, to the ones that are being developed now, like 4G and 5G, the technology is expanding in higher quality and accessibility. Besides, end users' expectations have grown from conventional mobile voice and simple text communication to even live streaming services and internet access, which greatly affecting the traffic in a network [1, 4]. These demands motivate the need for new emerging mobile communication system architectures and managements with issues related to quality of service, capacity and coverage. For this reason, the 3rd Generation Partnership Project (3GPP), which is currently the dominant specifications development group for mobile communication systems worldwide, is started. The Universal Mobile Telecommunication System (UMTS) is the 3rd Generation (3G) mobile communication system, which is developed by the 3GPP. 3G network architecture consists of three domains; namely the user equipment domain, the access network domain and the core network domain [1, 3, 4].

The main idea behind 3G is to prepare a universal infrastructure that is able to support both existing and future services. It aims at meeting the future demand for mobile user capacity, providing mobile data, multimedia communication services and also providing global roaming [3, 4].

For the consumers it provides video streaming, television (TV) broadcast, video calls, video clip news, music, sports, enhanced gaming, chat, location services, etc.

And for the business it provides high speed teleporting access, sales force automation, video conferencing and real-time financial information. It also has greater capacity with higher efficiency than first and second generation systems [3, 4].

The real time applications such as voice, video, voice conferencing and video conferencing are highly delay and loss sensitive. These applications require high data rate and high band width to guarantee QoS to the end users. QoS is the capability of the mobile communication systems' service providers to provide a satisfactory service, which includes voice quality, video quality, signal strength, low call blocking and dropping probability, high data rates for multimedia and data applications, etc. It determines how satisfied the users are with the services provided by the telecom operator. Also it refers to the ability of the network to deliver predictable and guaranteed performance for the applications that are running over the network. Implementation of QoS manager in the UMTS network has a great role in order to monitor the quality of the real and non-real time applications.

Providing the required end-to-end QoS for mobile communication systems is one of the challenges for service provider [1, 3, 4].

1.1 Statement of the Problem

In the ethio telecom's Addis Ababa city network, the UMTS system is one of the features of mobile communication system which is able to give quality services for end users. The current UMTS network infrastructure deployed in the Addis Ababa city, which is solely managed by ethio telecom, is undergoing major expansions in the last 4 years and resulted in a tangible improvement of coverage and quality performances. However, there are voice complaints from end users from various parts of the city.

To correctly resolve this end users' complaints the evaluation of QoS of voice over UMTS network is done in this research.

The analysis results show that, there is some disparities between the ethio telecom targets and analysis results. This might be because of improper optimization, shortage of resources (i.e. channel element (CE) licenses, power license, etc) and weak coverage.

1.2 Objective of the Thesis

1.2.1 General Objective

The main aim of this thesis is to evaluate the existing QoS of voice in the UMTS network in case of Addis Ababa and recommends possible solutions to improve the QoS of voice in UMTS network.

1.2.2 Specific Objectives

The specific objectives of the thesis are summarized as follows:

- Survey of existing literatures on QoS of voice over UMTS system.
- Collection of control and user plane systems' data.
- Evaluate the QoS of voice in UMTS network.
- Based on analysis results, recommend the possible solutions to improve the QoS of voice.

1.3 Literature Review

The demands for real-time and high quality services over mobile systems is on the rise. Also, providing real-time high quality services on mobile communication system is a big challenges for providers.

So, the evaluation of QoS of voice over UMTS network has got wide research attentions in the last few years. Some of the researches conducted in the area of QoS performance of voice over UMTS network are briefly reviewed below.

The specifications of 3GPP from [1, 2, 3, 4, 5] define the architecture, topology, and services of UMTS network.

Ruijun, et al. discussed about the architecture of QoS in UMTS network and the mapping of QoS in UMTS networks [21]. They showed that adaptation of application level QoS is effective for resource management and reliable voice transmission for improving the overall voice performance in the UMTS network.

Ali, et al. mapped the QoS for voice and video telephony services in UMTS network in order to minimize the interaction of delay and jitter [20]. They showed the effects of queuing delay by considering the weighted fair queuing (WFQ) scheduler.

ChuanLee, et al. defined heterogeneous mobile Resource Reservation Protocol (RSVP) for mobile host in order to meet required QoS service during roaming between UMTS and Wireless Local Area Network (WLAN) [22]. They also analyzed the performance comparison between heterogeneous mobile RSVP and two-tier resource management scheme in their work.

Sari, et al. analyzed the performance of multimedia services and QoS by using HSPDA in UMTS network [23]. They showed that unacknowledged mode is suitable for VOIP and video streaming while acknowledged mode is appropriate for FTP and web services.

Gurijala, et al. categorized the next generation applications in different classes and some essential QoS metrics for each classes [24].

Niyato, et al. investigated some approaches about designing the call admission control schemes in 4G networks [25]. They also defined some major challenges for provisioning the QoS in heterogeneous networks.

Sen, et al. presented a framework on QoS management for 3G wireless networks [17]. They explained a number of unique characteristics of the radio links and showed the required flexibility of resource management techniques for guaranteed QoS over the wireless network. They also proposed a framework for wireless QoS agent.

1.4 Methodologies

This thesis is entirely based on ethio telecom's UMTS network in Addis Ababa city. It serves more than 1,500,000 mobile subscribers and covers almost 95% of Addis Ababa city. The work started with deep study on QoS of voice in UMTS system and then evaluate QoS of voice over UMTS network in Addis Ababa city. Control and user plane systems' data collections of the existing Addis Ababa UMTS network have been done, by using control plane tools (i.e. M2000, PRS and nastar) and user plane tools (i.e. nemo handy, nemo outdodr, GPS, scanner, google earh, MapInfo and actix analyzer). After that, data analyses have been done to identify the performance of QoS of voice over UMTS network. Based on the analysis results, the possible solutions have been proposed.

In general the method is formulated as:

- Real data collected from UMTS network control and user plane systems;
- Collected real data analyzed;
- Analyzed data discussed in different statistical plots and tables;
- Finally recommend the possible solutions to improve the voice QoS.

1.5 Scopes and Limitations

The scopes of this thesis are:

- Evaluate the existing QoS of voice over UMTS network in the city.
- Recommend the possible solutions to improve the QoS of voice in the city.

The limitations of this thesis are:

- Due to the unavailability of some control and user plane systems' data concerning the QoS indicator parameters, some of the parameters used for calculating the network QoS is roughly estimated.
- The other major limitation of this thesis is:
  - Each layers which includes physical layer, media access control (MAC) layer, internet protocol (IP) layer, transmission control protocol (TCP) layer and application layer had been required their own QoS managers, which is not available in UMTS network of Addis Ababa city.

1.6 Contributions

The contributions of this thesis work are:

- The entire UMTS network of Addis Ababa city's QoS of voice is evaluated,
- The possible solutions to improve the QoS of voice in the city are recommended,
- Implementation of QoS managers in each layers of UMTS network to monitor the city's voice QoS is proposed.

1.7 Thesis Layout

The thesis work is organized in such a way that it gives a clear flow and understanding regarding the subject matter. Chapter one presents the introduction, statement of the problems, objective of the thesis, literature review, methodologies, thesis scope and limitation, contribution and thesis layout. Chapter two presents the 3G network introduction, 3G goals, 3G key technologies and 3G architecture and interface protocol. Chapter three presents the introduction of QoS in 3G network, QoS performance indicator parameters and type QoS managers. Chapter four presents the introduction of control and user plane systems. Chapter five presents the control and user plane systems' data collection and analysis and both analysis results are presented with reasonable explanation. Finally, conclusion is given followed by points of recommendation in chapter six.

Chapter 2

2. Background on 3G Mobile Networks

Now a day's mobile communication technology is developing rapidly. Based on the demand of the users, 3G mobile communication system is being able to provide a variety of applications for end users' satisfaction. In this chapter we describe the background and theoretical knowledge of 3G system [3].

2.1 3G Network Introduction

The 3rd Generation mobile communication system (3G) is put on agenda when the 2nd Generation (2G) digital mobile communication market was booming. The 2G mobile communication system has the following disadvantages: limited frequency spectrum resources, low frequency spectrum utilization, and weak support for mobile multimedia services (providing only speech and low-speed data services). Also, the 2G mobile communication system has a low system capacity, hardly meeting the demand for high-speed bandwidth services. Therefore, the 3G mobile communication technology is a natural result in the advancement of the 2G mobile communication [3, 4].

The 3G mobile communication aims at meeting the future demands for mobile user capacity and providing mobile data and multimedia communication services. Initially, 3G mobile communication technologies were developed separately, as various countries and technical organizations continued to develop their own technologies as shown in Figure 2.1.

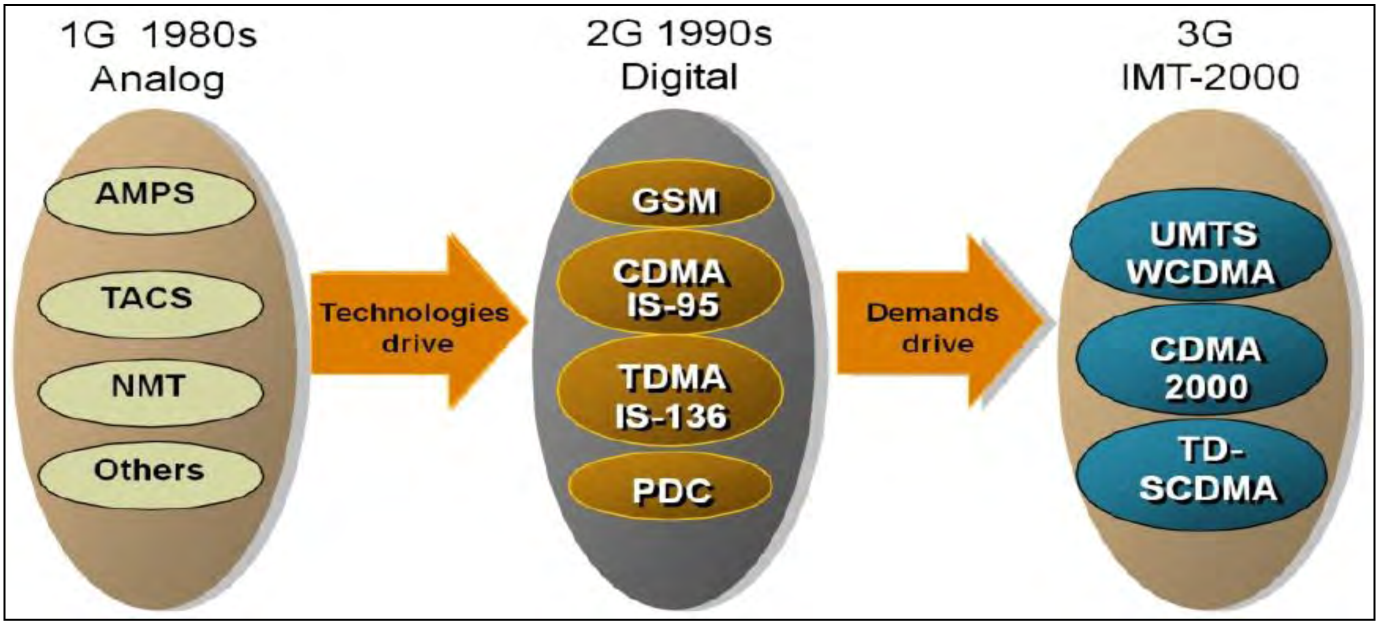


Figure 2.1 Technologies and demands drive services [2].

Thus, the USA has AMPS, D-AMPS, IS-136 and IS-95, Japan has PHS and PDC and the European Union has GSM. On one hand, this situation helped to meet the needs of the users at the early stage of mobile communication and expand the mobile communication market. On another hand, this situation created barriers between the regions and made it necessary to unify the mobile communication systems globally. The 3G mobile communication system, is the general term for the next generation communication system proposed by ITU in 1985, when it was actually referred to as Future Public Land Mobile Telecommunications System (FPLMTS). In 1996, it was officially renamed to IMT-2000. In addition, the 3G mobile communication technology extends the integrated bandwidth network service as far as it can to the mobile environment, transmitting multimedia information including high quality images at rates up to 10 Mbps [1, 2].

As the Internet data services become increasingly popular nowadays, the 3G mobile communication technology opens the door to a brand new mobile communication world.

It brings more fun to the people. In addition to clearer voice services, it allows users to conduct multimedia communications with their personal mobile terminals, for example, Internet browsing, multimedia database access, real-time stock quotes query, videophone, mobile e-commerce, interactive games, wireless personal audio player, video transmission, knowledge acquisition, and entertainments. What more unique are location related services, which allow users to know about their surroundings at anytime anywhere, for example, block map, locations of hotels and super markets, and weather forecast. The 3G mobile communication is bound to become a good assistant to people's life and work [3, 4].

2.2 3G Goals

Compared with the existing 2G system, the 3G system has the following characteristics [3, 4]:

- Support for multimedia services, specially Internet services;
- Easy transition and evolution;
- High frequency spectrum utilization.

Currently, the three typical 3G mobile communication technology standards in the world are CDMA2000, UMTS and TD-SCDMA as shown in Figure 2.2. CDMA2000 and UMTS work in the FDD mode, where the uplink and downlink of the system work in different frequency of the same timeslots, while TD-SCDMA works in the TDD mode, where the uplink and downlink of the system work in different timeslots of the same frequency [1, 2].

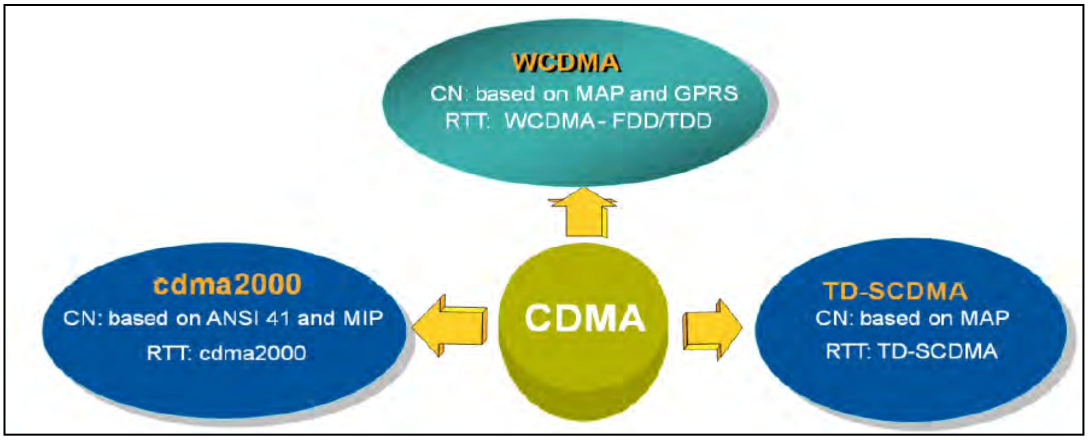


Figure 2.2 Core Technology of 3G.

The 3G mobile communication is designed to provide diversified and high-quality multimedia services. To achieve these purposes, the wireless transmission technology must meet the following requirements [3, 4]:

- High-speed transmission to support multimedia services;
    - Indoor environment: > 2 Mbps,
    - Outdoor walking environment: 384 kbps,
    - Outdoor vehicle moving: 144 kbps.
- Allocation of transmission rates according to needs;
- Accommodation to asymmetrical needs on the uplink and downlink;
- Capable of roaming globally: users can roam within the whole system, even in the whole world, and can be provided with guaranteed quality of service at different rates and in different statuses of motion;
- Providing diversified services: providing voice, data with variable rates, active video services, especially multimedia services;

- Capable of adapting to many kinds of environment: can integrate the existing Public Switched Telephone Network (PSTN), Integrated Service Digital Network (ISDN), cordless system, land mobile communication system and satellite communication system to provide seamless coverage;
- Sufficient system capacity, powerful management capability of multiple users, high security performance and quality of service.

2.3 3G Key Technologies

2.3.1 Code Division Multiple Access

Code division multiple access (CDMA) is a new, while mature wireless technology developed from the spread spectrum communication technology, a branch of the digital technology. It is a multiple access technology featuring high confidentiality. It was first developed in the Second World War to prevent interference from the enemies. It found wide application in anti-interference military communications during the war. After 1960's, it had been used in military satellite communication. Later, it was developed by Qualcomm into a commercial mobile communication technology [26].

CDMA is superior to TDMA in system capacity, anti-interference, communication quality, and confidentiality, so IMT-2000 (3G) launched by ITU and subsequent standards all employ CDMA. With a limited capacity, it has difficulty in offering voice quality equivalent to wired telephone. TDMA terminals support an access rate of only 9.6 kbps. The TDMA system does not support soft handover, so calls may easily be dropped, affecting the service quality. Therefore, TDMA is not the best technology for modern mobile communication system [26].

On the other hand, CDMA fully meets the requirements of modern mobile communication networks for large capacity, high quality, and integrated services, so it is well received by increasingly more operators and users.

CDMA emerges from the needs for wireless communications of higher quality. In the CDMA communication system, the signals used by different users for information transmission are distinguished not by frequencies or timeslots, but by different code sequences.

CDMA allocates one pseudo random binary sequence for each signal for frequency spreading, and different signals are allocated with different pseudo random binary sequences. In the receiver, correlators are used to separate the signals. The correlator of each user only receives the specified binary sequences and compresses their frequency spectrums, while ignoring all the other signals.

The CDMA concept can be illustrated with a party of many persons. At the party, many users talk at the same time in a room, and every conversation in the room is in a language you do not understand. From your perspective, all these conversations sound like noise. If you know these "codes", that is, relevant languages, you can ignore the conversations you do not want to hear, and focus on only the one you are interested in. The CDMA system filters the traffic in a similar way. However, even if you understand all the languages used, you do not necessarily hear clearly all the conversations you are interested in. In this case, you can tell the speakers to speak louder, and/or ask others to lower their voices. This is similar to the power control in the CDMA system. In the frequency domain or time domain, multiple CDMA signals overlap. The receiver can sort out the signals that use the preset code pattern from multiple CDMA signals by using correlators. Other signals using different code patterns are not demodulated, since their code patterns are different from those generated locally at the receiver [26].

2.3.2 Modulation and Demodulation

Modulation is the process to use one signal (know as modulation signal) to control another signal of carrier (known as carrier signal), so that a characteristic parameter of the later changes with the former. At the receiving end, the process to restore the original signal from the modulated signal is called demodulation [26].

During signal modulation, a high-frequency sine signal is often used as the carrier signal. One sine signal involves three parameters: amplitude, frequency and phase.

Modulation of each of these three parameters is respectively called amplitude modulation, frequency modulation, and phase modulation.

In the UMTS system, the modulation is quaternary phase shift keying (QPSK). If high speed downlink packet access (HSDPA) is used, the downlink modulation mode can also be 16QAM, 64QAM, etc. Modulating rate of UMTS uplinks/downlinks are both 3.84 Mega chips per seconds and modulate complex-valued code chip sequence generated by spread spectrum in QPSK mode [26].

2.3.3 Power Control

Quality of service (QoS) that radio cell network provides for each subscriber mainly depends on signal-to-interference ratio (SIR) of subscriber receiving signals. 3G mobile communication system's access technology is code division multiple access (CDMA). For CDMA system, all subscribers in same cell use same band and timeslot, and subscribers are isolated with each other only by the orthogonalization of spreading code. Correlation characteristics between each subscriber signals are not so good and signals of other subscribers interfere signals of current subscribers, due to multipath and delay of the radio channels. Increasing power of other subscribers may enhance interference on current subscriber [26].

Therefore, CDMA system is a strong power restricted system and strength of interference influences system capacity directly. Power control is regarded as one of the key technologies of CDMA system. Power control adjusts transmission power of each subscriber, compensates channel attenuation, countervails near-far effect and maintains all subscribers at lowest standard of normal communication. It reduces interference on other subscribers at most, increases system capacity and prolongs holding time of mobile phones. Power control is an important part in the UMTS system [26].

If all the UEs in a cell transmit signals at the same power, the signals from a near UE to the Node B are stronger, and the signals from a far UE to the Node B are weaker. As a result, the strong signals override the weak signals [26]. This is called near-far effect in the 3G mobile communication system. UMTS is a self-interference system and all users use the same frequency. Therefore, the "near-far effect" is more serious. In addition, for the UMTS system, the downlink of the Node B is power restricted.

To achieve acceptable call quality when the TX power is small, both the Node B and the UE are required to adjust power needed by the transmitter in real time according to the communication distance and link quality. This process is called "power control". Power control of UMTS includes inner loop power control and outer loop power control by effect. Inner loop power control is used to combat channel fade and loss, so that the SIR or power of the received signals can reach the specific target value.

Outer loop power control generates the SIR or power threshold for inner loop power control according to the QoS in the specific environment. By link, there is uplink power control and downlink power control [26].

Since the CDMA system capacity is mainly restricted by that of the uplink, uplink power control is particularly important. By link type, there is open loop power control and closed loop power control. Open loop power control is based on the assumption that the uplink and downlink channels are symmetric. It can counteract path loss and shadow fade. Closed loop power control does not need this assumption, and it can counteract fast fade [26].

- Open Loop power Control
    - Uplink power control

In the UMTS system, every UE is calculating the path loss from the Node B to the UE all the time. When the signal received by the UE from a Node B is very strong, it indicates that either the UE is very close to the Node B or the transmission path is excellent. In this case, the UE can lower the TX power, while the Node B can still receive signals normally. On the other way around, when the signal received by the UE is very weak, its TX power can be increased to counteract the attenuation. Open loop power control occurs only when the UE is powered on and only once [26].

- Downlink power control

It is the process of estimating the initial TX power of a new requested service. The system can estimate the initial TX power of the downlink channel according to the signal quality of the primary common pilot channel (P-CPICH) measured by the UE.

At the same time, the following factors have to be taken into account: QoS, data rate, quality factor Eb/No, real-time total TX power of the downlink, interferences on this cell by other cells, and so on. Open loop power control is simple and direct, without needing exchange of control information between the UE and the Node B. In addition, it features a higher control speed and needs few overheads.

However, in the UMTS system, different frequencies are used in uplink/downlink transmission. The frequency difference is far greater than the coherent bandwidth of channels. Therefore, it cannot be assumed that the fade characteristic of the downlink channel is equal to that of the uplink channel. This is the limitation of the open loop power control [26].

- Inner Loop power Control

In this case, the receiver compares the signal to interference ratio of the received signal with the target value of the control channel. Then, it returns a transmission power control (TPC) command to the sender. The sender determines whether to increase or reduce the TX power based on the closed loop power control algorithm specified by the upper layer, and makes adjustments at the specified step according to the received command [26].

- Outer Loop power Control

Outer loop power control is a supplement to closed loop power control. The working principle of uplink outer loop power control is: Compare the actual block error ratio (BLER) of the transmission channel with the target block error ratio (BLER), then, slowly adjust the target signal-interference ratio (SIR Target) so that the service quality is not affected by the change of the radio environment, and that a relatively constant communication quality can be maintained.

Outer loop power control usually adjusts the target SIR based on BLER, to make the QoS meet the requirements. Since different services have different QoS, there are different target SIRs. Downlink outer loop power control is similar to downlink inner loop power control [26].

2.3.4 Handover

UMTS distributes radio resources for UE in the new cell because of UE's moving and system load. Then UE synchronizes with the new cell and transmits data each other. Handover is a very important technology in mobile communication system networking [26].

Handover falls into:

- Soft handover: refers to adding a new link (the old and the new links exist at the same time) and deleting the old one after stabilization. Services continue in the handover.
    - Soft handover of cells under same Node B (softer handover),
    - Soft handover of cells between different Node Bs,
    - Soft handover of cells in same band between different RNCs (involving Iur interface).
- Hard handover: refers to deleting the old link and then building a new link. Services break off during the handover.
    - Hard handover between different operators,
    - Hard handover in same operator (forced hard handover),
    - Hard handover between systems (such as: with GSM, LTE, etc.),
    - Hard handover between different modes (such as, between FDD and TDD).

