METHODICAL EVALUATION OF QUALITY OF SERVICE FOR HETEROGENEOUS NETWORKS

by

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for the degree of
Doctor of Philosophy

School of Computing, Engineering and Mathematics
Western Sydney University

March 2015
Declaration

I declare that to the best of my knowledge the work described in this thesis is, except where otherwise stated, entirely my own work and has not been submitted for a degree at this or any other university.

______________________________

Farnaz Farid
March, 2015
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Abstract

Heterogeneous wireless networks expand the network capacity and coverage by leveraging the network architecture and resources in a dynamic fashion. However, due to the presence of different communication technologies, the Quality of Service (QoS) evaluation, management, and monitoring of these networks are very challenging tasks. Each communication technology has its own characteristics while the applications utilising them have their own QoS requirements. Most current methods are based on analysing the QoS of each application or access network separately. However, these methods do not combine the performance of all the applications and the radio access networks while reporting the QoS of the overall configuration. Therefore, it is hard to get any aggregate performance results using these methods.

To fill this gap, in this thesis, a methodical approach is adapted for the QoS analysis of these types of networks. At first, the approach uses a simple fixed weight-based method, and then moves to a more complex dynamic weight-based method and in the end integrates the concepts of fuzzy logic. The proposed methods consider the significance of QoS-related parameters, the available network-based applications, and the available Radio Access Networks (RANs) to characterise the network performance with a set of three integrated QoS metrics. The first metric denotes the performance of each available application on the network, the second one represents the performance of each active RAN on the network, and the third one characterises the QoS level of the entire network configuration.

Using the fixed weight-based method, the weights for the QoS-related parameters, applications, and RANs are defined based on their significance relevant to a given situation. Then, the dynamic weight-based method determines the weights of these entities dynamically by incorporating the changing circumstances of the network. Although, the dynamic weight-based method can account for the active changes of the network, it has some limitations. For instance, it cannot capture the underlying uncertainty of network dynamics. To overcome these limitations, the concepts of fuzzy logic are adapted for further enhancement of the QoS evaluations.
This results in a methodical approach that is particularly useful for QoS management, and monitoring of complex networks, consisting of different configuration settings, various network technologies supporting different applications. Therefore, it can quantify the performance of heterogeneous network-based service models by a unified QoS metric. This approach is also useful when some specific network-based service models are re-deployed from one region to another region. Each area has its own service requirements and technology availability. As a result, the service model, which demonstrates better performance in one context, is not necessarily going to have the same outcomes in another region. In such circumstances, it is possible to compare the resultant QoS level of any network-based service model with the expected QoS level by applying this approach.

To investigate the efficiency of the designed approach, a diverse range of simulation studies utilising different heterogeneous network-based service models are carried out. The simulation results indicate that the approach in this work facilitates better management and monitoring of heterogeneous network configurations and applications utilising them. The simulation studies also show that using the unified metrics, it is possible to choose a suitable network configuration for a particular application or service from among the range of available network configurations under investigation and classify them for their suitability to provide some specific services. Overall, the outcomes from the simulation results analysis clearly demonstrate that the proposed methods can significantly improve the QoS analysis of the heterogeneous networks.
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## Acronyms