Handover refers to the redistribution of radio resource in mobile communication system in cell structure, to keep discontinuous communication of mobile phones when moving a mobile station from a district to another [26].

2.3.5 RAKE Receiver

Since multipath signals contain useful information, CDMA receivers improve the S/N of the received signals by combining multipath signals. What a RAKE receiver does is to receive the various channels of signals from multipath signals through multiple related detectors, and then combine them.

The RAKE receiver is a classical diversity receiver specially designed for the CDMA system. Its theoretical basis is that when the propagation delay exceeds one chip code cycle, multipath signals are actually seen to be mutually irrelevant. RAKE reception separates and combines multipath signals. Different from the IS-95 A, UMTS has three times of multipath resolving power. In addition, in a UMTS system, the pilot information sent by the user can be used for coherent combination on the reverse link.

The theoretical analysis of UMTS shows that if the reverse link uses 8-path RAKE reception, over 75% signal energies are used. The suppression of the RAKE reception for multiple access interference depends on the correlation between the different user characteristic codes [26].

2.3.6 Code Resource Allocation

In UMTS, code resources fall into channelization codes, scrambling codes and synchronization codes. The primary scrambling codes are to differentiate cells, channelization codes are to differ physical channels on the downlink, and scrambling codes are to differentiate users on the uplink. Orthogonal variable spreading factor (OVSF) is precious scarce resource, so one cell corresponds to one code table. To access more users and to increase system capacity, make use of code resources reasonably. It is very important to plan and manage downlink channelization code resources. Although there are many scrambling codes on the uplink, it is necessary to plan scrambling of RNC, to avoid different users in different RNC use same scrambling codes. Synchronization codes fall into primary synchronization codes and secondary synchronization codes, which are the detection objects of cell search by a UE. All cells have same primary synchronization code [26].

2.3.7 Call Admission Control

Call admission control (CAC) is used in the call set up phase and applies to real-time media traffic as opposed to data traffic. CAC mechanisms complement and are distinct from the capabilities of QoS tools to protect voice traffic from the negative effects of other voice traffic and to keep excess voice traffic off the network. Since it averts voice traffic congestion, it is a preventive congestion control procedure [26]. It also ensures that there is enough bandwidth for authorized flows. CAC decides to accept or refuse a new subscriber, new radio access bearer (RAB) and new radio link (RL) according to current resource (such as, handover). CAC is applicable to original UE access, RAB designation, reconfiguration and handover. It may lead to different results because of primary rate interface (PRI) and actual situations. CAC meets QoS of new calls as much as possible premising the stability of the system, based on interference measurement, to avoid overloading. CAC falls into: Uplink call admission control and Downlink call admission control. UMTS is a self-interference system and there exists power climbing. Uplink capacity depends on whether total receiving interference power is beyond linear dynamic range of low noise amplifier or not. Downlink capacity depends on whether the distribution of transmission power completes or not [26]. Process of call admission control: Measure current load of system cell when making calls (new access calls and handover calls), forecast and estimate calls and judge whether to access calls. Admission control threshold of new calls is lower than that of handover calls. New calls are refused when current load of the cell is higher than admission threshold of new calls.

Handover calls are also refused when the load of the cell is higher than handover admission control threshold. Load control threshold is commonly higher than admission control threshold of handover, to prevent overloading when radio circumstances change and to ensure the stable running of the system [26].

2.3.8 Load Control/Congestion Control

The system measures the load of the system cell at real time continually. The system load is high and the load enters unstable running district of the system when load average value exceeds some threshold value in a set time. Load control is necessary at the moment [26]. Load control processes are:

- Downlink rapid load control: Refusing the command to increase the power from the UE by Node B;
- Uplink rapid load control: Reducing SIR destination value for uplink rapid power control by Node B;
- RNC makes judgment and changes maximum allowed transmission power, destination SIR value and TFCS by reconfiguring the RL. In this mode the system load can be reduced for a long time;
- Make negotiations if hoping to reduce the system load for a long time, that is, RNC negotiates with CN to reduce resource occupation of services during the communications;
- Share loads with adjacent cells in RNS to reduce the load of those overloading cells.

The taking-in and sending-out of adjacent cells (covering radius of one cell increases and that of another adjacent cell decreases) is called cell breathing.

2.4 3G Architecture and Interface Protocol

2.4.1 3G Architecture

3G mobile system including the RAN, CN and UE domains. The RAN is used to process all the radio-related functions, while the CN is used to process all voice calls and data connections within the UMTS system, and implements the function of external network switching and routing. Logically, the CN is divided into the CS (circuit switched) domain and the PS (packet switched) domain. UTRAN, CN and UE together constitute the whole UMTS system. A RNS is composed of one RNC and one or several Node Bs [27, 28].

The Iu interface is used between RNC and CN while the Iub interface is adopted between RNC and Node B. Within UTRAN, RNCs connect with one another through the Iur interface. The Iur interface can connect RNCs via the direct physical connections among them or connect them through the transport network [27, 28].

RNC is used to allocate and control the radio resources of the connected or related Node B. However, Node B serves to convert the data flows between the Iub interface and the Uu interface, and at the same time, it also participates in part of radio resource management.

Figure 2.3 shows the overall architecture of UMTS UTRAN (UMTS terrestrial radio access network) system.

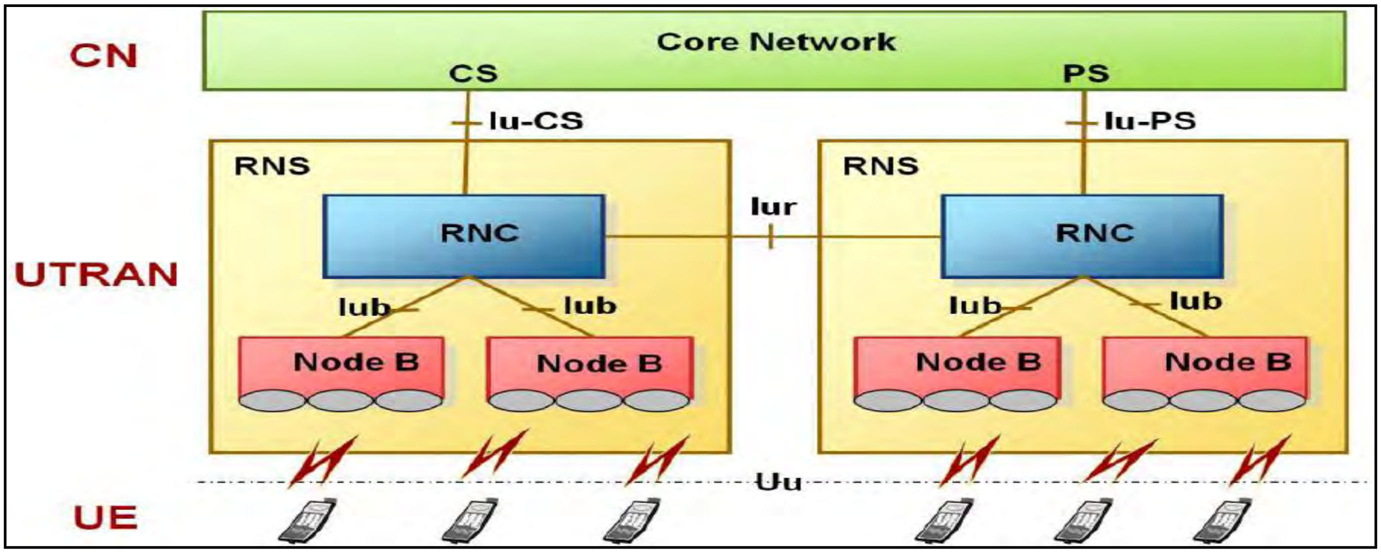


Figure 2.3 WCDMA system architecture [28].

❖ Radio network controller (RNC)

Radio network controller is the control unit of UTRAN, it performs various tasks and controls all radio resources within RNC and it is same as BSC in GSM network.

Most of the protocols between RAN and UE are implemented in the RNC, it communicates over Iu interface with a maximum of one fixed network nodes SGSN and MSC.

Each RNC is allocated between the SGSN and MSC. It also has an option of using Iur-interface to communicate over core network (CN) with neighboring RNC's [29].

RNC is responsible for call admission control, radio resource management, data transmission, radio bearer setup and release, code allocation, power control, packet scheduling, RNS allocation to handover control, encryption, protocol conversion and ATM switching. More details about the functionalities and the responsibilities of RNC are in [29].

- Node B

Node-B is similar to the base transceiver system (BTS) in GSM network. As like as BTS, Node-B has the capability for both transmission and reception.

It handles and provides radio coverage to the cells. Node-B is connected with RNC and user equipment (UE) via Iub and Uu interfaces respectively [29]. Both Node B and user equipment (UE) verify the status of the connection and level of interference and send the update to connected RNC. It is responsible for allocating the channel and radio resources in order to interact with user equipment. It also performs error detection and correction, modulation and handover management [29].

- Core network

Core network is the control unit of UMTS network. This unit controls all types of operations in the network. Core network is responsible for establishing, maintaining and terminating the connection with the user. The Core network interacts with external networks for different types of communication such as data, voice and video etc. The major difference between GSM and UMTS network is the radio interfaces. Other responsibilities of UMTS core network are authentication, location updating and registration.

A detail about the functionalities of core network is in [29]. The UMTS core network is combination of two types of switching such as Packet Switching (PS) and Circuit Switching (CS). Packet switching elements are responsible for packet switching and circuit switching elements are responsible for circuit switching.

- Circuit switched (CS) elements of core network

The circuit switched elements is responsible for performing circuit switching.

The core network consists of the following circuit switching elements: the mobile-services switching center (MSC), the gateway MSC (GMSC), the home location register (HLR), the visitor location register (VLR), the equipment identity register (EIR) and the authentication center (AuC) [29].

❖ Packet switched (PS) elements of core network

The PS elements of core network provide support for packet routing. This unit is connected with packet switched network such as Internet.

Two additional nodes are required to provide support for packet switching; these nodes are called serving GPRS support node (SGSN) and gateway GPRS node (GGSN). Service GPRS support node (SGSN) is one of the important elements to support the packet switched network. It is responsible for routing the incoming and outgoing packet data from and to GPRS user via radio access network.

It is also responsible for user authentication, user location update, data encryption and decryption, establishment, maintaining, terminating sessions and mobility management procedure for UMTS users [29]. Gateway GPRS support node (GGSN) is the main component to support the packet switched network. It acts as a gateway of external packet data network (PDN) such as Internet.

This component is connected with external IP network to route the IP packets. The protocol which is used in PDN is called Packet Data Protocol (PDP). GGSN converts the packet data into PDP format to send the external IP networks. The GGSN is responsible dynamically generating IP addresses [29].

❖ User equipment (UE)

The UE contains various mobile terminals and electronic smart card that is removable as well as portable.

UMTS subscriber identity module (USIM) is an electronic smart card that contains the information about the user to identify in the network. To enable the user in the network the SIM should be connected in the ME. Mobile equipment (ME) is the terminal equipment that the subscriber can access the network by using this equipment. It is capable both for transmission and reception. It is a sort of transceiver [29].

❖ UMTS network interfaces.

- lub: Responsible for connection between RNC and node-B.
- lur: This interface is used to connect two RNCs.
- Uu: UTRAN and UE are connected through this interface.
- Iu: This inter face help to establish link between RNC and 3G core network.
- Iu-CS: Circuit switched domain connects with the RNC using this interface.
- Iu-PS: Packet switched domain connects with the RNC through this interface.

2.4.2 UTRAN Common Protocol Model

Figure 2.4 shows the general protocol model for UTRAN Interfaces. The structure is based on the principle that the layers and planes are logically independent of each other.

Therefore, as and when required, the standardization body can easily change protocol stacks and planes to fit future requirements [28].

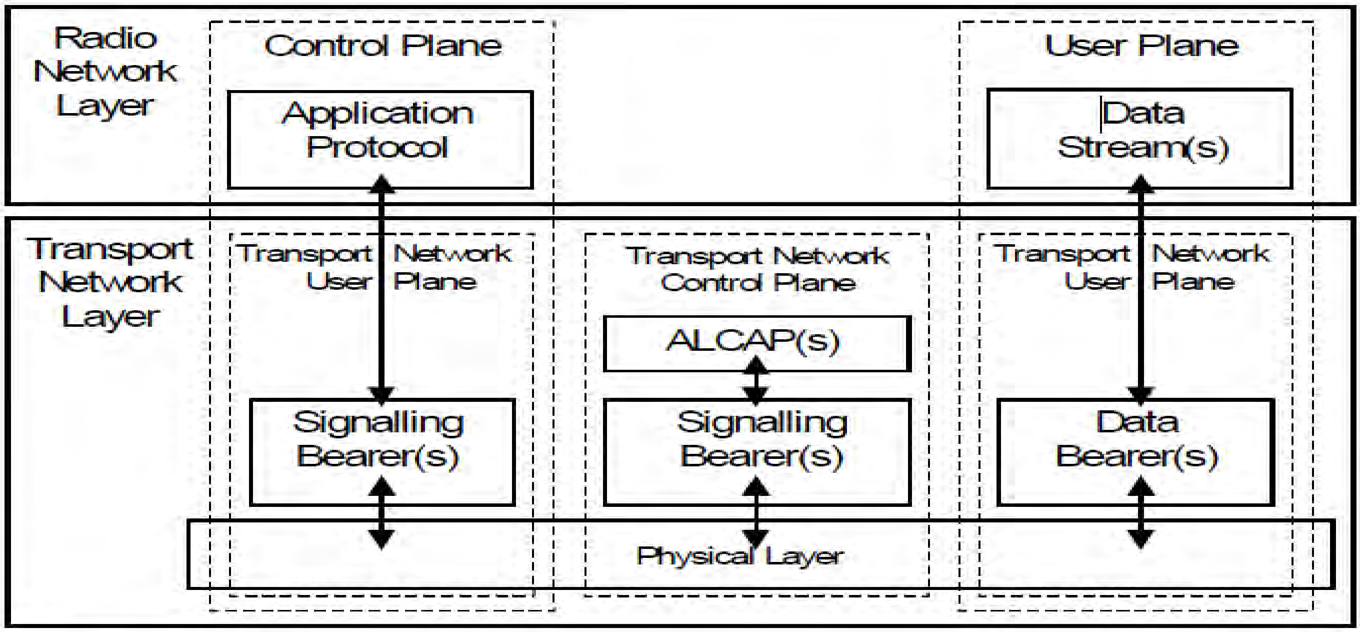


Figure 2.4 UTRAN Common Protocol Model [28].

In horizontal, the protocol structure consists of two main layers, radio network layer, and transport network layer. All UTRAN related issues are visible only in the radio network layer and the transport network layer represents standard transport technology that is selected to be used for UTRAN, but without any UTRAN specific requirements.

In vertical, UTRAN falls into the following 4 planes: control plane, user plane, transport network control plane and transport network user plane.

The control plane includes the application protocol, i.e., RANAP, RNSAP or NBAP, and the signaling bearer for transporting the application protocol messages. Among other things, the application protocol is used for setting up bearers for (i.e., radio access bearer or radio link) in the radio network layer.

In the three plane structure the bearer parameters in the application protocol are not directly tied to the User Plane technology, but are rather general bearer parameters.

The signaling bearer for the application protocol may or may not be of the same type as the signaling protocol for the ALCAP. The signaling bearer is always set up by O&M actions.

The user plane includes the data stream(s) and the data bearer(s) for the data stream(s). The data stream(s) is/are characterized by one or more frame protocols specified for that interface.

The transport network control plane does not include any radio network layer information, and is completely in the transport layer. It includes the ALCAP protocol(s) that is/are needed to set up the transport bearers (data bearer) for the user plane. It also includes the appropriate signaling bearer(s) needed for the ALCAP protocol(s).

The transport network control plane is a plane that acts between the control plane and the user plane. The introduction of transport network control plane makes it possible for the application protocol in the radio network control plane to be completely independent of the technology selected for data bearer in the user plane.

When transport network control plane is used, the transport bearers for the data bearer in the user plane are set up in the following fashion. First there is a signaling transaction by the application protocol in the control plane, which triggers the set up of the data bearer by the ALCAP protocol that is specific for the user plane technology.

The independence of control plane and user plane assumes that ALCAP signaling transaction takes place. It should be noted that ALCAP might not be used for all types data bearers. If there is no ALCAP signaling transaction, the transport network control plane is not needed at all. This is the case when pre-configured data bearers are used.

It should also be noted that the ALCAP protocol(s) in the transport network control plane is/are not used for setting up the signaling bearer for the application protocol or for the ALCAP during real time operation.

The signaling bearer for the ALCAP may or may not be of the same type as the signaling bearer for the application protocol. The signaling bearer for ALCAP is always set up by O&M actions.

The data bearer(s) in the user plane, and the signaling bearer(s) for application protocol, belong also to transport network user plane.

The data bearers in transport network user plane are directly controlled by transport network control plane during real time operation, but the control actions required for setting up the signaling bearer(s) for application protocol are considered O&M actions [28].

Chapter 3

3. Quality of Service in 3G Network

QoS in 3G mobile communication system is defined as the capability of the mobile communication system service providers to provide a satisfactory service which includes voice quality, video quality, signal strength, low call blocking and dropping probability, high data rates for multimedia and data applications etc. QoS is a comprehensive reflection of the service capability of a 3G mobile communication system. It determines how satisfied the users are with the services provided by the telecom operator. Therefore, it is an important factor to be considered in the 3G mobile communication system. The QoS of a wireless network is affected by attenuation, multi-path interference, spectrum interference, noise, mobility and limited capacity. To ensure the end-to-end QoS, all the nodes from the transmitter to the receiver need to cooperate with each other [5, 6, 7]. In this chapter we describe the QoS in 3G network, QoS performance indicator parameters and QoS managers.

3.1 Quality of Service and Quality of Experience

QoS is the ability of the network to provide a service at an assured service level. In order to provide the best Quality of Experience (QoE) to users in a cost-effective, competitive and efficient manner, network and service providers must manage network QoS and service provisioning efficiently and effectively. QoE is the term used to describe user perceptions of the performance of a service. QoS and QoE are so interdependent, that they have to be studied and managed with a common understanding, Figure 3.1.

The aim of the network and services should be to achieve the maximum user rating (QoE), while network QoS is the main building block for reaching that goal effectively.

QoE, however, is not just limited to the technical performance of the network, there are also non-technical aspects, which influence the overall user perception [8, 12].

- QoS encompasses all functions, mechanisms and procedures in the mobile communication network and terminal that ensure the provision of the negotiated service quality between the UE and the core network (CN).

- QoE is how a user perceives the usability of a service when in use – how satisfied they are with a service in terms of, for example, usability, accessibility, retainability and integrity of the service. Service accessibility relates to unavailability, security, activation, access, coverage, blocking, and setup time of the related bearer service; service retainability, in general, characterizes connection losses.

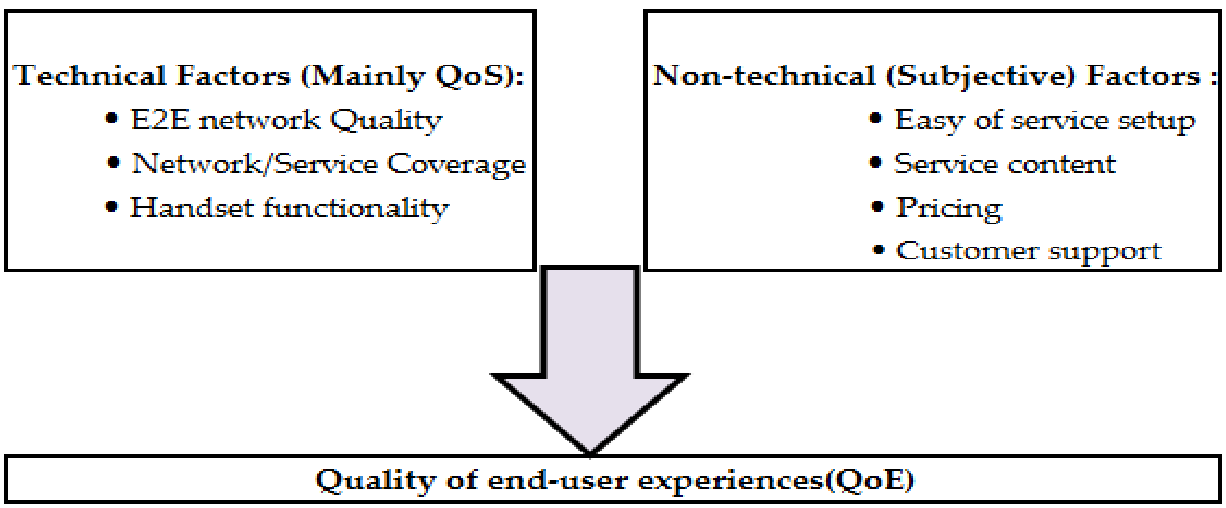


Figure 3.1 QoE is affected by technical (QoS) and non-technical aspects of service [12].

QoE and QoS concepts are shown in Figure 3.2, QoE is expressed in "feelings" rather than metrics. QoS relates to all mechanisms, functions and procedures in the network and terminal that implement the quality attributes (bearer service) negotiated between the UE and the CN.

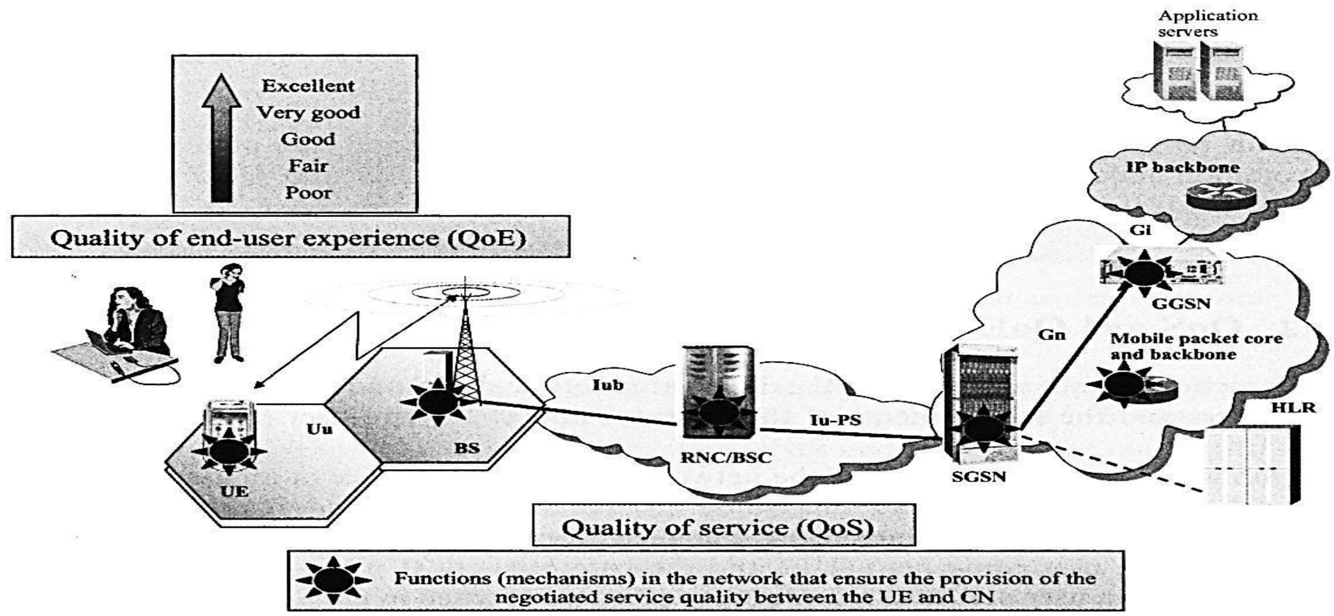


Figure 3.2 QoE and QoS concepts [12].

QoE is a function of multiple protocol layers and network elements. Radio interface is the bottleneck, with its restrictions in bandwidth and coverage.

3.2 3G QoS Architecture

3GPP TS 23.107 describes the QoS concept and architecture. In addition, it provides QoS parameters based on the UMTS bearer service. To ensure the QoS of a network, bearer services with explicit attributes and functions must be set up between the transmitter and the receiver. A bearer service involves all the aspects that are required to ensure specific QoS.

These aspects are included in control plane signaling, user plane transmission and QoS management. Figure 3.3 shows the UMTS QoS architecture. As shown in Figure 3.3, the traffic from one terminal equipment (TE) to another passes different levels of bearer services. The TE is connected to the UMTS network through a mobile terminal (MT).

The end-to-end service at the application layer is implemented through the bearer services of the underlying networks [12].

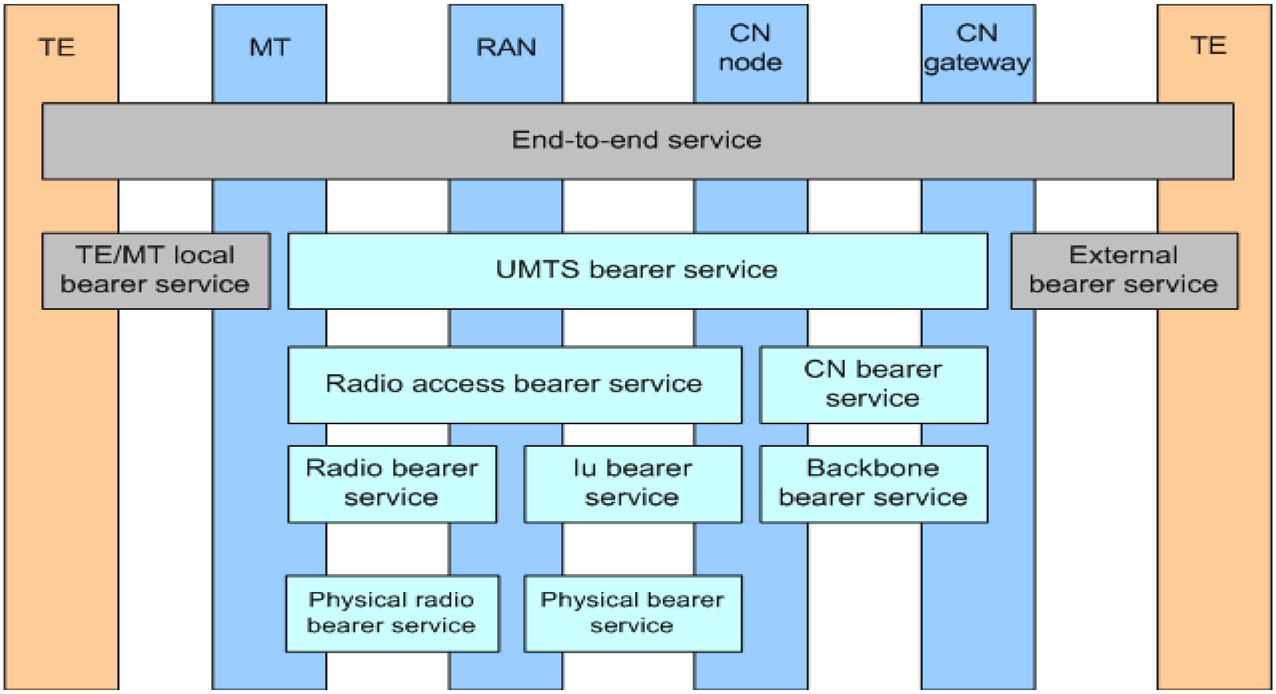


Figure 3.3 UMTS QoS architecture [12].

The end-to-end service consists of the local bearer service, UMTS bearer service, and external bearer service. These services ensure the QoS of the end-to-end service. They are described as follows [12]:

- The external bearer service is coordinated by the telecom operator with the connected networks. Between the UMTS bearer service and the external bearer service, QoS mapping is required. Through the QoS mapping, the QoS requirement is sent to the next network element (NE).
- The coordination and QoS mapping between UMTS bearer services is very important for implementing the end-to-end QoS of UMTS.

The UMTS bearer service consists of the radio access bearer (RAB) service and the core network bearer (CNB) service.

The RAB service is implemented through the radio bearer (RB) service and Iu bearer service. The RB service covers all the aspects of the transmission on the radio interface, and the Iu bearer service provides the transmission between the UTRAN and the CN. For PS services, the Iu bearer service can provide different QoS classes.

The role of the core network bearer (CNB) service is to provide a negotiated UMTS bearer service. The CN provides different QoS classes for different backbone bearer services. A specific backbone bearer service can be selected to meet the QoS requirement of the CN bearer service. The RAB service involves the Uu, Iub, Iur, and Iu interfaces.

3.3 UMTS QoS Classes

3GPP TS 23.107 defines the following four UMTS traffic classes: conversational, streaming, interactive and background. Conversational class is the class that involves real-time communication e.g., audio, video calls. Generally the requirements for speech are low delay, low jitter and clarity with no reverberation. Failure to meet the basic requirement can result in lack of quality and could be unacceptable. Subjective evaluations have shown that the end-to-end delay has to be less than 400 ms for video and audio conversation. Video application is another application that will use conversational class for UMTS transport [6].

Interactive class is mostly used for client-server application for remote service. It allows smooth interaction of humans and machines with remote devices. Buffering is allowed in this class, there is no guaranteed bit rate because of using best effort service model while the delay factor is also kept at the minimum.

The examples of applications using this class are web browsing, server access and database retrieval. In this type of service, the users request to the server for information and the server responses [6].

Streaming class is used for streaming multimedia such as high quality audio and video streaming. Streaming class is little delay tolerance. The supported services of this class are video downloading, high quality video and audio streaming, still image transferring and watching online TV etc [6].

Background class is used for best-effort service in order to download emails and files. This class is highly delay tolerance and the traffic of this class has lowest precedence. Buffering is allowed in this class and high variable delay acceptable. Bit rate does not provide any guarantee because of using best effort service model [6]. Table 3.1: End-to-end audio performance exceptions.

Medium	Application	Degree of symmetry	Data Rate (Kbps)	Key performance parameters and target			
				End-to-end one way delay (ms)	Delay variation within a call (ms)	Information Loss	MOS
Audio	Conversational voice	Two-way	4-25	< 150, preferred < 400 limit.	< 1	< 3% FER	>3

Table 3.1 End-to-end performance expectation [6].

The most of QoS parameters are associated to application layer, network layer or physical layer and the QoS parameters for all the layers are not the same & the mechanism for different layers are also different.

The relationship of QoS parameters in terms of voice transmission is given in the Table 3.2 .

Layer / Service	Application Layer	Network Layer	Physical Layer
Voice Transmission	Call rates	Jitter	Bite error rate
	Jitter	Call handover delay	Signal to noise ratio(Ec/Io)
	Quality of Speech	Call transfer delay	Noise figure
	Accessibility in Service	Accessibility in network	Path loss
	Call Start-up delay	Error rate and loss rate	Received signal(RSCP) strength
	Response time	Traffic handling priority	Channel capacity
	Codec delay	Network allocation	

Table 3.2 QoS parameters association with layers [30, 31, 32, 33].

3.4 QoS Performance Indicator Parameters

QoS indicator parameters are widely used by UMTS systems with the aim of evaluating the QoS delivered to end-users; which can be given with a planed structure: 1) the first plane represents network availability as the QoS from the network's point of view; 2) the second plane represents network access as the basic requirement from the user's point of view. In our case to evaluate QoS of voice transmission over UMTS network, we consider the following parameters [24]:

- Network geographic observation parameters;
- Network coverage and quality analysis parameters;
- Network Performance analysis parameters;
- Drive test service quality analysis parameters.

3.4.1 Network Geographic Observation Parameters

Network geographic observation uses radio link measurement reports (MRs) of subscribers on the live network to locate exceptions. This function combines data based on the call times in call history records (CHRs) and MRs to achieve location positioning and then displays the distribution of coverage, traffic, and exceptions of wireless networks on the geographic information system (GIS) at the grid level. This helps network optimization engineers evaluate network performance, identify hot spots, and quickly locate problematic areas in a straightforward way [13].

It replaces traditional drive tests (DTs) and addresses the problem that DTs fail to cover all areas, thereby providing telecom operators with cost-effective network evaluation and problem identification means. And also it replaces traditional simulation means to display the coverage of each serving cell on the live network, helping telecom operators identify problematic areas such as those with a poor coverage. Network coverage geographic observation supports the geographic rendering for CPICH Ec/No, CPICH RSCP, soft handover area, pilot pollution area, LAC stability, primary serving cell, etc [13].

3.4.2 Network Coverage and Quality Analysis Parameters

The Nastar analyzes the measurement reports (MRs) sent by UEs to display the coverage, quality, and subscriber distribution of the test cell. The analysis result helps users to determine whether problems such as weak cell coverage, cross coverage, and poor service quality occur on the live network. An MR reported by a UE contains downlink received signal code power (RSCP), Ec/No, transmit propagation delay (TP i.e., 1TP = 234 meters), DL BLER, and UE TX POWER [13].

3.4.3 Network Performance Analysis Parameters

The Nastar provides the UMTS cell performance analysis function to help quickly identify abnormal cells and obtain data of abnormal calls in these cells.

The obtained data helps network optimization engineers detect the causes of abnormal cells, facilitating in-depth problem analysis. The Nastar locates and analyzes network problems by monitoring performance counters, and then analyzes the counters of the entire network or for a specific RNC. These counters are: RAB setup failure, abnormal release, QoS problem, delay problem, short call problem, etc [13].

The performance surveillance (PRS) is a platform for analyzing performance data of mobile networks, customizing reports, and displaying reports (i.e., Packet loss, Voice quality indicator, KPIs, Counters, etc). The PRS is applicable to routine maintenance of mobile network. It can monitor and analyze the performance of the entire network [14].

3.4.4 Drive Test Service Quality Analysis Parameters

Physical channels are used to transfer data across the air interface. There are two categories of physical channel; common and dedicate channels, i.e. the same categories as used for transport channels. Common channels can be used by more than a single UE whereas dedicated channels can be used by only a single UE. Many of the measurement reporting events specified within 3GPP TS25.331 are based upon CPICH (common pilot channel) measurements. The two most important measurements obtained from the CPICH are CPICH received signal code power (RSCP) and CPICH received energy per chip per interference level (Ec/Io) [34, 35, 36].

RSCP service quality: In the UMTS cellular communication system RSCP is the power value after the process of correlation/separation (dBm) and should be measured for each code. It denotes the power measured by a receiver on a physical channel communications in particular.

It is used as an indication of signal strength, as a handover criterion, in downlink power control, and to calculate path loss. And can only be measured once the receiver has found the dominant pilot [34]. It is calculated by:

$$\boldsymbol{RSCP(dBm) = RSSI(dBm) + E_c/I_0\ (dB)} \qquad (1)$$

For each Node B, there is a threshold point below which connection break with active Node B. Therefore the signal strength must be greater than threshold point to maintain the connection with active Node B. The signal gets weaker as mobile moves far away from active Node B and gets stronger signal towards new Node B as it move closer as shown in Figure 3.4.

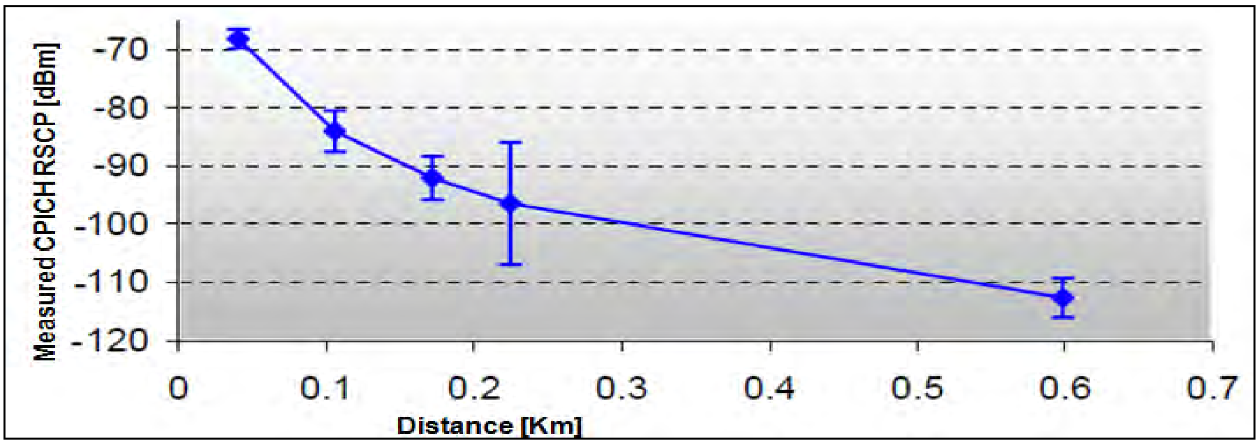


Figure 3.4 CPICH RSCP Vs distance graph for 111008_BOLE MICHAEL_U11.

Table 3.3 shows the classification of coverage based on CPICH RSCP level

RSCP (dBm)	**Grade**	**Some Explanation**	**User Satisfaction**
-115<X<-95	Poor	a higher interference area	Many dissatisfied
-95≤X≤-85	Fair	an interference area	Some dissatisfied
-85<X≤-75	Good	a good signal coverage level	Satisfied
- 75 <X≤-65	Very good	very good signal coverage level	Very satisfied
-65<X≤Max	Excellent	strongest signal coverage level	Remarkably satisfied

Table 3.3 Classification of coverage based on CPICH RSCP level [34].

The received signal strength indicator (RSSI) is a value that takes into account both RSCP and Ec/Io. It is used to measure the power between the received radio signals. It should be noted that the RSSI is never considered in a UMTS system as an indication of coverage. As with RSCP and Ec/Io, it can only be measured in the code domain and needs the special monitoring equipment [36]. It is calculated by:

$$\boldsymbol{RSSI(dBm) = RSCP(dBm) - \mathbf{E_c/I_0}\ (dB)} \qquad (\mathbf{2})$$

Ec/Io and Ec/No service quality: In UMTS, Ec/No and Ec/Io are often used interchangeably. Ec/Io is the ratio of the energy per chip in CPICH to the total received power density (including CPICH itself), usually in dB. Interference is typically measured by the energy per chip to total received power (Ec/Io) of the CPICH, in other words, how clear is the signal received. In a UMTS network the UE normally receives signals from multiple Node Bs, all transmitting on the same frequency. Therefore it is possible that even at a location close to a Node B, with a high RSCP, no logon is possible, due to high interference levels from a second nearby Node B. This effect is called “pilot pollution” and network planners try to avoid too close spacing of base stations to minimize regions where it can occur [34].

The CPICH Ec/Io can be calculated by:

$$\boldsymbol{\mathbf{E_c/I_0}\ (dB) = RSCP(dBm) - RSSI(dBm)} \qquad (\mathbf{3})$$

Figure 3.5 show the plot of Ec/lo at various distance from the Node B. The trend line fitted indicates that the QoS decreases with increase in the distance of the UE from the Node B as expected. This is expected but the implication of this is that data rate decreases, network logging becomes impossible, calls cannot be initiated, network signal disappears, and calls are dropped. This could have been caused by the effect of missing neighbors, low 3G network coverage or no 3G network coverage at all.

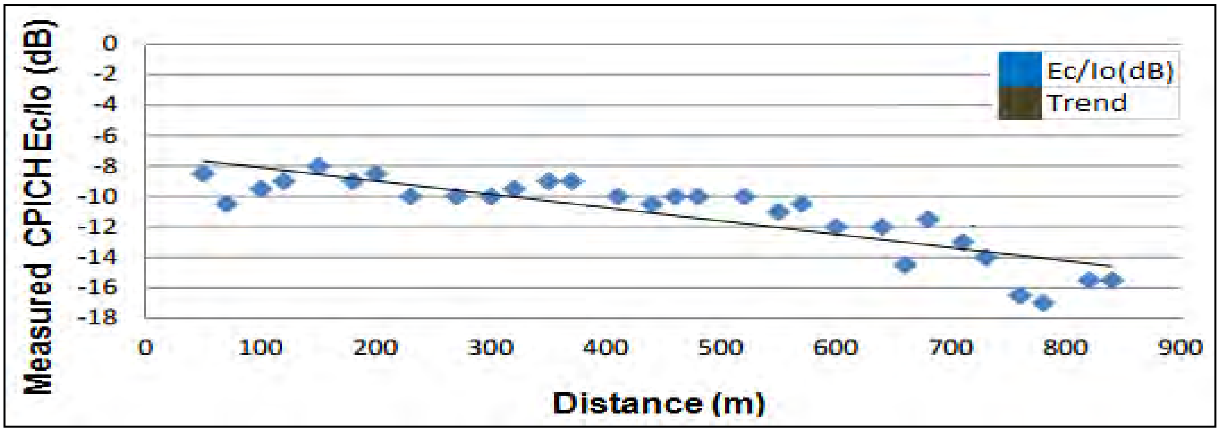


Figure 3.5 CPICH Ec/Io Vs distance graph for 111008_BOLE MICHAEL_U11.

Table 3.4 shows the classification of quality based on CPICH Ec/Io level

Ec/No (dB)	**Grade**	**Some Explanation**	**User Satisfaction**
-24<X<-12	Poor	a higher interference area	Many dissatisfied
-12≤X≤-9	Fair	an interference area	Some dissatisfied
-9<X≤-7	Good	a good signal quality level	Satisfied
- 7<X≤-5	Very good	very good signal quality level	Very satisfied
-5<X ≤Max	Excellent	strongest signal quality level	Remarkably satisfied

Table 3.4 Classification of quality based on CPICH Ec/Io level [36].

The relation between RSPC and Ec/No is mainly impacted by the loading of the system and the quality of the network plan. Figure 3.6 shows the high range of Ec/No for a given RSCP value. It should be noted that the quality of the network plan would be reflected by the number of cells detected at a given location, the cell overlap: a high quality network plan would be one where a single cell is detected over the majority of the cell area and transition between cells are done over clear boundaries.

When the loading of the system increases the Ec/Io degrades but the RSCP stays constant. Degrading Ec/Io is an indication of increased other cell interference which will also increase the need for downlink traffic power.

Power being a limited resource, the higher required transmit power may not be available, thus the coverage not being met in loaded condition: this represent the coverage and capacity trade-off for the downlink in a UMTS systems. In a similar way, adding sites to provide deeper coverage indoor without controlling the footprint of each of them will increase other cells interference and impact service quality and capacity of the system.

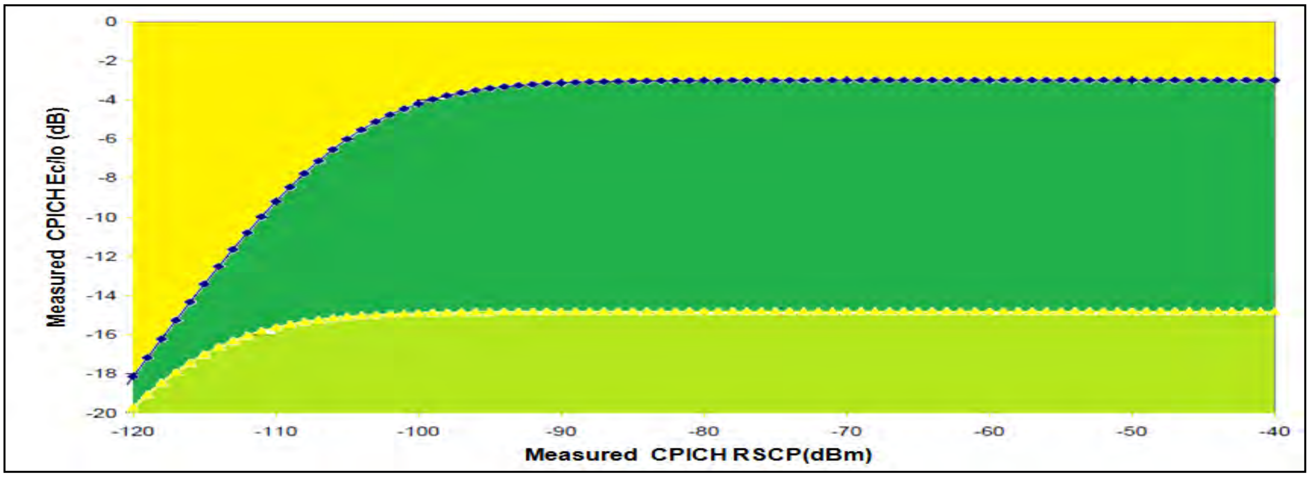


Figure 3.6 CPICH Ec/Io Vs RSCP graph for 111008_BOLE MICHAEL_U11.

Mean opinion score is used to provide a numerical measure of the quality of human speech at the destination end of the circuit. This method of voice quality test has been in used for decades to obtain human user's view of the quality of the network [37].

The values of MOS are rated from 1-5 as shown in Table 3.5.

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

Table 3.5 Subject MOS [37].

3.5 Types of QoS Manager

In heterogeneous network different types of applications are available from different type of networks. Different traffic flows are available from different networks. In order to manage the various traffic flows appropriate QoS manager is required for proper resource management. It can allocate the resources dynamically for appropriate application. It is able to manage the network resources for different application that comes from other network as well as for its own network application. QoS manager plays an important role in different types of handovers. QoS guaranty is required due to mobility of the users. QoS manager reduce packet losses and delay by provisioning the appropriate resources during handover. Every network/domain contains QoS manger which is defined as domain QoS (DQoS) manager. IP core network also hold a QoS manager which interact and share knowledge with domain QoS manager [12, 20].

Integrated service, Differentiated service and Multiprotocol label switching (MPLS) are the three common types of QoS managers.

Integrated service is responsible to provide the service for end user. It is suitable for simple network. It works per flow basis. The major responsibility of integrated service is resource reservation and call setup. It is used to occupy the network resources for the users.

The resource with highest priority is served first, the lowest priority gets low opportunity and the similar priorities are provisioning into queue. It has different kind of services such as best effort service, controlled load service and guaranteed service. But the main drawback is that it is not scalable to work in large network [22].

Differential service works in core network. It is scalable and suitable for large network. Differential service does not deal with per flow basis; on the contrary it evaluates the service based on the class. The flows that are received get treatment by class.

It is more efficient, scalable in order to provide service in large network. In differential service network differentiated service field is used instead of type of service field (ToS). In differential service field there are two types of forwarding such as expedited forwarding (EF) and assured forwarding (AF) [22].

Multiprotocol label switching is a protocol for speeding up and shaping network traffic flows. The fundamental concept behind MPLS is that of labeling packets. In a traditional routed IP network, each router makes an independent forwarding decision for each packet based solely on the packet's network-layer header. Thus, every time a packet arrives at a router, the router has to "think through" where to send the packet next. With MPLS, the first time the packet enters a network, it's assigned to a specific forwarding equivalence class (FEC), indicated by appending a short bit sequence (the label) to the packet. Each router in the network has a table indicating how to handle packets of a specific FEC type, so once the packet has entered the network, routers don't need to perform header analysis.

Instead, subsequent routers use the label as an index into a table that provides them with a new FEC for that packet. This gives the MPLS network the ability to handle packets with particular characteristics (such as coming from particular ports or carrying traffic of particular application types) in a consistent fashion.

Packets carrying real-time traffic, such as voice or video, can easily be mapped to low latency routes across the network, this is something that's challenging with conventional routing [22].