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<th>Acronym</th>
<th>Definition</th>
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<tbody>
<tr>
<td>3GPP</td>
<td>Third Generation Partnership Programme</td>
</tr>
<tr>
<td>1G</td>
<td>First generation</td>
</tr>
<tr>
<td>2G</td>
<td>Second generation</td>
</tr>
<tr>
<td>3G</td>
<td>Third Generation</td>
</tr>
<tr>
<td>4G</td>
<td>Fourth Generation</td>
</tr>
<tr>
<td>AHP</td>
<td>Analytical Hierarchy Process</td>
</tr>
<tr>
<td>ANS</td>
<td>Access Network Selection</td>
</tr>
<tr>
<td>ASN</td>
<td>Access Service Network</td>
</tr>
<tr>
<td>ASN-GW</td>
<td>ASN Gateway</td>
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<tr>
<td>AMPS</td>
<td>Advanced Mobile Phone System</td>
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<tr>
<td>AMR</td>
<td>Adaptive Multi-Rate</td>
</tr>
<tr>
<td>APN</td>
<td>Access Point Name</td>
</tr>
<tr>
<td>AM</td>
<td>Acknowledged Mode</td>
</tr>
<tr>
<td>BS</td>
<td>Base Station</td>
</tr>
<tr>
<td>BAS</td>
<td>Basic Access Signal</td>
</tr>
<tr>
<td>BE</td>
<td>Best Effort</td>
</tr>
<tr>
<td>CA</td>
<td>Carrier Aggregation</td>
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<tr>
<td>CoS</td>
<td>Class of Service</td>
</tr>
<tr>
<td>CDMA</td>
<td>Code Division Multiple Access</td>
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<td>CDMA2000</td>
<td>Code Division Multiple Access 2000</td>
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<tr>
<td>CCN</td>
<td>Common Core Network</td>
</tr>
<tr>
<td>CPC</td>
<td>Continuous Packet Connectivity</td>
</tr>
<tr>
<td>CFM</td>
<td>Context Fusion Module</td>
</tr>
<tr>
<td>CS</td>
<td>Circuit-Switched</td>
</tr>
<tr>
<td>CPC</td>
<td>Continuous Packet Connectivity</td>
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<tr>
<td>CSN</td>
<td>Connectivity Service Network</td>
</tr>
<tr>
<td>CN</td>
<td>Core network</td>
</tr>
<tr>
<td>DC-HSPA+</td>
<td>Dual Carrier HSPA Evolved</td>
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<tr>
<td>DCF</td>
<td>Distributed Coordination Function</td>
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<tr>
<td>DIFS</td>
<td>DCF Interframe Space</td>
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<tr>
<td>Abbreviation</td>
<td>Description</td>
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<tr>
<td>DS3</td>
<td>Digital Signal 3</td>
</tr>
<tr>
<td>DSSS</td>
<td>Direct-Sequence Spread Spectrum</td>
</tr>
<tr>
<td>DSL</td>
<td>Digital subscriber line</td>
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<tr>
<td>DFS</td>
<td>Dynamic Frequency Selection</td>
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<tr>
<td>DHCP</td>
<td>Dynamic Host Control Protocol</td>
</tr>
<tr>
<td>DFTS</td>
<td>Discrete Fourier Transform Spread</td>
</tr>
<tr>
<td>DFTS-OFDM</td>
<td>DFTS Orthogonal Frequency Division Multiplexing</td>
</tr>
<tr>
<td>DWQEM</td>
<td>Dynamic Weight based QoS Evaluation Method</td>
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<tr>
<td>ELECTRE</td>
<td>Elimination and Choice Translating Reality</td>
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<tr>
<td>EDGE</td>
<td>Enhanced Data rates for GSM Evolution</td>
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<tr>
<td>EPC</td>
<td>Evolved Packet Core</td>
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<td>EPS</td>
<td>Evolved Packet System</td>
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<tr>
<td>EV-DO</td>
<td>Evolution-Data Optimized</td>
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<tr>
<td>EV-DV</td>
<td>Evolution- Data and Voice</td>
</tr>
<tr>
<td>E-UTRAN</td>
<td>Evolved UMTS Terrestrial Radio Access Network</td>
</tr>
<tr>
<td>erPS</td>
<td>extended real-time Polling Service</td>
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<tr>
<td>EAP</td>
<td>Extensible Authentication Protocol</td>
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<tr>
<td>E2Edelay</td>
<td>End-to-End Delay</td>
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<tr>
<td>FAHP</td>
<td>Fuzzy Analytical Hierarchy Process</td>
</tr>
<tr>
<td>FDD</td>
<td>Frequency Division Duplex</td>
</tr>
<tr>
<td>FLC</td>
<td>Fuzzy Logic Controller</td>
</tr>
<tr>
<td>FIS</td>
<td>Fuzzy Inference System</td>
</tr>
<tr>
<td>FTP</td>
<td>File Transfer Protocol</td>
</tr>
<tr>
<td>FL</td>
<td>Fuzzy Logic</td>
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<td>FWQEM</td>
<td>Fixed Weight QoS Evaluation Method</td>
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<tr>
<td>FR</td>
<td>Frame Rate</td>
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<tr>
<td>GBR</td>
<td>Guaranteed Bit Rate</td>
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<tr>
<td>GGSNs</td>
<td>Gateway GPRS Support Nodes</td>
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<tr>
<td>GMSK</td>
<td>Gaussian Minimum Shift Keying</td>
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<tr>
<td>GPRS</td>
<td>General Packet Radio Service</td>
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<tr>
<td>GAs</td>
<td>Genetic Algorithms</td>
</tr>
<tr>
<td>GSA</td>
<td>Global mobile Suppliers Association</td>