Chapter 4

4. Control and User Plane Systems Introduction

4.1 Introduction of Control Plane Systems

Performance management system, which is called in this thesis “control plane systems” is a protocol model defined by ITU-T for managing open systems in a communications network. It provides a framework for achieving interconnectivity and communication across heterogeneous operations system and telecommunication networks and also it provides the method of efficiently monitoring network performance and facilitates network optimization and troubleshooting. It includes element management system and operating support system. An element management system consists of systems and applications for managing network elements (NE) on the network element management layer of the performance management system. As recommended by ITU-T, the element management system's key functionality is divided into five key areas: fault, configuration, accounting, performance and security. An operational support system (OSS) is a group of computer programs or an IT system used by communications service providers for monitoring, controlling, analyzing and managing a computer or telephone network system. Generally speaking, performance management system is applicable to the data reported by the NEs and the operations performed on the operating support system (OSS) [13, 14, 15]. Figure 4.1 shows the performance management architecture.

The performance management system architecture includes: performance surveillance (PRS), element management systems (i.e. M2000) and network elements (NEs). NEs communicate with each other based on the TCP/IP protocol. It includes:

- BSC and eGBTS on the GSM network;
- RNC and NodeB on the UMTS network;
- eNodeBs, including macro, micro, and pico eNodeBs on the LTE network;
- MBSC and multimode base station on the single RAN network.

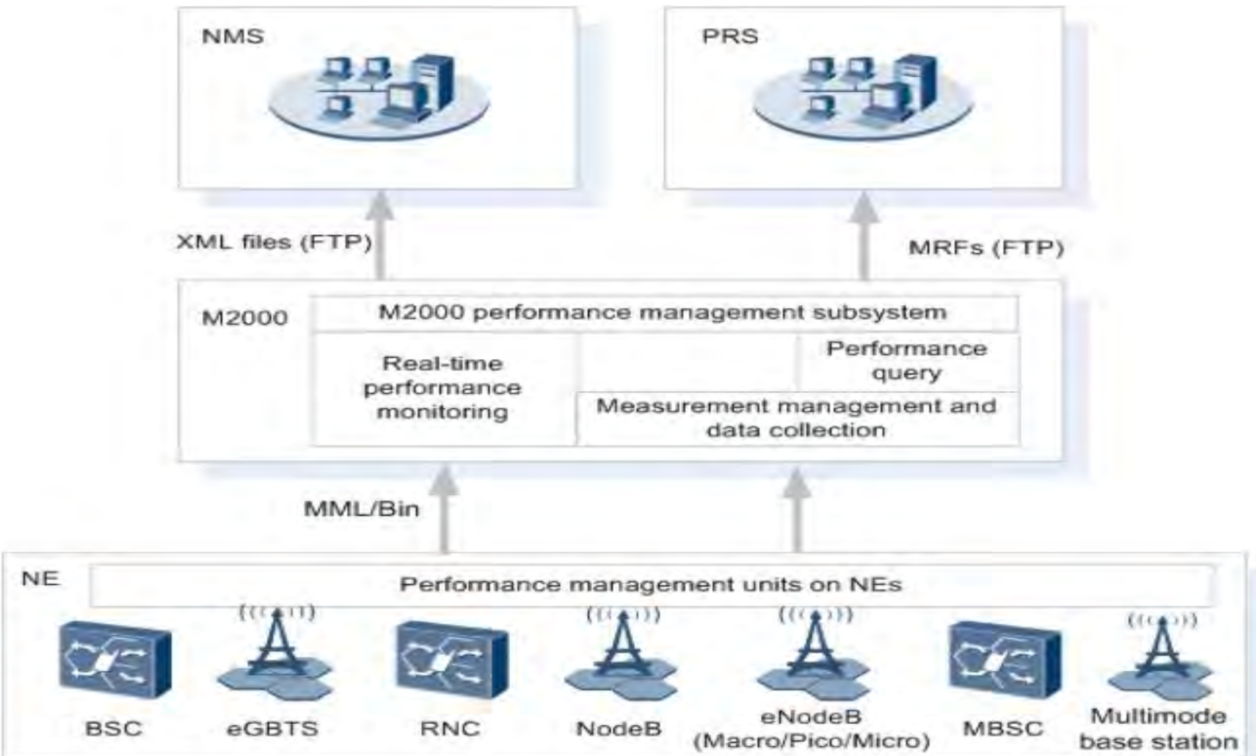


Figure 4.1 Performance management system architecture [15].

4.1.1 Introduction of the M2000

The M2000 in figure 4.1 is an element management system (EMS) for mobile NEs. It is the specific carrier of performance management system, which plays an important role in the performance management system architecture. M2000 performance management provides source data for the network management system (NMS) and PRS.

The network element management system can obtain the files from specified directories on the M2000, after NEs send performance measurement result files to the M2000 [15]. More specifically, M2000 provides the following performance management functions [15]:

- data collection and storage,
- data query,
- threshold alarm,
- data export.

4.1.2 Introduction of the Performance Surveillance (PRS)

The performance surveillance (PRS) in figure 4.1 is a platform for analyzing performance data of mobile networks, customizing reports, and displaying reports. The PRS is applicable to routine maintenance of mobile network. It can monitor and analyze the performance of the entire network. The PRS provides open performance interfaces, centrally managing the performance data collected from multiple operation support systems (OSSs). It also aggregates multi-dimension performance data, provides specific data storage policies, and stores key data for a long period of time. Using these features, users can quickly query performance data. The M2000 provides the PRS with performance data, configuration data and license data of network devices through the file interface [14].

4.1.3 Nastar Radio Network Simulator

4.1.3.1 Introduction

In mobile communication systems, the need for an advanced radio network optimization simulator is increasing because of the demands of real time services and high quality services, are increasing.

In UMTS radio access, there is a need for an advanced radio network optimization simulator. The UMTS air interface is more dynamic than the GSM air interface and therefore, optimized radio resource management (RRM) algorithms need to be developed to fully utilize the UMTS capabilities. The capacity and the achieved service quality in a UMTS network depend on many different parameters like power, code, processing gain, service distributions, etc. In order to effectively study RRM algorithms complicated tools are needed. A Nastar Optimization simulator is one of these complicated tools which is used in this thesis to evaluate the city's UMTS network voice QoS performance. It is used to obtain capacity, coverage, and quality of live network, to support optimization engineer and tuning the radio network parameters. It is a network optimization system that is used for the simulation, analysis and troubleshooting of QoS related problems on the live mobile networks. It is deployed on the distributed network in Client mode and connected to M2000. The Nastar and the M2000 must be in the same LAN and their IP addresses must be on the same network segment. Figure 4.2 shows the position of the Nastar radio network simulator on network.

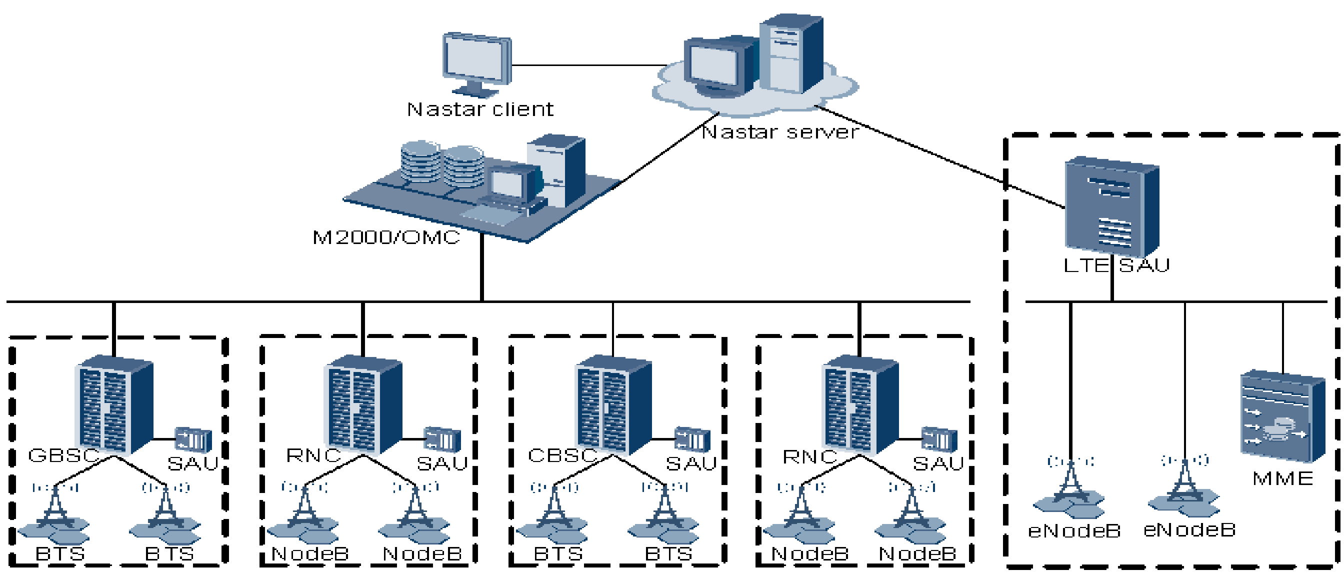


Figure 4.2 Position of the Nastar on the network [13].

The Nastar collects required simulation, analysis and troubleshooting (i.e., MR, CHR, counters and KPIs) data from NEs through the M2000 data center and provides theme analysis for network optimization [13]. It also simulates and analyzes these data to locate and display the coverage, quality and subscriber distribution of the studied network.

The Nastar mainly uses: configuration data, engineering parameters and matrices related to mobile subscribers during network simulation, analysis and troubleshooting. The analysis result helps users to determine whether problems such as weak coverage, cross coverage, and poor service quality occur on the live network.

The Nastar optimization simulator uses hybrid model (i.e., standard propagation model, okumura model, cost 231 Hata model, ray tracing model, path loss model, shadowing model, fast fading model, mobility model, traffic model, interference model, RRM model, etc) to locate and display required information on a map based on MR data, CHR data and property matrices related to mobile subscribers to show the distribution of network coverage, quality, traffic and abnormal events in an intuitive and comprehensive manner.

The Nastar radio network simulation is divided into two parts: link and system level simulations. Link level simulation is used for link level performance simulation. It is complex, because its simulation including everything from each specific link in the network with respect to its link, which is far too high with the required simulation resolutions and simulation times. It is used to build a model for the system level simulator which is used to simulate the whole studied network's performance. System level simulation is used to simulate the performance of a system with a large number of mobile terminals, base stations, and algorithms operating in such a system. It requires at least 10-20 minutes to show simulation output.

The data used in the link level simulation are averaged and used in the system level simulation. In this thesis we preferred, system level simulation to evaluate the existing UMTS network's voice QoS of the city. In the next sections, some of the models are described roughly.

4.1.3.2 Okumura Model

The okumura model is classical empirical model to measure the radio signal strength in built up areas. This model is perfect in those cities which having dense and tall structure [41]. This model is applicable for less than 3GHz frequency range. In this model the path loss calculated in urban, sub-urban and rural area as below:

$$\mathbf{PL(dB) = L_f + A_{mn}(f, d) - G(h_{te}) - G(h_{re}) - G_{AREA}} \quad (1)$$

Where L_f is free space path loss, $A_m(f,d)$ is median attenuation relative to free space, $G(hte)$ is NodeB antenna height gain factor, $G(hre)$ is NodeB antenna height gain factor, h_{te} is transmitter antenna height, h_{re} is receiver antenna height is G_{AREA} is gain due to type of environment and **d** is distance between Tx & Rx

4.1.3.3 Standard Propagation Model

The standard propagation model is a propagation model based on the Hata formulas and is suited for predictions in the 150 to 3500 MHz band over long distances (from one to 20 km). It is best suited to GSM 900/1800, UMTS, CDMA-1 and CDMA-2000, and WiMAX radio technologies [41].

$$\mathbf{P_R = P_{Tx} - K_1 + K_2 \times Log(d) + K_3 \times Log(H_{Txeff}) + K_4 \times DiffractionLoss + K_5 \times Log(d) \times Log(H_{Txeff}) + K_6 \times H_{Rxeff} + K_7 \times Log(H_{Rxeff}) + K_{clutter} \times f(clutter) + K_{hill,LOS}} \quad (2)$$

Where $\mathbf{P_R}$ is received power , $\mathbf{P_{Tx}}$ is transmitted power, $\mathbf{K_1}$ is constant offset, $\mathbf{K_2}$ multiplying factor for Log(d), **d** is distance between the receiver and the transmitter,

K_3 is multiplying factor for Log(HT_{xeff}), H_{Txeff} is effective height of the transmitter antenna, **K4** is multiplying factor for diffraction calculation, must be a positive number, , **DiffractionLoss** is losses due to diffraction over an obstructed path, **K5** is multiplying factor for Log(H_{Txeff}) x Log(d), **K6** is multiplying factor for H_{Rxeff}, **K7** is multiplying factor for Log(H_{Rxeff}), H_{Rxeff} is mobile antenna height, $K_{clutter}$ multiplying factor for f(clutter), **f(clutter)** is average of weighted losses due to clutter and K_{hill} is LOS corrective factor for hilly regions (0 in case of NLOS).

4.1.3.4 Interference Model

The calculation of interference is an essential process of the Nastar simulator. The better the interference modeling is, the more accurate results can be obtained. On the other hand, the interference calculation is very computer time consuming: the received interference has to be calculated every time when the interference situation changes due to the fast power control.

The total interference power $I_{bs(k)}$ received by a NodeB **k** is calculated as follows:

$$I_{bs(k)} = \sum_{\substack{n=1 \\ n \neq m}}^{N} I_{n,k} \cdot \left[\frac{\sum_{i=1}^{L} g_{i,n,k}}{\sum_{i=1}^{L} \hat{g}_{i,n,k}} \right] \cdot p_{ms(n)} \qquad (3)$$

where **N** is the total number of active NodeBs in the system and **m** is index for the observed user. $L_{n,k}$ is path loss (attenuation due to distance and slow fading) between the NodeB **k**, and the UE **n**. $\Sigma g_{i,n,k}/\Sigma \hat{g}_{i,n,k}$ is the multipath fading normalized to having long term average equal to one and **L** is the number of multipath components. $p_{ms(n)}$ is the transmission power of the UE **n**.

After the interference calculations, the uplink signal to noise ratio $SNR_{ul(m,k)}$ can be calculated for the user **m** connected to the NodeB **k** as:

$$\mathbf{SNR_{ul(m,k)}} = \sum_{i=1}^{L} \left(\frac{\mathbf{Gp_{ms(\mathit{m})} a_i}^2}{\mathbf{I_{bs(k)}} + \mathbf{N}} \right) \tag{4}$$

where $\mathbf{Gp_{ms(m)}}$ is the processing gain, $\mathbf{a_i}$ is amplitude attenuation of path $\mathbf{i}$ and $\mathbf{L}$ is the number of allocated RAKE receiver fingers. In equation (4) it is assumed that the received signals are combined coherently with maximal ratio combining [42].

In downlink the effect due to orthogonal codes has to be considered. Because of the multipath propagation perfect orthogonality cannot be assumed. For optimal maximal ratio combining, the downlink signal-to-noise-ratio $\mathbf{SNR_{dl(m)}}$ for a user $\mathbf{m}$ can be calculated as:

$$\mathbf{SNR_{dl(m)}} = \sum_{k=1}^{M} \left(\sum_{i=1}^{L_k} \left(\frac{\mathbf{Gp_{ms(\mathit{m},\mathit{k})} a_{k,i}}^2}{\mathbf{I_{ms(m)}} - \mathbf{P_{bs(k)} a_{k,i}}^2} \right) \right) \tag{5}$$

where $\mathbf{I_{ms(m)}}$ is the total interference power received by the NodeB $\mathbf{m}$, $\mathbf{M}$ is number of NodeBs in the active set, $\mathbf{P_{bs(m,k)}}$ is the transmitting power for the observed user from the NodeB $\mathbf{k}$, $\mathbf{P_{bs(k)}}$ is the total power transmitted from the NodeB $\mathbf{k}$, $\mathbf{a_{k,i}}$ is amplitude attenuation of the channel tap $\mathbf{i}$ and $\mathbf{L_k}$ is the number of allocated RAKE receiver fingers from NodeB $\mathbf{k}$.

4.1.3.5 Mobility Model

In Nastar simulator the users' mobility data from MR, CHR and property matrices related to users are located and displayed on the simulation map according to the mobility model.

The real handover generation and completion processes are collected from mobility data and used by this simulator. The mobility model in the Nastar simulator is developed for micro and macro cellular environments [42].

4.1.3.6 Traffic Model

The construction of an adequate traffic model is an important task in the performance evaluation of wireless communication networks. The behavior of the developed traffic model is the mimic behavior of the real network traffic. Nastar radio network simulator, simulates the real network traffic behavior by using the users' traffic data from MR, CHR and property matrices related to users to locate and display the distribution of traffic on map according to the traffic model. The real call generation and completion processes are collected from traffic data and used by this simulator, this is made according to a Poisson process [42].

4.1.3.7 Nastar Radio Network Simulator Hybrid Model

Consider a UMTS network with a set of cells denoted by **I**. The service area is represented by a grid of bins (small square or rectangular areas) with a certain resolution, assuming the same signal propagation conditions across every bin. The set of bins is denoted by **J** .

Let $\mathbf{P}_i^{max}$ be the maximum available transmit power in cell **i**. The total amount of power available in the cell depends on the output power of the NodeB power amplifier and the software parameters that define RRM in the cell. The output power of a NodeB power amplifier is a hardware-specific parameter restricted by the 3GPP linearity requirements [43]. For macro cells, the typical output power is 20–30 W.

We use $\mathbf{P}_i^{Tot}$ ($\mathbf{P}_i^{Tot} \leq \mathbf{P}_i^{max}$) to denote the total allocated (actually used) DL transmit power in cell **i**. The power is used for both control signaling and traffic data in the cell.

In a real network, the amount of power $\mathbf{P}_i^{Tot}$ depends on the current DL traffic. Moreover, the DL load of cell **i** is often measured as $\mathbf{P}_i^{Tot} / \mathbf{P}_i^{max}$. Therefore, total allocated DL power can be expressed as $\mathbf{P}_i^{Tot} = \eta_i^{DL} * \mathbf{P}_i^{max}$, where η_i^{DL} is the *DL load* factor.

The amount of power allocated in cell **i** to CPICH is denoted by $\mathbf{P}_i^{CPICH}$ ($\mathbf{P}_i^{CPICH} < \mathbf{P}_i^{Tot}$). For a single cell, a higher value of $\mathbf{P}_i^{CPICH}$ means larger coverage area of cell **i**, but, on the other hand, less power available to serve user traffic in the cell and therefore a decrease incapacity. This is especially important if the transmit power levels of other common channels are set relative to CPICH, which is a common practice among operators [43]. Therefore, it is natural to require that the amount of CPICH power does not exceed some percentage (e.g.,10% in [44]) of the total available power $\mathbf{P}_i^{max}$ in the cell.

We use a set of power gain parameters $\{\mathbf{g}_{ij},\ \mathbf{i} \in \mathbf{I},\ \mathbf{j} \in \mathbf{J}\}$ to account for factors that affect the strength of the pilot signal received at a user terminal. Parameter $\mathbf{g}_{ij}$ ($0 < \mathbf{g}_{ij} < 1$) is the power gain between the antenna in cell **i** and bin **j**. The parameter aggregates all losses and gains between the NodeB transmit unit and the receiver of the user terminal. The power gain value depends on the antenna line characteristics (feeder cable, jumper, and connector losses, loss/gain due to mast head amplifier), antenna configuration (e.g., height, azimuth, mechanical tilt, electrical tilt), user's equipment, and path loss which depends on the radio propagation environment and distance. For a specific NodeB equipment, the NodeB and antenna characteristics are usually known from the manufacture or can be derived.

Path loss predictions can be obtained by hybrid model [44]. The accuracy of the propagation predictions does not only depend on the chosen model, but also, among others, the accuracy in terrain information and data resolution (bin size).

By the 3GPP standard [43], to achieve the CPICH quality of cell **i** in bin **j**, the **Ec/I0** of CPICH from cell **i** must meet a given threshold, i.e.,

$$\frac{\mathbf{g_{ij}\, P_i}^{CPICH}}{\mathbf{I_j}} \geq \boldsymbol{\gamma_0} \qquad (6)$$

Where $\mathbf{g_{ij}}\, \mathbf{P_i}^{\mathbf{CPICH}}$ is the received CPICH power from cell **i** in bin **j**, $\mathbf{I_j}$ is the total received power spectral density in bin **j**, and $\gamma\mathbf{0}$ is the ***Ec/I*0** target.

The ***Ec/I*0** threshold is specific for each user terminal and it is fixed. The 3GPP standard [43] enforces the mobile terminals to be able to detect a pilot signal with ***Ec/I*0 ≥ −20 dB**. However, a slightly higher threshold, e.g., −18 dB, is usually used in practice (see, for example, [42,43]).

We compute the amount of interference in bin **j** as:

$$\mathbf{I_j} = \sum_{i\in I} \mathbf{g_{ij}}\, \boldsymbol{\eta_i}^{DL} \mathbf{P_i}^{Max} + \mathbf{V_j} + \mathbf{I_j}^{\mathbf{A}} \qquad (7)$$

where $\mathbf{v}_j$ is the thermal noise power in bin **j**, I_j^A is the adjacent channel interference in bin **j**.

The adjacent channel interference is the amount of power leaking from an adjacent carrier either from the operator's own network or from the competitor's network due to non-linearity in power transmitters. Moreover, the user equipment is typically not able to completely filter out signals received on adjacent channels.

For DL, the limit on the out-of-band power leakage is controlled by 3GPP [43] which specifies the Adjacent Channel Leakage power Ratio (ACLR) for NodeB power amplifiers. The 3GPP standard [8] specifies the requirements on user equipment's ability to receive a signal at the assigned channel in the presence of an adjacent channel signal.

The measure of this ability is known as Adjacent Channel Selectivity (ACS). In the own network, the operator can compute the adjacent channel interference as the total received power on the channel but with power gains scaled by the Adjacent Channel Protection (ACP) factor which is computed from ACLR and ACS.

The information about the competitor's network is typically not available, therefore the adjacent channel interference in such a situation can be modeled as a grid of location-specific interference values obtained either empirically or theoretically. From equation (43), the DL interference is at its maximum value, when η_i^{DL} = 1.0 for all cells.

In order to utilize power resources efficiently, the CPICH transmit power should not be

set to more than what is necessary. That is, to provide CPICH coverage in bin **j**, cell **i** does not need to use CPICH transmit power higher than $\mathbf{P}_{ij}$, which can be derived from (7) as follows:

$$\mathbf{P_{ij}} = \frac{\boldsymbol{\gamma_0}}{\mathbf{g_{ij}}} \cdot \left(\sum_{i \in I} \mathbf{g_{ij}} \, \boldsymbol{\eta_i}^{DL} \mathbf{P_i}^{Max} + \mathbf{V_j} + \mathbf{I_j}^{\mathbf{A}} \right) \qquad \mathbf{(8)}$$

To achieve CPICH coverage, the received pilot signal must not only satisfy the Carrier-to-Interference Ratio condition but also the minimum CPICH received signal code power (RSCP) requirement that enables the user equipment to properly decode pilot signals is:

$$\frac{\mathbf{g_{ij}} \, \mathbf{P_i}^{CPICH}}{\mathbf{I_j}} \geq \boldsymbol{\gamma_1} \qquad \mathbf{(9)}$$

where γ_1 is the receive sensitivity threshold.

A typical value of $\gamma\mathbf{1}$ is in the range [−120,−114] dBm. Thus, combining the **CPICH Ec/I0** and **RSCP** requirements, the minimum CPICH power needed to provide coverage by cell **i** in bin **j** can be found as follows:

$$\mathbf{P_{ij}} = \boldsymbol{max}\left\{\frac{\boldsymbol{\gamma_1}}{\mathbf{g_{ij}}}, \frac{\boldsymbol{\gamma_0}}{\mathbf{g_{ij}}} . \left(\sum_{i \in I} \mathbf{g_{ij}}\ \boldsymbol{\eta_i}^{DL} \mathbf{P_i}^{Max} + \mathbf{V_j} + \mathbf{I_j}^{\mathbf{A}}\right)\right\} \qquad \mathbf{(10)}$$

Our CPICH coverage modeling approach is demonstrated in Figure 4.3, where the **CPICH RSCP** is at least γ_1 in the area bounded by a thick solid line and the **CPICH *Ec*/*I*0** satisfies below in the colored area. Thus, the CPICH coverage in the this example is represented by the intersection of the two areas. In practice, the area with the CPICH signal satisfying the **RSCP** requirement is typically at least as large as the area where the **CPICH *Ec*/*I*0** threshold is met. This is especially true in urban scenarios where the interference is high. Therefore, planning for the **CPICH *Ec*/*I*0** quality with the following up CPICH power adjustment with respect to the **RSCP** requirement is justified in practice.

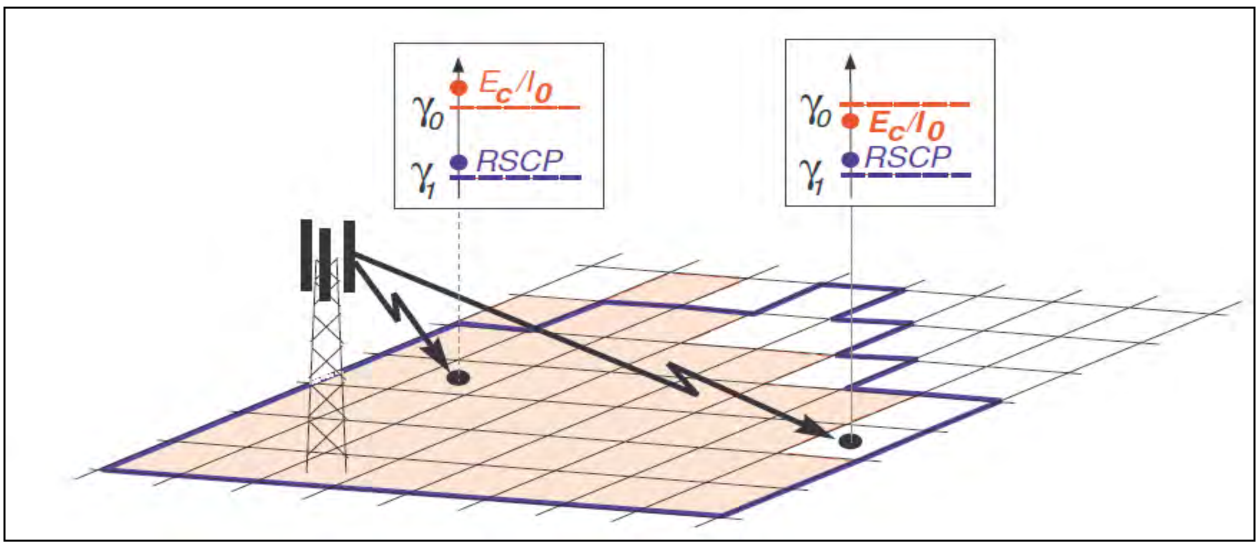


Figure 4.3 Modeling CPICH coverage and quality [43]

Let $\mathbf{\Pi}_i^{max}$ ($\mathbf{\Pi}_i^{max} < \mathbf{P}_i^{\mathbf{max}}$) be the upper limit for pilot power in cell **i**, and $\mathbf{\Pi}_i^{min}$ ($0 \le \mathbf{\Pi}_i^{min} \le \mathbf{\Pi}_i^{max}$) be the lower pilot power limit in cell **i**.

The pilot power is said to be unbounded, if $\Pi_i^{min} = 0$ and $\Pi_i^{max} = P_i^{max}$), but such a scenario is unlikely to be used in a real-life network and therefore should also be avoided when planning a network.

In UMTS networks, the lower and the upper pilot power limits in a cell are usually set to **1–3%** and **15%** of the total transmission power of the cell, respectively [46].

The pilot power limits can be either introduced into the model as a set of constraints or they can be accounted for in the preprocessing step. The latter allows us to not only reduce the number of constraints, but also to significantly reduce the problem size. The smaller are the Π_i^{max} values, the smaller the sets of feasible pilot power levels become. To ensure the lower pilot power limit in the preprocessing step, P_{ij} values that are below Π_i^{min} are set to this minimum value. To consider the upper limits, P_{ij} values that exceed the upper limit Π_i^{max} are excluded from the list of possible pilot power settings. This can be done through the use of sets I_j and J_i.

For each bin j, we introduce a set $I_j \subseteq I$ which contains all the cells that may cover bin j with a feasible pilot power level i.e.,

$$\mathbf{I_j} = \left\{\mathbf{i} \in \mathbf{I} : \mathbf{P_{ij}} \leq \mathbf{\Pi_i}^{\boldsymbol{Max}}\right\} \qquad (11)$$

For each cell i, we define a set $J_i \subseteq J$ that contains all bins that may be covered by the cell i.e.,

$$\mathbf{J_i} = \left\{\mathbf{j} \in \mathbf{J} : \mathbf{P_{ij}} \leq \mathbf{\Pi_i}^{\boldsymbol{Max}}\right\} \qquad (12)$$

4.2 Introduction of User Plane Systems

Drive testing (DT) system is a method of measuring and assessing the coverage, capacity and QoS of a mobile radio network from user perspective point of views, which is called in this thesis "user plane systems". It can detect and record a wide variety of the physical and virtual parameters of mobile communication service in a given geographical area.

The technique consists of using a car, Nemo out door, Nemo handy, global positioning system (GPS), DT route, MapInfo, Google earth, engineering parameters, scanner and laptop. By measuring what a wireless network subscriber would experience in any specific area, operators can make directed changes to their networks that provide better coverage and service to their customers [38].

The DT Analysis is a post-processing optimization platform that analyzes cellular interface data that was collected using the drive test wireless network optimization software.

The analyzed data streams are combined in the collection system and come from two sources.

- Air interface data fields that relate to the user's specific radio technology.
- Navigation position (or geodetic reference) and time stamp for each data reading.

This combined data lets you evaluate the characteristics of the cellular system to determine problem areas and plan improvements based on time of day and the physical location of the data readings [38]. Figure 4.4 shows several drive test equipments.

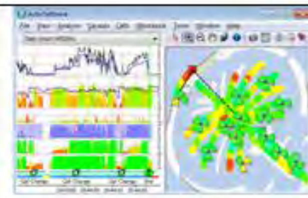
 Nemo handy	 USB cable	 Scanner	 Nemo out door
 Laptop	 GPS	 Actix Analyzer	 All tools connected

Figure 4.4 Several drive test equipments.

Nemo handy is a state of the art handled tool for testing mobile real time applications QoS/QoE and measuring the performance of wireless networks.

Its extensive application testing features are complete with voice quality testing, full application level metrics on voice and video calls, UL/DL data transfers, Web browsing, SMS/MMS messaging and ping. It provides a complete and detailed picture of the QoS performance of the end users. Nemo Outdoor is a laptop-based drive test tool for wireless network testing which supports over 300 terminals and scanning receivers from various vendors and all major network technologies. It is one of the DT tools, which is used to collect the DT data of the network, which can show the users' QoS. Through the Nemo out door, the network performance can be evaluated, the network optimization can be guided and the fault can be rectified. The collected DT data of the network on the radio network can be saved as the logfile. This facilitates the data analysis after the logfile is imported to other post-processing software (such as Actix analyzer).

The Actix is DT logfiles analysis software, which is used to analyze and process test data of networks. The Actix can also generate network test reports to meet network analysis requirements of customers. The generated test reports effectively reflect the operation status of radio networks and provide guidelines for network verification, network evaluation, network optimization, and fault location. Therefore, the test reports help operators learn about network performance, quickly locate network problems and improve work efficiency. A scanner is a radio receiver that can automatically tune or scan, two or more separate frequencies, stopping when it finds a signal on one of them and then continuing to scan other frequencies when the initial transmission stop.

Chapter 5

5. Data Collection and Analysis Results

5.1 Control Plane System Data Collection and Analysis Results

The Nastar is deployed on the EMS side of an operator's network. The Nastar collects required analysis data from NEs through the EMS data center and provides theme analysis for network optimization. The Nastar mainly uses the following three types of data during UMTS network analysis: UMTS configuration data, UMTS engineering parameters and data related to Nastar analysis. UMTS configuration data are basic information for network optimization and used for the Nastar to correctly retrieve data of each cell from data sources. UMTS engineering parameters are a basic information for network optimization that include longitudes, latitudes, etc of sites, and azimuths, electrical tilt and mechanical tilt angles of antennas and used for displaying the analysis data geographically.

5.1.1 Network Geographic Observation Data

UMTS service geographic analysis uses to obtaining UMTS service geographic analysis data, creating a UMTS service geographic analysis task, viewing the UMTS service geographic analysis result and exporting a UMTS service geographic analysis report. The UMTS service geographic analysis function enables the Nastar to locate and display required information on a map based on measurement report (MR) data, call history record (CHR) data and property matrices related to mobile subscribers to show the distribution of network coverage, quality, traffic and abnormal events in an intuitive and comprehensive manner. This function helps us to identify problem areas on the network and also reduce drive test costs [13].

Geographic observation task is created on Nastar for duration of continuous seven days (March 12-18, 2016), Figure 5.1.

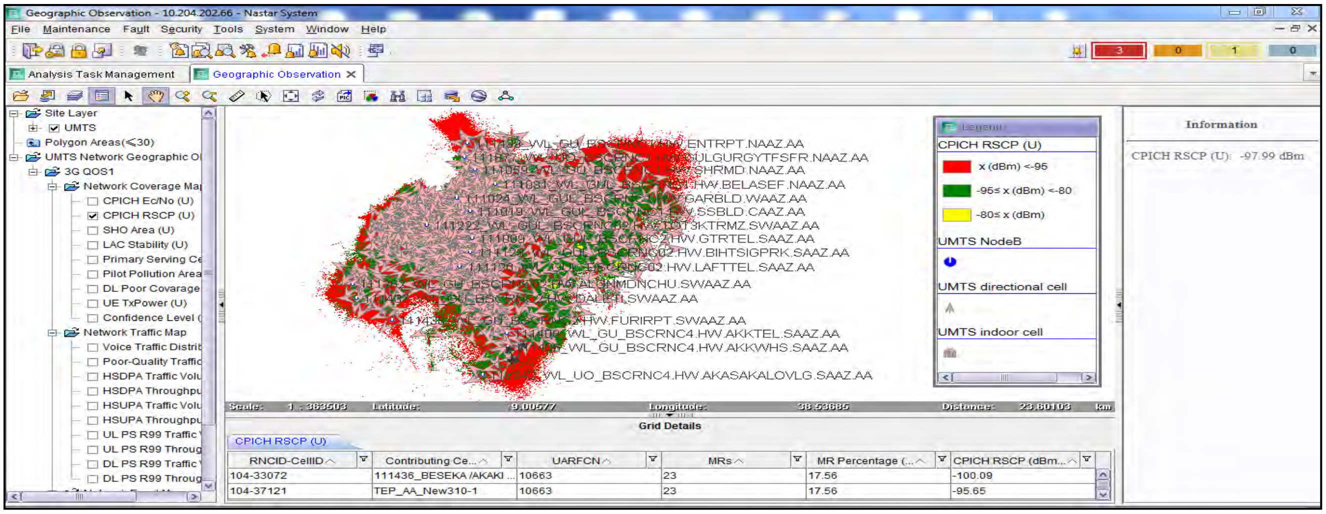


Figure 5.1 Interface for showing UMTS network geographic observation results.

5.1.1.1 Addis Ababa UMTS Network Coverage Observation

The coverage simulation snapshot is geographically rendered based on the average RSCP value in MRs in the raster view. Its range is from -115dBm to -25dBm.

Figure 5.2 shows the Addis Ababa's UMTS network coverage performance simulation snapshot. On the snapshot the coverage results presented as a plot of dots (user locations) of different color depending on whether or not the expected coverage is achieved. Red color represents (RSCP <-95dBm) poor coverage, green color represents (-95dBm <= RSCP < -80dBm) good coverage and yellow color represents (RSCP >= -80dBm) excellent coverage. From this simulation observation, in general 51.8% and 48.2% of the city's coverage was good and poor respectively, where expected good coverage threshold is greater than or equal to 95%. The simulation result shows that the central Addis Ababa's coverage is good and the edge of Addis Ababa's coverage is poor (e.g. north AA).

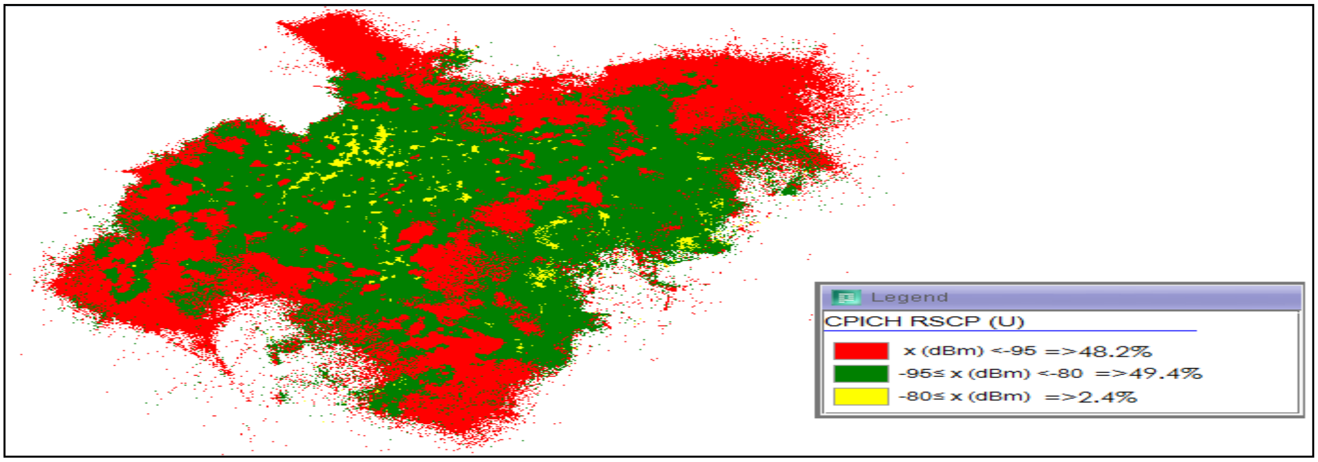


Figure 5.2 Addis Ababa UMTS coverage simulation snapshot.

5.1.1.2 Addis Ababa UMTS Voice Traffic Distribution Observation

The voice traffic distribution simulation snapshot is rendered based on the traffic volume (unit: number of MRs) of voice services (service, AMR) in the raster view. Figure 5.3 shows the Addis Ababa UMTS network voice traffic distribution simulation snapshot. On the snapshot the voice traffic distribution results presented as a plot of dots (user locations) of different color depending on their volume. Red and turquoise colors represent high traffic volume distribution. The simulation result shows that the central Addis Ababa's voice traffic distribution is high and the edge of Addis Ababa's voice traffic distribution is low (e.g. north AA).

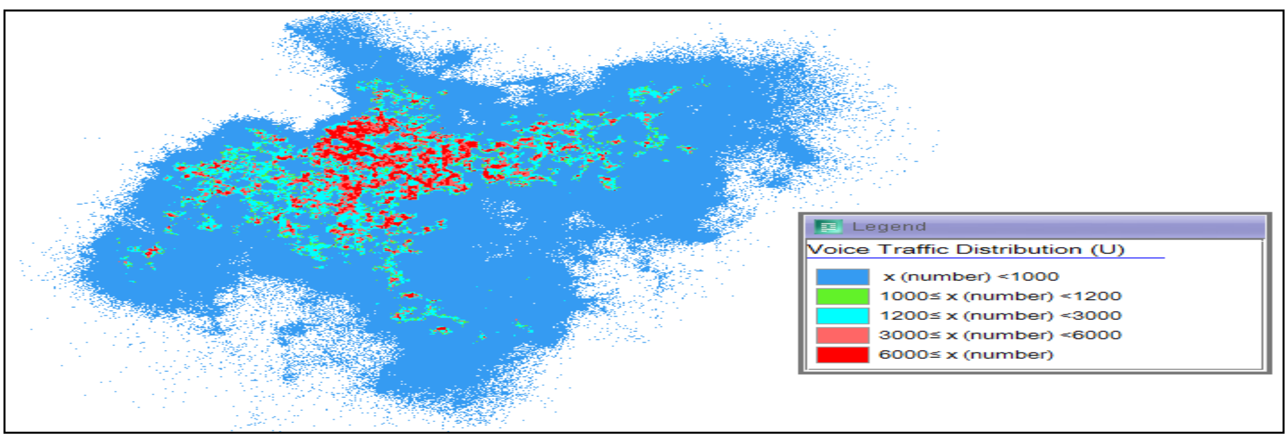


Figure 5.3 Addis Ababa UMTS traffic distribution simulation snapshot.

5.1.1.3 Addis Ababa UMTS Quality of Service Observation

The quality of service simulation snapshot is rendered based on the average received signal quality (Ec/No) in MRs in the raster view. Its range is from -24dB to 0dB. Figure 5.4 shows the Addis Ababa UMTS network quality of service performance simulation snapshot. On the snapshot the quality of service results presented as a plot of dots (user locations) of different color depending on whether or not the expected quality of service is achieved. Red color represents (Ec/No <-13dm) poor quality of service, green color represents (-3dB <= Ec/No < -7dB) good quality of service and yellow color represents (Ec/No >= -7dB) excellent quality of service. From this simulation observation, in general 72.5% and 27.5% of the city's quality of service was good and poor respectively, where expected good quality of service threshold is greater than or equal to 93%. The simulation result shows that the central Addis Ababa's quality of service seems good and the edge of Addis Ababa's quality of service is poor (e.g. west AA).

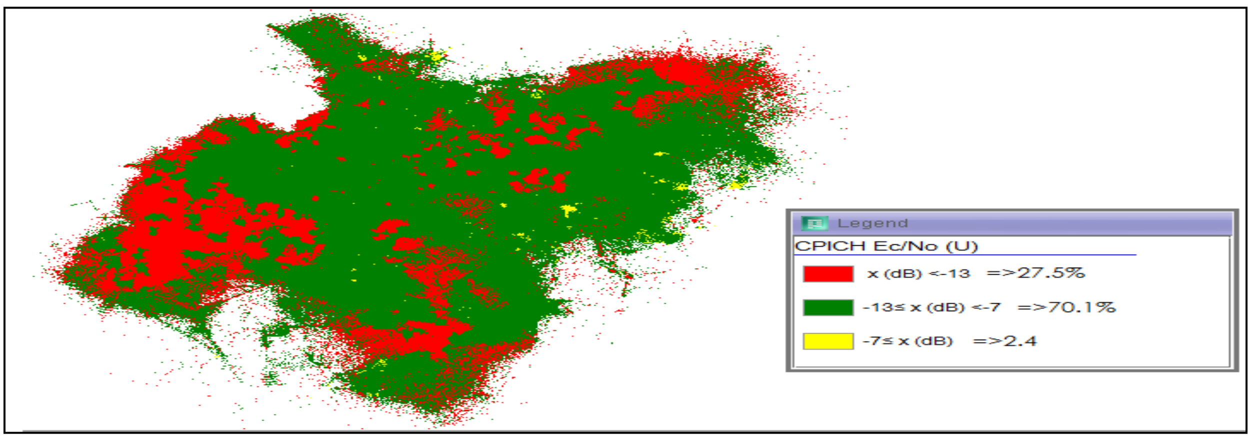


Figure 5.4 Addis Ababa UMTS quality of service simulation snapshot.

5.1.1.4 Addis Ababa UMTS Poor Quality Traffic Distribution Observation

The poor voice quality traffic distribution simulation snapshot is rendered based on the traffic volume (measured by the number of MRs) of voice services with poor quality in the raster view.

Figure 5.5 shows the Addis Ababa UMTS network poor quality traffic distribution simulation snapshot. On the snapshot the poor voice quality traffic distribution results presented as a plot of dots (user locations) of different color depending on their volume. Red and yellow colors represent high traffic volume distribution with poor voice quality. The simulation result shows that the central Addis Ababa's poor voice quality traffic distribution is high and the edge of Addis Ababa's poor voice quality traffic distribution is low (e.g. north AA).

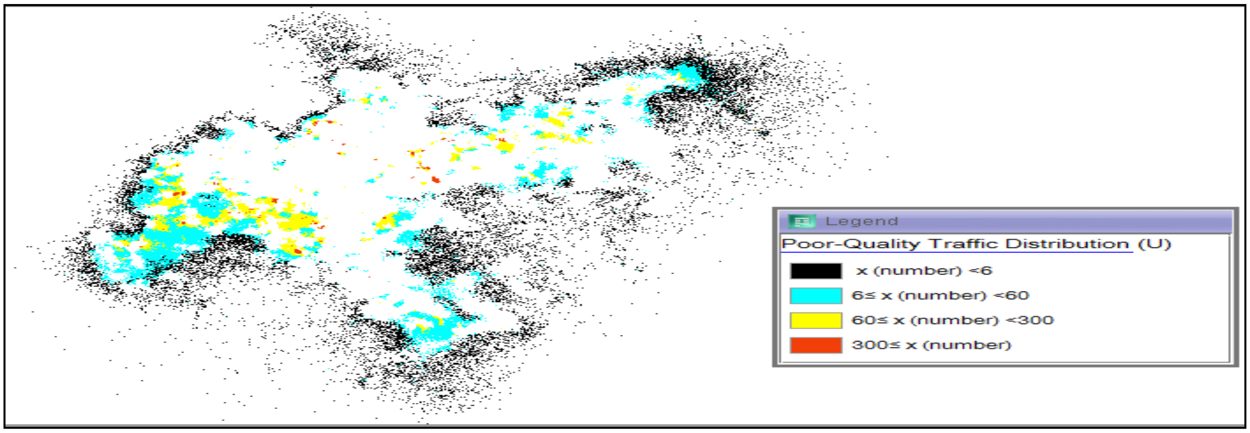


Figure 5.5 Addis Ababa UMTS poor quality traffic distribution simulation snapshot.

5.1.1.5 Addis Ababa UMTS Inter-RAT HO Failure Observation

The inter-RAT HO failure distribution simulation snapshot is rendered based on the number of inter-RAT handover failures (3G to 2G handover) in the raster view. Figure 5.6 shows the Addis Ababa UMTS network inter-RAT HO failure simulation snapshot. On the snapshot the inter-RAT HO failure distribution results presented as a plot of dots (user locations) of different color depending on their volume. Red and dark red colors represent high inter-RAT HO failure volume distribution. The simulation result shows that the central Addis Ababa's inter-RAT HO failure distribution is high and the edge of Addis Ababa's inter-RAT HO failure distribution is low (e.g. east AA).

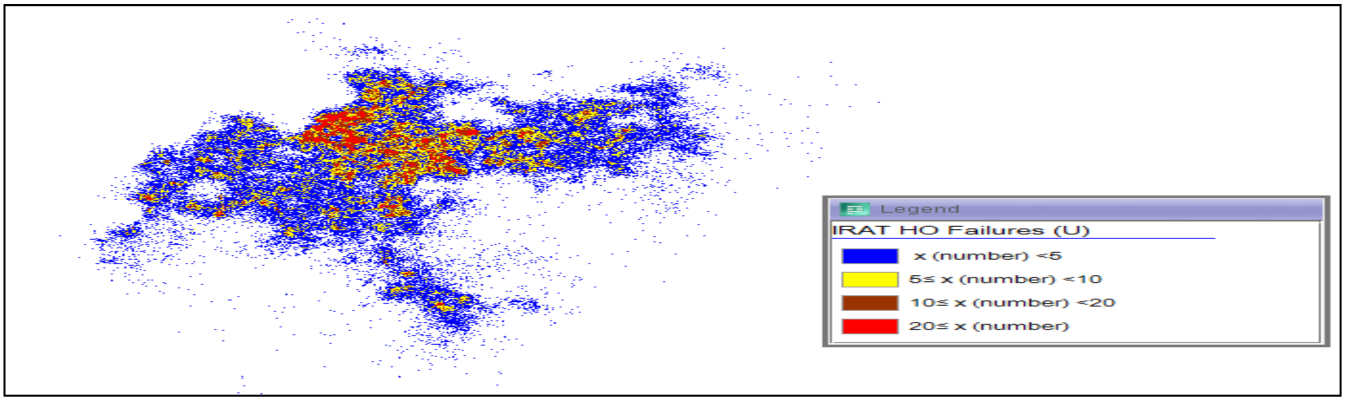


Figure 5.6 Addis Ababa UMTS inter-RAT HO failure simulation snapshot.