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</tbody>
</table>
GSM  Global System for Mobile Communications
GSM FER  GSM Frame Erasure Rate
GoS  Grade of Service
GoP  Group of Pictures
GRA  Grey Relational Analysis
HSDPA  High-Speed Downlink Packet Access
HWNs  Heterogeneous Wireless Networks
HSUPA  High-Speed Uplink Packet Access
HDMI  High-Definition Multimedia Interface
HA  Home Agent
HRM  Handover and Resource Manager
HSPA+  HSPA Evolved
HSS  Home Subscriber Server
iDEN  Integrated Digital Enhanced Network
ITU  International Telecommunication Union
IEEE  The Institute of Electrical and Electronics Engineers
IP  Internet Protocol
IPTV  Internet Protocol Television
IMS  IP Multimedia Subsystem
JTACS  Japanese Total Access Communication System
JAC  Joint Admission Control
JSC  Joint Scheduling Control
LANs  Local Area Networks
LTE  Long Term Evolution
MATLAB  Matrix Laboratory
MAC  Media Access Control
ME  Mobile Equipment
MOS  Mean Opinion Score
msec  Milliseconds
MIP  Mobile Internet Protocol
MMR  Mobile Multi-hop Relay
MSCs  Mobile Switching Centers
MT  Mobile Terminal
MME  Mobility Management Entity
MRW  Move Receiving Window
MADM Multi-attribute Decision Making
MCDM Multi-criteria Decision Making
MMS Multimedia Messaging Service
MIMO Multiple-Input Multiple-Output
NDM Network Discovery Module
nrtPS non-real-time Polling Service
NMT Nordic Mobile Telephone
NLoS Non-Line of Sight
non-GBR non-Guaranteed Bit Rate
PS  Packet-Switched
PDN Packet Data Network
PDG Packet Data Gateway
PingER Ping End-to-end Reporting
PCRF Policy Control and Charging Rules Function
PLMN Public Land Mobile Network
PoC Push-to-Talk over Cellular
PDN-GW Packet Data Network Gateway
PHY Physical Characteristics
OFDM Orthogonal Frequency-Division Multiplexing
OPNET Optimized Network Engineering Tool
QoE Quality of Experience
QoS Quality of Service
QAM Quadrature Amplitude Modulation
RABs Radio Access Bearers
RNC Radio Network Controller
RF  Radio Frequency
rtPS real-time Polling Service
RAN Radio Access Network
RTT Radio Transmission Technology
RAS Rural Area Segment
RPs Reference Points
<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Full Form</th>
</tr>
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<tbody>
<tr>
<td>RLC</td>
<td>Radio Link Layer</td>
</tr>
<tr>
<td>RSS</td>
<td>Received Signal Strength</td>
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<tr>
<td>RM</td>
<td>Resource Manager</td>
</tr>
<tr>
<td>RTMM</td>
<td>Real-time Monitoring Module</td>
</tr>
<tr>
<td>RTP</td>
<td>Real-time Protocol</td>
</tr>
<tr>
<td>RNS</td>
<td>Radio Network Subsystem</td>
</tr>
<tr>
<td>S-GW</td>
<td>Serving Gateway</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SS</td>
<td>Subscriber Station</td>
</tr>
<tr>
<td>TDD</td>
<td>Time Division Duplex</td>
</tr>
<tr>
<td>TDMA</td>
<td>Time Division Multiple Access</td>
</tr>
<tr>
<td>TACS</td>
<td>Total Access Communication System</td>
</tr>
<tr>
<td>TOPSIS</td>
<td>Total Order Preference by Similarity to the Ideal Solution</td>
</tr>
<tr>
<td>TFN</td>
<td>Triangular Fuzzy Numbers</td>
</tr>
<tr>
<td>TD-SCDMA</td>
<td>Time Division - Synchronous Code Division Multiple Access</td>
</tr>
<tr>
<td>TPC</td>
<td>Transmission Power Control</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>TTI</td>
<td>Transfer Time Interval</td>
</tr>
<tr>
<td>TM</td>
<td>Transparent Mode</td>
</tr>
<tr>
<td>USIM</td>
<td>UMTS Service Identity Module</td>
</tr>
<tr>
<td>UM</td>
<td>Unacknowledged Mode</td>
</tr>
<tr>
<td>UMTS</td>
<td>Universal Mobile Telecommunications System</td>
</tr>
<tr>
<td>UTRAN</td>
<td>UMTS Terrestrial Access Network</td>
</tr>
<tr>
<td>USB</td>
<td>Universal Serial Bus</td>
</tr>
<tr>
<td>UGS</td>
<td>Unsolicited Grant Service</td>
</tr>
<tr>
<td>UE</td>
<td>User Equipment</td>
</tr>
<tr>
<td>UAS</td>
<td>Urban Area Segment</td>
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<td>VC</td>
<td>Video conferencing</td>
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<td>VHO</td>
<td>Vertical Handover</td>
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<td>VDC</td>
<td>Virtual Domain Controller</td>
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<td>VoIP</td>
<td>Voice over Internet Protocol</td>
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<tr>
<td>Abbreviation</td>
<td>Description</td>
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<tr>
<td>VS</td>
<td>Video streaming</td>
</tr>
<tr>
<td>WCDMA</td>
<td>Wideband Code Division Multiple Access</td>
</tr>
<tr>
<td>WiFi</td>
<td>Wireless Fidelity</td>
</tr>
<tr>
<td>WiMAX</td>
<td>Worldwide Interoperability for Microwave Access</td>
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<tr>
<td>WLAN</td>
<td>Wireless Local Area Network</td>
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<tr>
<td>WMAN</td>
<td>Wireless Metropolitan Area Network</td>
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<td>WWAN</td>
<td>Wireless Wide Area Network</td>
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<tr>
<td>WG</td>
<td>Working Group</td>
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<tr>
<td>WWW</td>
<td>World Wide Web</td>
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Introduction