5.1.1.6 Addis Ababa UMTS Voice Call Drop Observation

The voice call drop distribution simulation snapshot is rendered based on the number of call drops in the raster view. Figure 5.7 shows the Addis Ababa UMTS network voice call drop simulation snapshot. On the snapshot the voice call drop distribution results presented as a plot of dots (user locations) of different color depending on their volume. Red and bright green colors represent high voice call drop volume distribution. The simulation result shows that the central Addis Ababa's voice call drop distribution is high and the edge of Addis Ababa's voice call drop distribution is low (e.g. north AA).

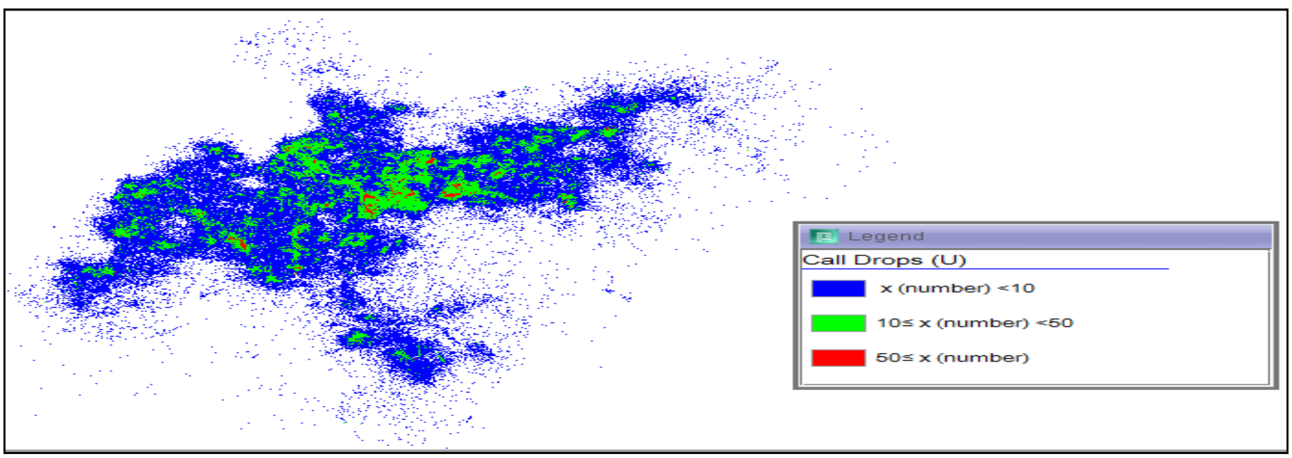


Figure 5.7 Addis Ababa UMTS voice call drop simulation snapshot.

5.1.1.7 Addis Ababa UMTS Pilot Pollution Area Observation

The pilot pollution area distribution simulation snapshot is rendered based on the number of times that pilot pollution occurs during which the average Ec/No of the CS service is smaller than a specific threshold value in the raster view. Figure 5.8 shows the Addis Ababa UMTS network Pilot Pollution simulation snapshot. On the snapshot the pilot pollution area distribution results presented as a plot of dots (user locations) of different color depending on their volume. Red and dark red colors represent high pilot pollution area distribution. The simulation result shows that the central Addis Ababa's pilot pollution area distribution is high and the edge of Addis Ababa's pilot pollution area distribution is low (e.g. north AA).

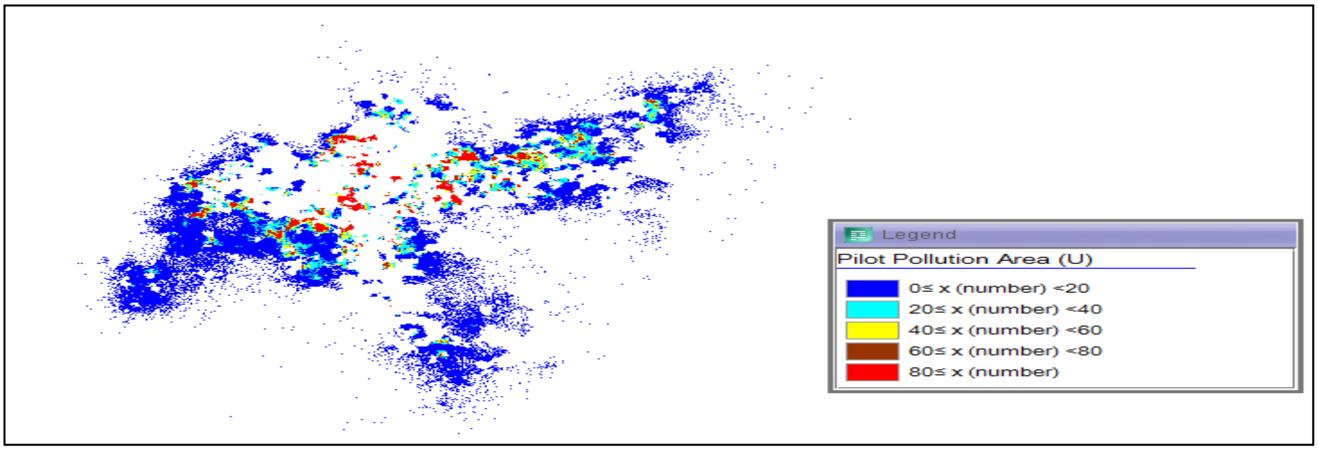


Figure 5.8 Addis Ababa UMTS pilot pollution simulation snapshot.

5.1.2 Coverage and Quality Analysis Data

The UMTS coverage and quality analysis function enables us to obtain UMTS coverage and quality analysis data, create UMTS coverage and quality analysis tasks, query UMTS coverage and quality analysis results, and export UMTS coverage and quality analysis reports from the Nastar.

The Nastar analyzes the measurement reports (MRs) sent by UEs and displays the coverage status (RSCP), signal quality (EcNo), and subscriber distribution of a measurement cell (TP size). This function helps to locate wireless network problems such as weak coverage, cross coverage, and poor service quality [13].

UMTS coverage and quality analysis task is created on Nastar for duration of continuous seven days (March 12-18, 2016). Figure 5.9 shows interface that displays UMTS coverage and quality analysis results.

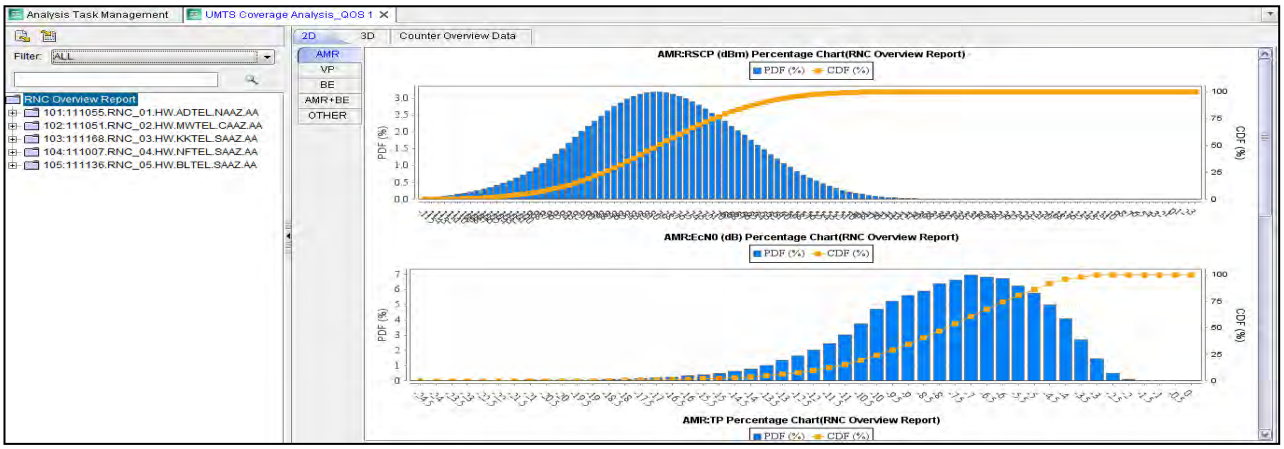


Figure 5.9 Interface for UMTS coverage analysis results.

5.1.2.1 Addis Ababa UMTS Coverage Analysis

It indicates the AMR:RSCP of the selected network in the selected time segment. Figure 5.10 shows Addis Ababa UMTS coverage analysis report. From this analysis result, in general 85.6% and 14.4% of the city's coverage was good and poor respectively, where expected good coverage threshold is greater than or equal to 95%.

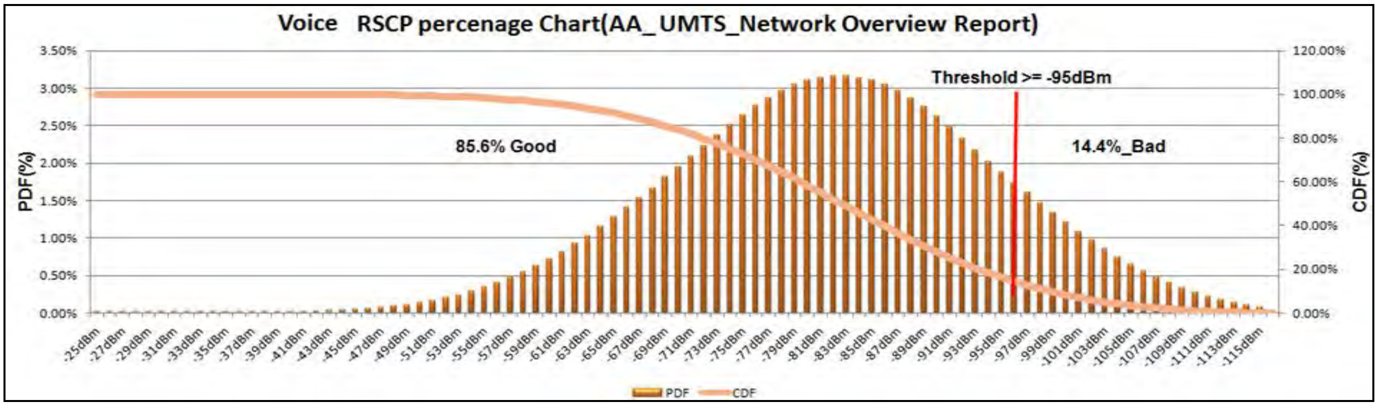


Figure 5.10 Addis Ababa UMTS coverage analysis report.

5.1.2.2 Addis Ababa UMTS Quality Analysis

It indicates the AMR:Ec/No of the selected network in the selected time segment. Figure 5.11 shows Addis Ababa UMTS quality analysis report. From analysis result, in general 89% and 11% of the city's quality of service was good and poor respectively, where expected good quality of service threshold is greater than or equal to 93%.

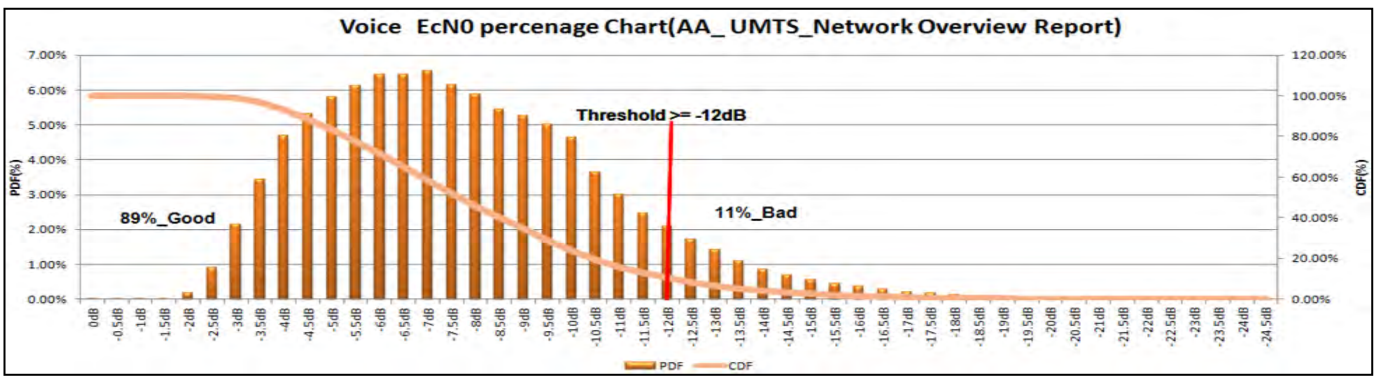


Figure 5.11 Addis Ababa UMTS quality analysis report.

5.1.2.3 Addis Ababa UMTS Overshoot Analysis

It indicates the TP MRs of the selected network in the selected time segment. Figure 5.12 shows Addis Ababa UMTS overshoot (TP size) analysis report. From analysis result, in general there are TP size which are greater than or equal to 4 in the city, where expected TP size is less than or 4.

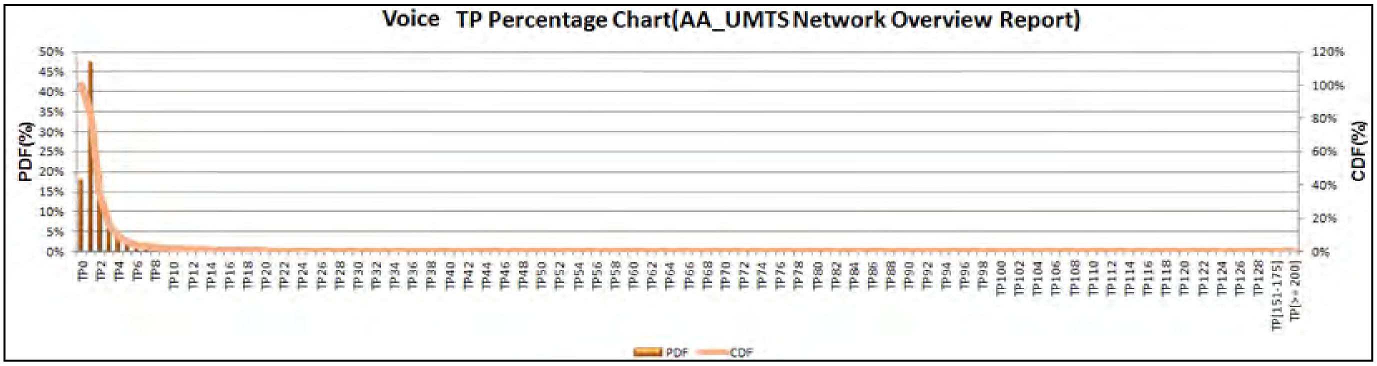


Figure 5.12 Addis Ababa UMTS TP size analysis report.

5.1.3 Network Performance Data Analysis

The UMTS network performance analysis function enables us to obtain UMTS cell performance analysis data, create UMTS cell performance analysis tasks, query UMTS cell performance analysis results, and export UMTS cell performance analysis reports. We can learn the actual causes of abnormal calls in cells by querying abnormal call data. This function helps us to improve the service quality and end users satisfaction. The voice performance is analyzed by RAB setup failure, abnormal release, delay problem, short call problem and QoS problem [13]. Cell performance analysis task is created on Nastar for duration of continuous seven days (March 12-18, 2016). Figure 5.13 shows interface that displays UMTS cell performance analysis results.

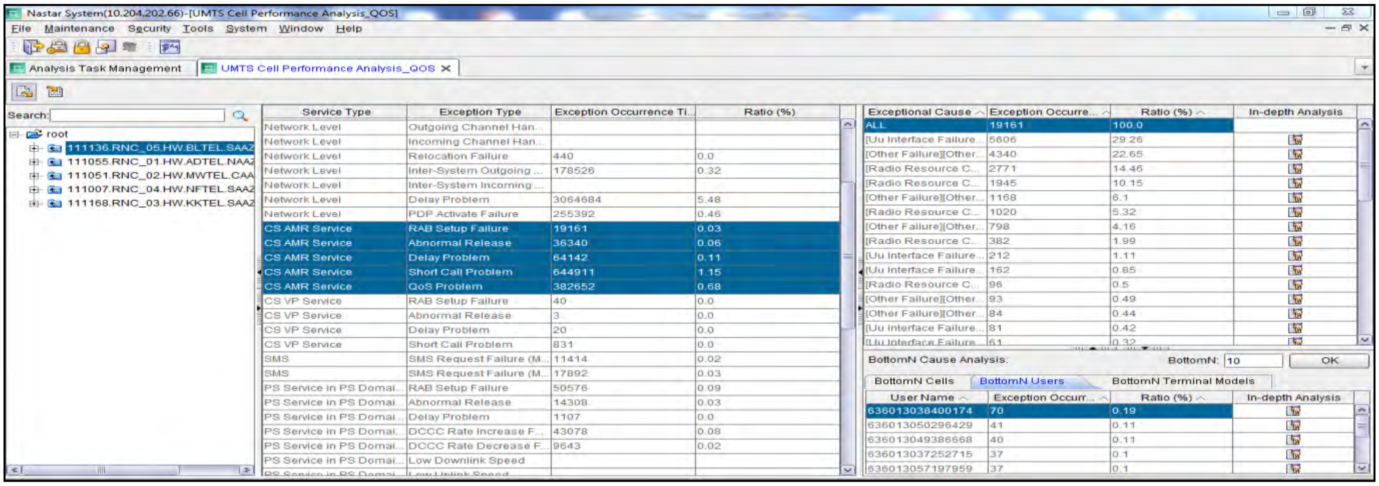


Figure 5.13 Interface for cell performance analysis results.

5.1.3.1 CS AMR Performance Analysis

Generally, the Nastar locates and analyzes network problems by monitoring performance counters, and then analyzes the counters of the entire network or for a specific RNC (i.e. RAB setup failure, abnormal release, delay problem, short call problem and QoS problem). Figure 5.14 shows Addis Ababa UMTS network performance of CS AMR services analysis results. From analysis result, in general most of the QoS of Voice is affected due to short call problem (57.75%) and QoS problem (31.38%).

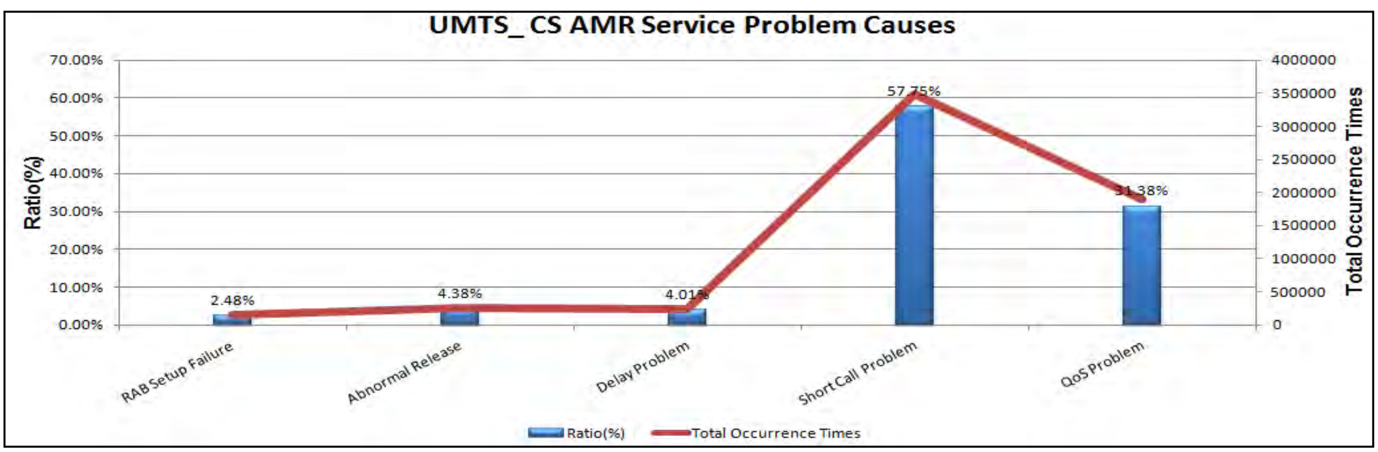


Figure 5.14 Addis Ababa UMTS network performance of CS AMR services analysis report.

5.1.3.2 CS AMR RAB Setup Failure

After a subscriber starts accessing the network, an exception occurs during the process from the RAB setup starts until the RAB setup is complete. Figure 5.15 shows Addis Ababa UMTS network RAB setup failure analysis results.

[Uu Interface Failure][Timer Expired] is occurs in the following scenarios:

Scenario 1: The RNC sends the radio bearer setup message to a UE but does not receive an ACK message before the **RbSetupRspTmr** timer expires.

Scenario 2: The RNC sends the radio bearer setup message to a UE and receives an ACK message. However, the RNC does not receive the radio bearer setup complete message from the UE before the **RbSetupRspTmr** timer expires. Channel element (CE) is a basic unit that measures the channel demodulation capabilities of a Node B. CEs are classified into uplink (UL) CEs and downlink (DL) CEs. The number of UL and DL CEs supported by a Node B is determined by the Node B hardware capabilities and the licensed CE capacity. The number of UL and DL CEs supported by the Node B hardware is called the physical CE capacity. The licensed CE capacity may differ from the physical CE capacity. CE is a concept of the Node B side. On the RNC side, it is called Node B credit.

The RNC performs admission and congestion control based on the Node B credit. In the UL, the number of Node B credit resources is twice that of CEs.

In the DL, the number of Node B credit resources equals that of CEs. When the usage of Node B credit resources in a cell exceeds the threshold for triggering basic congestion, the cell enters the basic congestion state (i.e. DL or UL CE congestion).

DL power congestion provides the percentage of RRC connection rejection messages sent by the RNC to the UE according to power congestion reasons in the cell. From analysis result, in general most of the RAB setup failure is due to resource congestion.

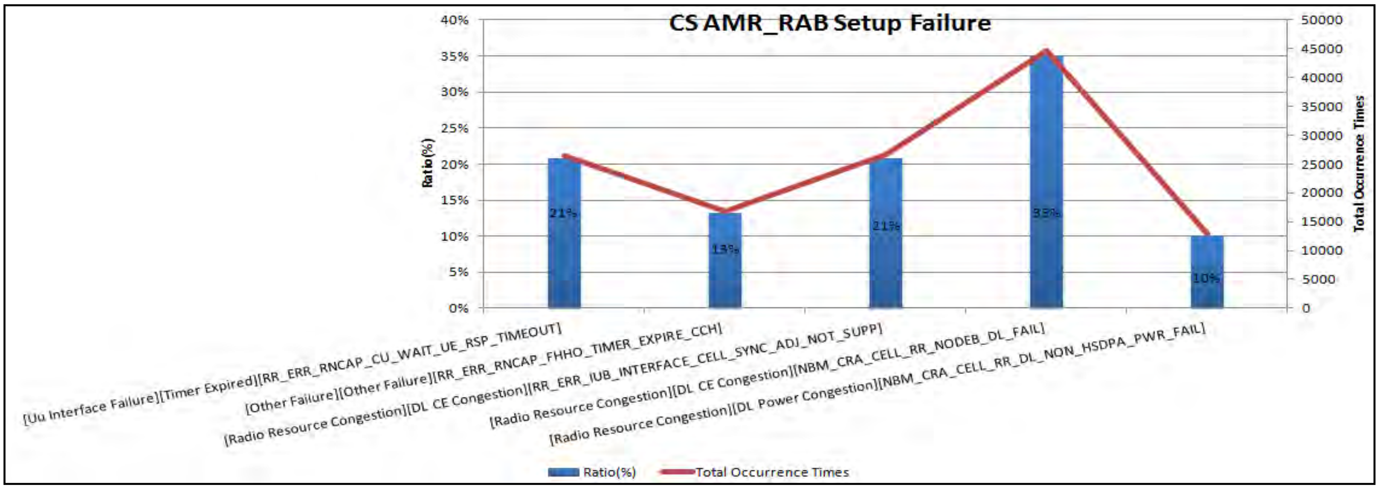


Figure 5.15 Addis Ababa UMTS network RAB setup failure analysis report.

5.1.3.3 CS AMR Abnormal Release

Abnormal release is a number of times that a call is abnormally released between the time when a subscriber hears the alerting and the time when the subscriber hears the voice of the peer-end subscriber during a successful call. Figure 5.16 shows Addis Ababa UMTS network abnormal release analysis result. From analysis result, in general most of the abnormal release is due to resource congestion.

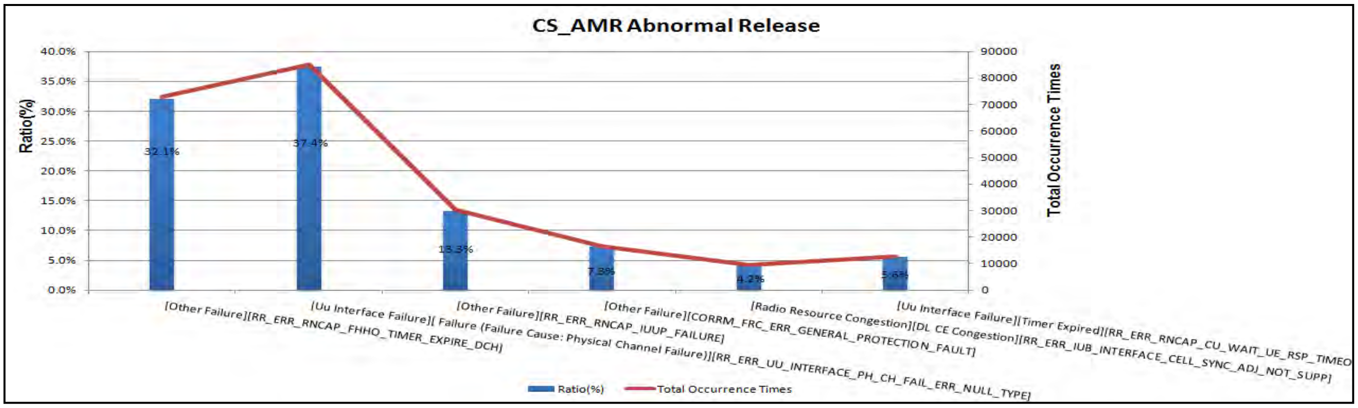


Figure 5.16 Addis Ababa UMTS network abnormal release analysis report.

5.1.3.4 CS AMR Delay Problem

Delay problem indicating that the call setup delay exceeds the predefined threshold or RAB setup delay exceeds the predefined threshold. Figure 5.17 shows Addis Ababa UMTS network delay analysis results.

The following delay exceptions might occur when a subscriber attempts to access the network:

- The call setup delay exceeds the predefined threshold (mobile-originated).
- The call setup delay exceeds the predefined threshold (mobile-terminated).
- The call setup delay exceeds the predefined threshold (alerting) (mobile-originated).
- The call setup delay exceeds the predefined threshold (alerting) (mobile-terminated).
- The RAB setup delay exceeds the predefined threshold.

Overlong call setup delay (MOC) is the call setup duration exceeds the predefined threshold when a mobile-originated subscriber attempts to access the network, if the first RAB is used for the service in the CS domain and the RRC connection setup is caused by the mobile-originated service.

Overlong call setup delay (MTC) is the call setup duration exceeds the predefined threshold when a mobile-terminated subscriber attempts to access the network, if the first RAB is used for the service in the CS domain and the RRC connection setup is caused by the mobile-terminated service. **Overlong call completion delay (alerting) (MOC**) is the call connection delay exceeds the predefined threshold when a mobile-originated subscriber attempts to access the network, if the first RAB is used for the mobile-originated service in the CS domain. We can obtain the call connection delay by using the following formula:

Call connection delay (alerting) = Time stamp of the alerting message-Signaling access time.

Overlong call completion delay (alerting) (MTC) is the call connection delay exceeds the predefined threshold when a mobile-terminated subscriber attempts to access the network, if the first RAB is used for the mobile-terminated service in the CS domain. We can obtain the call connection delay by using the following formula:

Call connection delay (alerting) = Time stamp of the alerting message signaling access time.

Overlong RAB setup delay is the RAB setup delay exceeds the predefined threshold when a subscriber attempts to access the network. We can obtain the RAB setup delay by using the following formula:

RAB setup delay (s) = RAB request complete time – RAB request time.

From analysis result, in general most of the delay problem is due to overlong call setup delay, this is due to resource congestion and weak coverage.

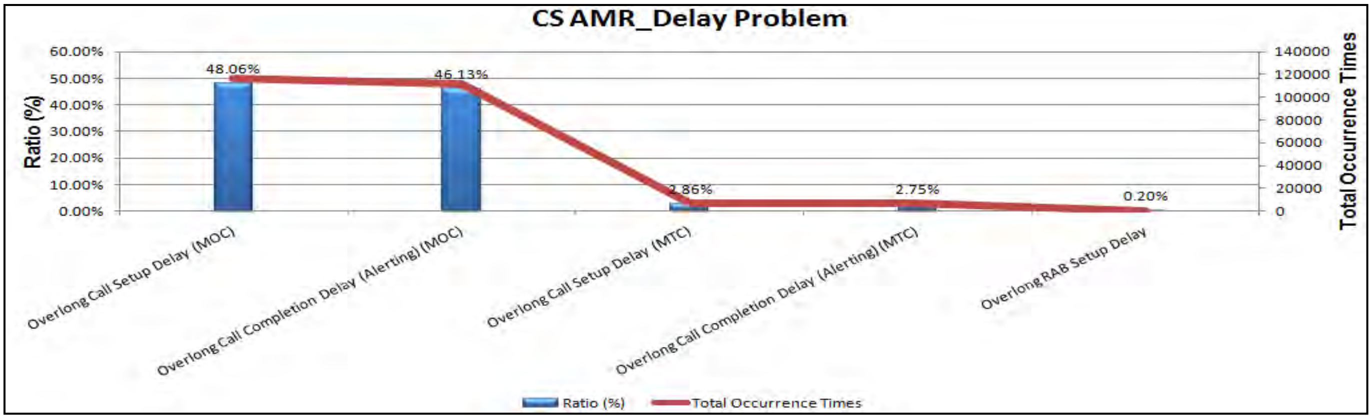


Figure 5.17 Addis Ababa UMTS network delay problem analysis report.

5.1.3.5 CS AMR Short Call Problem

Figure 5.18 shows Addis Ababa UMTS network short call problem analysis results. If the first RAB is used for the AMR service in the CS domain, the actual call duration [call duration = time stamp of the disconnect message -time stamp of the connect acknowledgment message] of the mobile originated call is shorter than the short call threshold. It can set the exception threshold by using the source data subscription function.

Over short RAB duration is the RAB duration delay exceeds the predefined threshold when a subscriber attempts to access the network. We can obtain the RAB duration by using the following formula:

RAB duration (s) = RAB release time – RAB request complete time.

Short call (MOC) means if the first RAB is used for the service in the CS domain, a call lasts shortly after a mobile-originated subscriber accesses the network and successfully initiates the call, the probable causes are one-way audio or cross talk.

Short call (MTC) means if the first RAB is used for the service in the CS domain, if a call lasts shortly after a mobile-terminated subscriber accesses the network and successfully initiates the call, the probable causes are one-way audio, cross talk, and disturbance call.

The actual call duration (call duration = time stamp of the disconnect message – time stamp of the connect acknowledge message) of the mobile originated or terminated subscriber is shorter than the threshold for mobile originated or terminated short call. From analysis result, in general most of the short call problem is due to over short RAB duration, this is due to resource congestion and weak coverage.

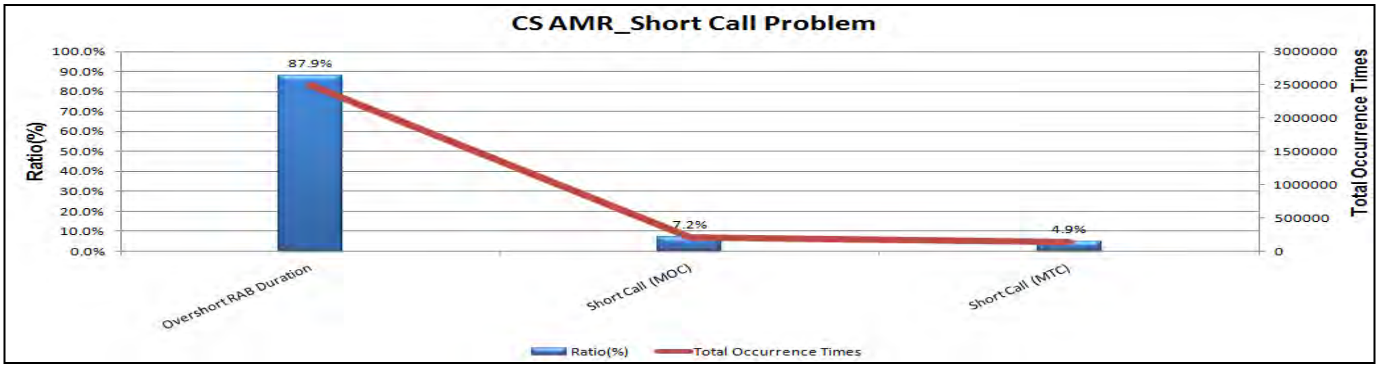


Figure 5.18 Addis Ababa UMTS network short call problem analysis report.

5.1.3.6 CS AMR QoS Problem

This parameter indicates the voice quality during a call. **UL one way audio** is a proportion of successful call setups where one-way audio occurs in the uplink in all successful call setups. Proportion of successful call setups where one-way audio occurs in the uplink in all successful call setups = number of times that one-way audio occurs in the uplink/number of successful call setups (Connect Ack) x 100%. **Downlink one way audio** is a proportion of successful call setups where one-way audio occurs in the downlink in all successful call setups.

Proportion of successful call setups where one-way audio occurs in the downlink in all successful call setups = number of times that one-way audio occurs in the downlink/number of successful call setups (Connect Ack) x 100%. **High ratio of bad SQI** is a service quality problem indicating a high proportion of services with bad SQI. The proportion of services with bad SQI is the number of successful calls, the SQIs of which are bad, against all successful calls initiated by subscribers. **Speech noise** is a proportion of calls with static problems.

Proportion of calls with static problems (%) = number of calls with static problems/number of connected calls (Connect Ack) x 100%. Figure 5.19 shows Addis Ababa UMTS network QoS problem analysis results. From analysis result, in general most of the QoS problem is due to UL and DL one way audio, this is due to UL and DL resource congestion and weak coverage.

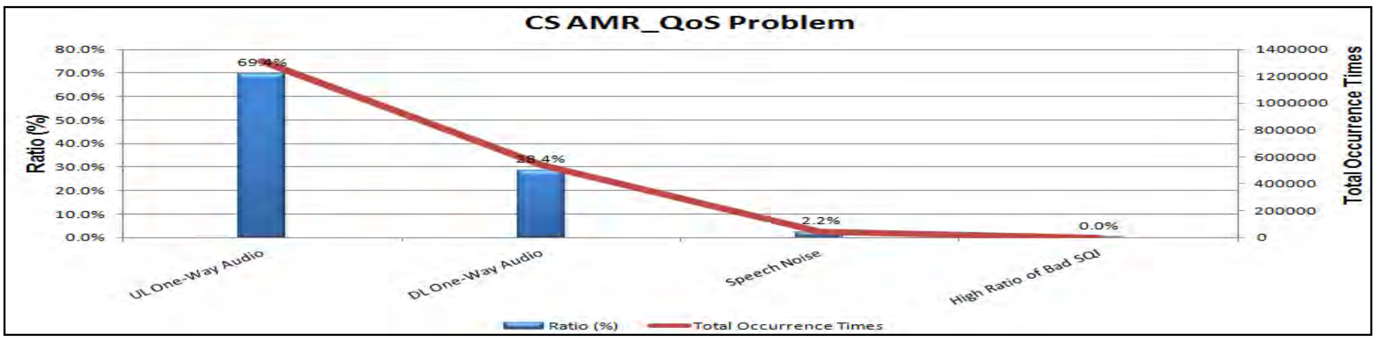


Figure 5.19 Addis Ababa UMTS network QoS problem analysis report.

5.1.4 Signal Quality Indicator Data Analysis

Signal quality indicator (SQI) indicates the number of valid mean opinion score reports for the voice service on the uplink. The SQI statistics are collected by the RNC. When the statistical period is less than 9.6 seconds, the value of SQI is 0. SQI reflects the voice quality during the calling process. Its value ranges from 0 to 500. The greater the value is, the better the voice quality is. The RNC can statistics based on the average SQI value and divides signal quality into different levels according to SQI values [14,15]:

Table 5.1 shows the range of SQI.

Excellent	Good	Accept	Poor	Bad
400<=SQI<=500	300<=SQ<400	200<=SQI<300	100<=SQI<200	0<=SQI<100

Table 5.1 Range of SQI.

The SQI equals to the score for a voice service. Good indicates excellent. Accept indicates qualified. Bad indicates not qualified. Average SQI indicates the average SQI value during the calling process. The segment intervals of levels can be modified by running the SET USQICOUNT command on the RNC. The threshold for each level is determined by the RNC configuration, the default configuration is as follows: the score lower than 200 is bad and the score higher than 313 is Good [14]. Figure 5.20 shows Addis Ababa UMTS network voice quality indicator analysis results, for March 2016. From analysis result, in general, in average the VQI of the city's UMTS network is 310.86, which is satisfactory.

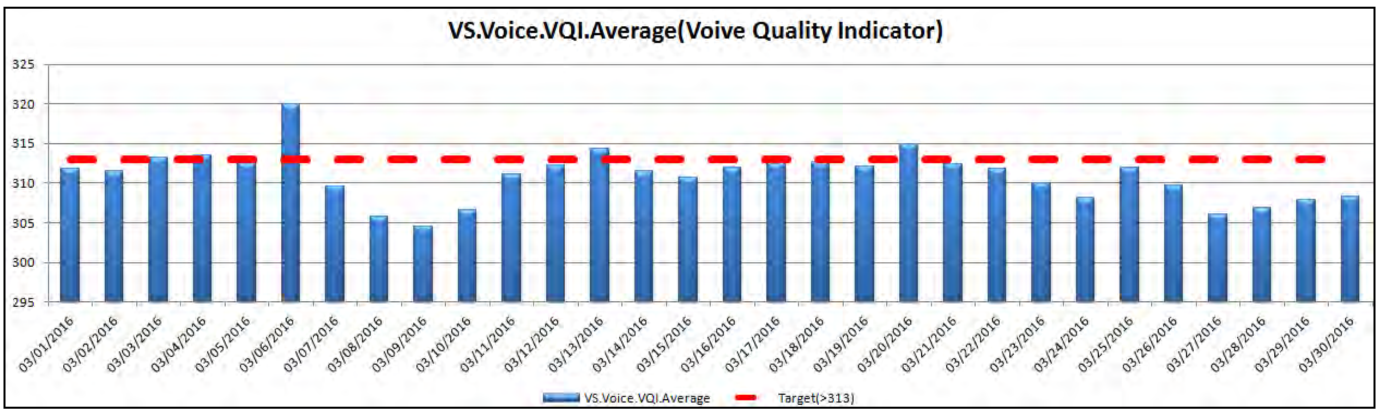


Figure 5.20 Addis Ababa UMTS network voice quality indicator analysis report.

5.1.5 Overshoot (TP size) Data Analysis

Transmit propagation delay (TP) size indicates the propagation delay of the signaling access during a call by service type. Figure 5.21 shows Addis Ababa UMTS network overshoot analysis results, for April 2016. From analysis result, in general there is overshooting in the RNC_02.

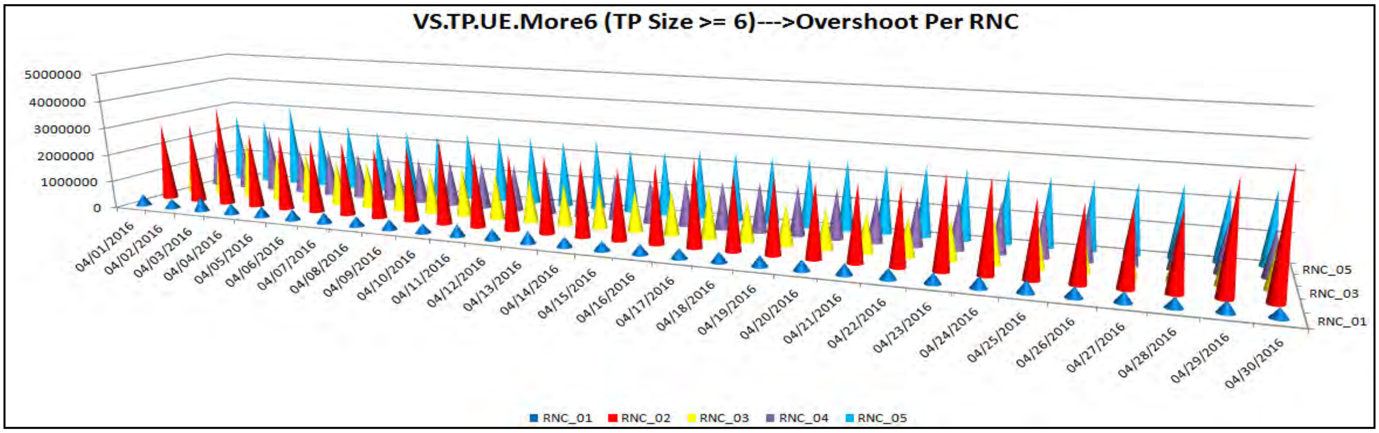


Figure 5.21 Addis Ababa UMTS network overshoot analysis report.

5.2 User Plane System Data Collection and Analysis Results

The main objective of the drive test was to investigate voice performance of Addis Ababa UMTS network. Drive test is normally conducted to investigate network problem associated to poor coverage and quality. Accordingly drive test has been done for evaluating coverage and quality of voice in Addis Ababa UMTS network, March 2016.

The DT was done by using test tools called Nemo handy, Nemo outdoor, laptop, GPS, engineering parameters and DT routes to gathers data, which are later, analyzed using standard tool actix software to give a picture of the coverage footprint of the Addis Ababa UMTS network. Typically coverage is identified by the coverage performance indicator (RSCP) and quality is identified by the quality performance indicator (Ec/Io), which will show the signal strength.

5.2.1 Voice DT Testing Information

The drive test is performed according to the need and the types of test calls, which are the same that the network supports. The calls are voice short call and MOS. Everything depends on the UMTS technology. The mobile configured in the collecting software (nemo outdoor), performing calls for a specific number (ethio telecom automatic server 118) from time to time. Short calls should last the average of a user call: a good reference value is 180 seconds. Server to check whether the calls are being established and successfully completed (being a good way to also check the network setup time).

And the Nemo handy is used for MOS to measure changes or degradation in the quality of the voice connection. Table 5.2: shows the voice DT test information.

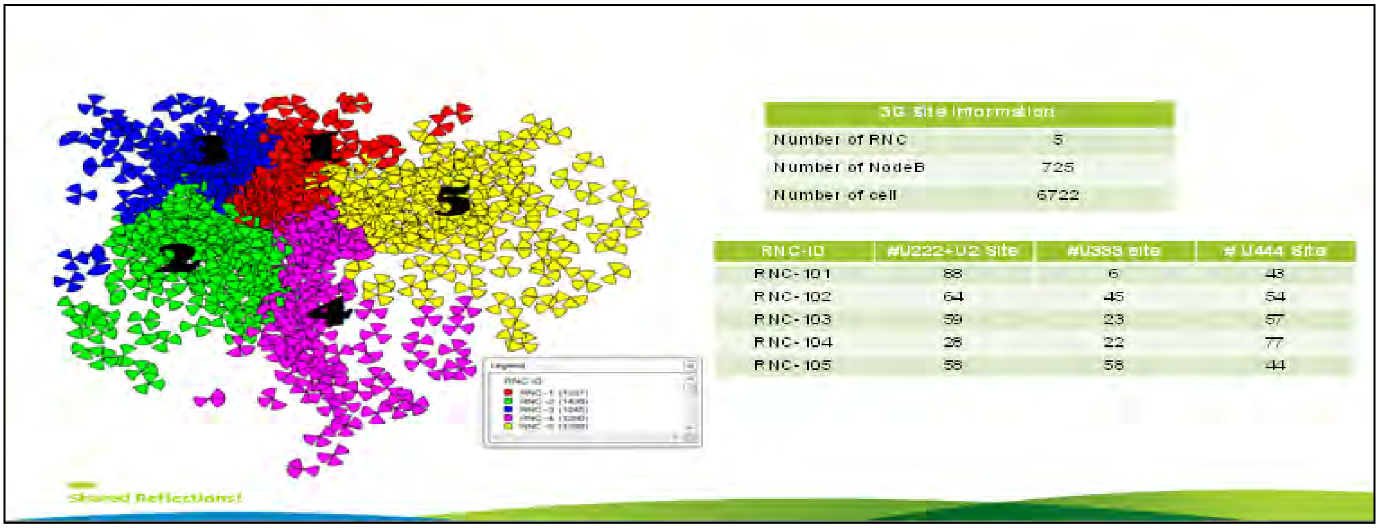
Voice Test Details	Specifications
RAN System	UMTS 2100MHz
Voice Test Algorithm	
DT type	UMTS Short Call
Voice Quality Measurement	MOS
Test Procedure	
Uu_ActiveSet_RSCP_0	Coverage performance of the service areas are checked
Uu_ActiveSet_EcNo_0	Quality performance of the service areas are checked

Table 5.2 the voice DT test information.

5.2.2 Addis Ababa UMTS Network Information

Addis Ababa UMTS network has 5 RNCs, 725 Node Bs or sites and around 6722 cells.

Figure 5.22 shows Addis Ababa UMTS RNCs geographic location information.



3G Site information	
Number of RNC	5
Number of NodeB	725
Number of cell	6722

RNC-ID	#U222+U2 Site	#U333 site	# U444 Site
RNC-101	88	6	43
RNC-102	64	45	54
RNC-103	59	23	57
RNC-104	28	22	77
RNC-105	58	58	44

Figure 5.22 Addis Ababa UMTS RNCs information.

5.2.3 Drive Test Measurement Route for Addis Ababa

DT routes are predefined pathways by using MapInfo. These routes should cover the relevant areas of the network, such as large customers and companies, bridges and major avenues, etc. The best way to define drive test routes is drawing the same observing all existing conditions. In other words, plot the routes so that is really what we'll find in the road. Figure 5.23 shows Addis Ababa UMTS network DT routes. DT routes cover 300 to 350 kilometers.

Figure 5.23 Addis Ababa UMTS network DT routes.

5.2.4 Accessibility, Mobility and Retainability UMTS DT Summary

The UE in the survey kit is configured to make 180 seconds calls so that accessibility and retainability could be assessed. The inter call gap for all calls was 10 seconds to allow the UE to stabilise in idle mode. A call on UMTS was allowed to continue until the duration had elapsed and the call was then terminated as per normal. Again the call is established on the UMTS network. Accessibility is a measure of subscriber experience associated with the process of obtaining access to the network resources. It consists of access failures and blocks associated with both RF and core network.

Mobility is the process of transferring an ongoing call from one channel connected to the core network to another channel. Retainability is a measure of subscriber experience associated with the continued use of network resources that have been granted. It consists of radio link drops or abnormal network-initiated releases. Table 5.3: shows the UMTS DT call statics summary.

CS domain overview	Count
Call attempt	422
Call connected	383
Call disconnect	379
Call dropped	39
Call drop rate	9%
Call setup success rate	91%
Call completion rate	99%

Inter-system handover statistics	Count
UMTS to GSM handover attempt	42
UMTS to GSM handover failure	0
UMTS to GSM handover success	42
UMTS to GSM handover success ratio	100%
Soft handover statistics	
Soft handover attempt	11321
Soft handover failure	0
Soft handover success	11321
Soft handover success ratio	100%

RRC connection statistics	Count
RRC connection attempt	400
RRC connection attempt failure	8
RRC connected	392
RRC disconnected	389
RRC connection dropped	3
RRC connection success rate	98%
RRC connection completion rate	99%

Color Legends	
Yellow	Below Target (Low)
Green	Already Meeting Target
Red	Below Target (High)

Table 5.3 UMTS DT call statics summary.

5.2.5 Addis Ababa UMTS DT Coverage Analysis

In the definition of network coverage, the requirements of effective coverage for a certain sampling point is that its Uu_ActiveSet_ RSCP_0 should be better than the specified threshold (RSCP > -90dBm). Figure 5.24 shows Addis Ababa UMTS network DT coverage performance analysis. From DT road analysis result, in general 97% and 3% of the city's road coverage was good and poor respectively, where expected good coverage threshold is greater than or equal to 95%. The DT road analysis result shows that the central Addis Ababa's coverage is good and the edge of Addis Ababa's coverage is poor.

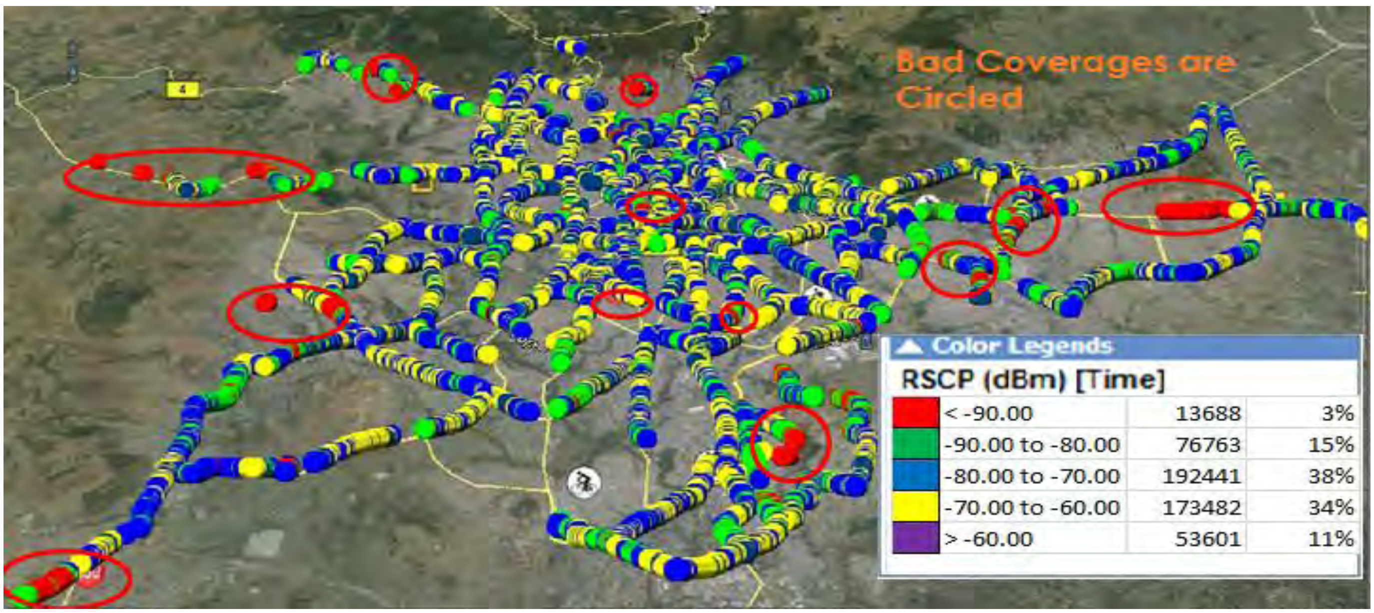


Figure 5.24 Addis Ababa UMTS network DT coverage performance.

5.2.6 Addis Ababa UMTS DT Quality Analysis

In the definition of network quality, the requirements of effective quality for a certain sampling point is that its Uu_ActiveSet_EcNo_0 should be better than the specified threshold. Figure 5.25 shows Addis Ababa UMTS network DT quality performance summary.

From DT road analysis result, in general 93% and 7% of the city's road quality of service was good and poor respectively, where expected quality of service threshold is greater than or equal to 93%. The DT road analysis result shows that the central Addis Ababa's quality of service seems good and the edge of Addis Ababa's quality of service is poor.

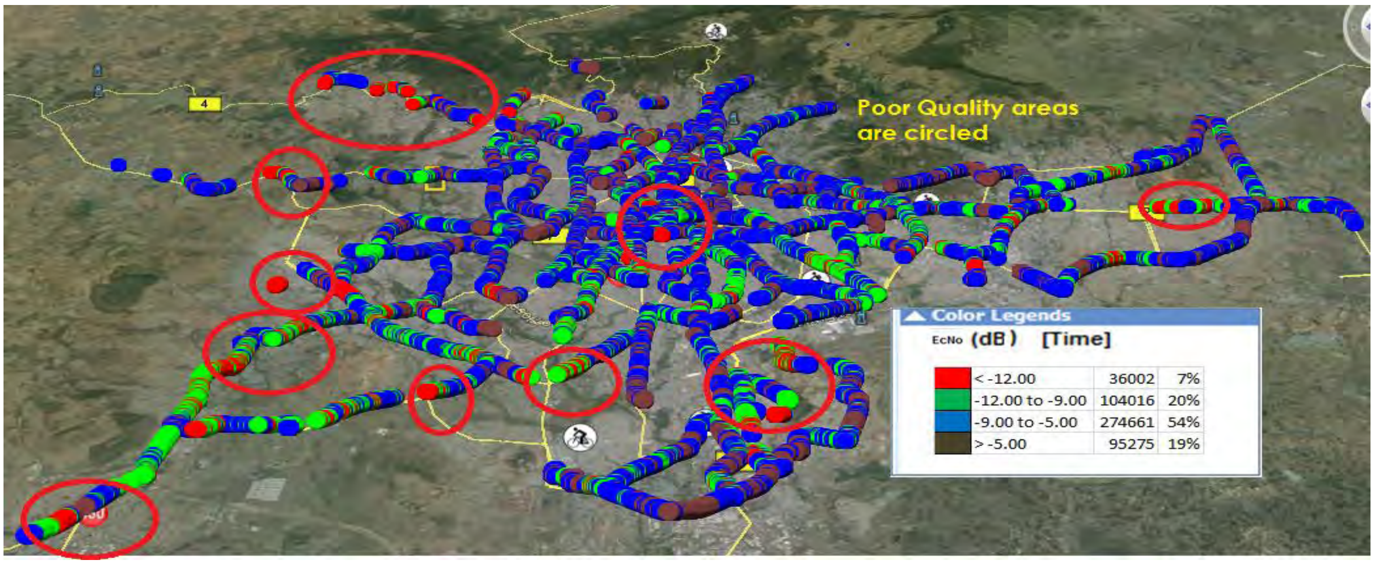


Figure 5.25 Addis Ababa UMTS network DT quality performance.

5.2.7 Addis Ababa UMTS DT Call Drop Analysis

Uu_CallDroped can be defined from the following two sides.

UE side: call drops refer to call releases caused by not normal clearing, not normal or unspecified when the message on the air interface satisfying any of the following three conditions:

- The UE receives any BCH information (system information).
- The call is released for not normal and the UE receives the RRC release information.

- The UE receives Call Control (CC) disconnect, CC release complete and CC release information.

RNC side: Call drops refer to call releases when the RNC has sent the Iu release request to the CN through the Iu interface, or when the RNC has sent the RAB release request information to the CN through the user panel.

Figure 5.26 shows Addis Ababa UMTS network DT call drop summary. The DT road analysis result shows that the central Addis Ababa's voice call drop distribution is high and the edge of Addis Ababa's voice call drop distribution is low.

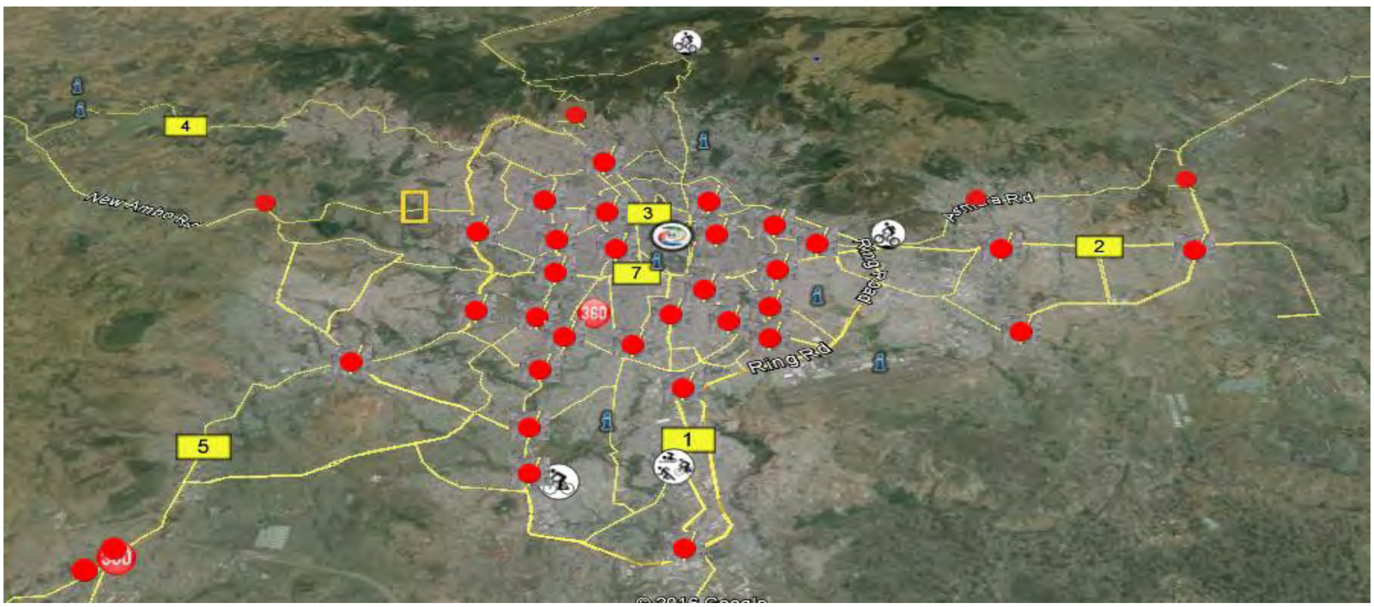


Figure 5.26 Addis Ababa UMTS network DT call drop events.

5.2.8 Addis Ababa UMTS Pilot Pollution Analysis

Uu_Pilot pollution typically means there are many signals with close Ec/Io values or there are strong signals unfamiliar to the planning design. Figure 5.27 shows Addis Ababa UMTS network pilot pollution summary. The DT road analysis result shows that the central Addis Ababa's pilot pollution is high and the edge of Addis Ababa's pilot pollution is low.

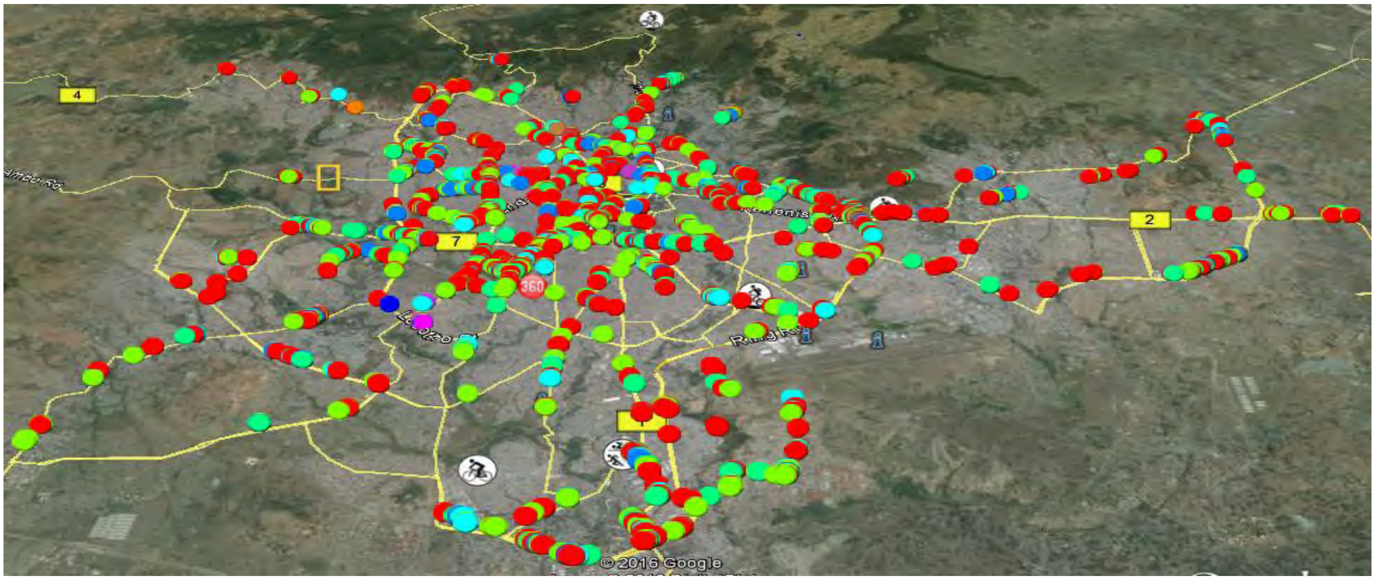


Figure 5.27 Addis Ababa UMTS network pilot pollution events.

5.2.9 Addis Ababa UMTS Voice MOS Analysis

MOS is used to provide a numerical measure of the quality of human speech at the destination end of the circuit.

This method of voice quality test has been in use for decades to obtain human user's view of the quality of the network. Figure 5.28 shows Addis Ababa UMTS network mean opinion score summary. From analysis result, in general, in average the MOS of the DT road of the city's UMTS network is 3.32, which is satisfactory.

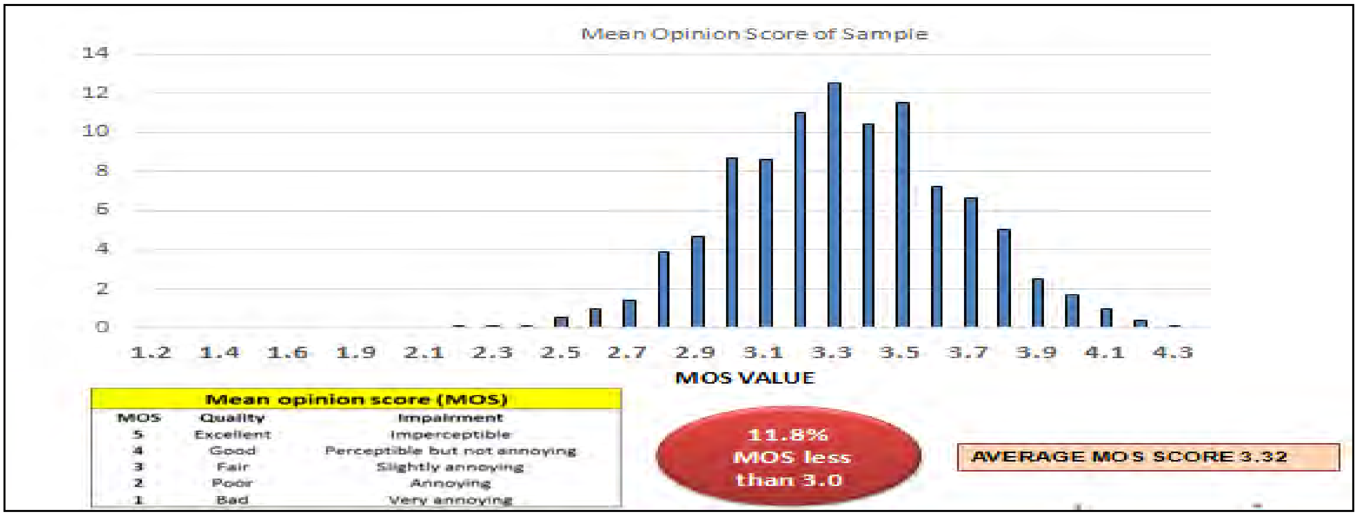


Figure 5.28 Addis Ababa UMTS network MOS performance.

5.2.10 Voice QoS Simulation and Analysis Result Summary

In our case, the voice performance measurement is collected from the average value of a certain QoS parameter after the application is run for a certain period of time in the UMTS network. These measured values are used with the acceptable QoS parameter values which are defined earlier to calculate the voice QoS performance in the UMTS network. Then this voice QoS performance value is used to measure the network's QoS performance. This approach also unfolds a way for bridging network QoS with QoE. User satisfactions of a network are closely related to the application performance. For instance, a voice based service with a lower delay, normally results in higher user satisfactions. To find out the performance of a particular network, the performances of application-based traffic flows present in that network are analyzed.

For each QoS indicator parameter **i** under **n** service, there is an upper and lower bound value $\mathbf{Um_i}$ and $\mathbf{Lm_i}$.

These upper and lower bound values for the specific QoS indicator and the average $\boldsymbol{\tau}$ for the same QoS indicator value derived from voice flow in the network are put into equation below to calculate the normalization value for each QoS indicator [47].

$$\mathbf{m_{i,n}} = \begin{cases} 0, & \boldsymbol{\tau} \geq \mathbf{Umi} \\ 1, & \boldsymbol{\tau} \leq \mathbf{Lmi} \\ \dfrac{\mathbf{Umi} - \boldsymbol{\tau}}{\mathbf{Umi} - \mathbf{Lmi}}, & \mathbf{Lm} < \boldsymbol{\tau} < \mathbf{Umi} \end{cases} \qquad (13)$$

Where **n** is Voice service and $\mathbf{m_i}$ is measurement of QoS indicator parameters.

After defining the acceptable value for each QoS indicator, a weight, **w** is assigned to each indicator. This weight depends on the relevant importance of a particular QoS indicator to the voice performance. Upon calculation of normalized values for each of the indicators, an overall QoS value corresponding to voice service is calculated using [47]:

$$\mathbf{QoS_n} = \sum_{i=1}^{q} \mathbf{m_{i,n} w_i} \qquad \mathbf{(14)}$$

Let as calculate the average $\boldsymbol{\tau}$ for the same QoS indicator of both control and user plane systems one by one and map it to MOS (**Um**$_i$ **= 5 and Lm**$_i$ = 3, i.e, in percent 100% and 93%).

$\boldsymbol{\tau}_{\text{i, voice}}$ **= (good quality (Ec/Io) % X Um**$_i$ **or Lm**$_i$**)/(100% or 93%)**.

- $\boldsymbol{\tau}_{\text{network geographic observation, voice}}$ = (72.5%X3)/93% = 2.339 (in MOS)
- $\boldsymbol{\tau}_{\text{network quality analysis, voice}}$ = (89%X3)/93% = 2.87 (in MOS)
- $\boldsymbol{\tau}_{\text{VQI, voice}}$ = (310.86X3)/313 = 2.98 (in MOS)
- $\boldsymbol{\tau}_{\text{quality analysis from DT, voice}}$ = (93%X3)/93% = 3 (in MOS)
- $\boldsymbol{\tau}_{\text{MOS from DT, voice}}$ = 3.32

Therefore, $\boldsymbol{\tau}_{\text{average, voice}}$ = (2.339+2.87+2.98+3+3.32)/5 = 2.902

Let as also calculate measurement (mi) of QoS indicator parameters of both control and user plane systems one by one.

- $\boldsymbol{\tau}_{\text{network geographic observation, voice}}$ = 2.339 (i.e. $\boldsymbol{\tau} < \mathbf{Lm_i}$, $m_i = 1$)
- $\boldsymbol{\tau}_{\text{network quality analysis, voice}}$ = 2.87 (i.e. $\boldsymbol{\tau} < \mathbf{Lm_i}$, $m_i = 1$)
- $\boldsymbol{\tau}_{\text{VQI, voice}}$ = 2.98 (i.e. $\boldsymbol{\tau} < \mathbf{Lm_i}$, $m_i = 1$)
- $\boldsymbol{\tau}_{\text{quality analysis from DT, voice}}$ = 3 (i.e. $\boldsymbol{\tau} = \mathbf{Lm_i}$, $m_i = 1$)
- $\boldsymbol{\tau}_{\text{MOS from DT, voice}}$ = 3.32 (i.e. $\mathbf{Um_i} > \boldsymbol{\tau} > \mathbf{Lm_i}$, $m_i = 0.84$)

Finally let as calculate overall QoS value corresponding to voice service. The weight **w** assigned to MOS is 0.6.

$$\mathbf{QoS_n} = \sum_{i=1}^{q} \mathbf{m_{i,n} w_i} = (\mathbf{1 + 1 + 1 + 1 + 0.84}) \times \mathbf{0.6} = \mathbf{2.904}$$

The overall QoS calculated value is 2.904 in MOS, this indicates the City's voice QoS target of UMTS network is not meet (target > 3).

Chapter 6

6. Conclusions and Recommendations

6.1 Conclusions

ethio telecom is vested with the responsibility of realizing the telecommunication sub sector expansion plan of the successive Ethiopian Government's Growth and Transformation Plans (GTPs). It has carried out intensive expansion work mainly on mobile network to extend the coverage to 85% and scale up the capacity to more than 60 million subscribers as part of the country GTP II and introduce the new technologies including 4G network.

For Addis Ababa city's case, it has carried out intensive expansion work to extend the coverage of UMTS network to 100% and scale up the capacity to more than 1.5 million subscribers. The current UMTS network infrastructure deployed in the Addis Ababa city, which is solely managed by ethio telecom, is undergoing major expansions in the last 4 years and resulted in a tangible improvement of coverage and quality performance. However, there are complaints from subscribers (end users) from various parts of the city.

The main motivation of this thesis work is to evaluate the existing QoS performance of voice transmission in UMTS network in case of Addis Ababa. The evaluation is made by analyzing collected real data from control plane and user plane systems.

The analysis results show that, in general, there is some disparities between the ethio telecom targets and analysis results, which indicating the need to further improve the network's QoS.

Based on the analysis result and literature review, integration means is proposed to improve the QoS of voice in UMTS network.

6.2 Recommendations

To improve the quality of voice transmission in UMTS network, here are the lists of the recommended possible solutions of the works of this thesis research:

- Implementation of QoS manager in different levels of network;
- Appropriate resource allocation in the network;
- In urban area like Addis Ababa city the inter node B distance or inter system distance (ISD) must be less than or equal 500 meters (micro Site);
- In area where, there are no high buildings, the coverage and quality are bad and also the traffic is high, installing the new site will be the solution;
- In area where, there are no high buildings, the coverage is good, the quality is bad and also the traffic is high, the RF optimization (i.e. CE license addition, power license addition, site expansion, parameters adjustment, etc) will be the solution;
- In area where, there are high buildings, the coverage is good, the quality is bad and also the traffic is high, adding the boosters or repeaters and installing indoor BTS will be the solution;
- Finally, organize the Addis Ababa UMTS network in a hierarchical way (i.e. femto cells, pico cells, microcells, macro cells and global cells served by satellites) is recommended.

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