The advancement and proliferation of modern wireless and cellular technologies have changed the way people work and communicate. By 2018, the mobile data traffic is expected to reach 15.9 exabytes per month, and 69 percent of this will consist of video traffic. There will be over 10 billion mobile-connected devices by 2018, which will exceed the world’s expected population at that time [1]. To deal with this growing number of devices and this massive increase of traffic, the wireless networks are moving towards an all-heterogeneous architecture.

A heterogeneous communication network provides transparent and self-configurable services across Wireless Local Area Networks (WLANs), Wireless Metropolitan Area Networks (WMANs), and Wireless Wide Area Networks (WWANs). Primarily, heterogeneous networks were anticipated as an integration of IEEE 802.11 WLANs and various cellular technologies, with mobile WiMAX as the major player in the middle. However, LTE-advance has added one more new technology to the picture, which would play a crucial role in this integrated architecture, forming the 4th Generation (4G) or next-generation of wireless networks. The heterogeneous wireless access, the exclusive all IP-based architecture and the advanced mobility support are the key drivers of this Generation [2].

All the technologies in a heterogeneous network pose the characteristics that complement each other [3]. 3G and 2G-based cellular communication technologies are well-known for their wide area coverage, complete mobility and roaming ability. However, traditionally these technologies offer low-bandwidth and expensive data traffic solutions [4]. In response to overcome the limitations of the conventional cellular technologies, LTE has been developed. On the other hand, WLANs provide high data rate at low-cost, but with limited coverage, whereas WiMAX delivers last mile mobile broadband access and backhaul for WLANs [5].
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The heterogeneous-based 4G wireless networks will offer a number of advantages for both users and network operators [6]. The users will benefit from the vibrant coverage and capacity, as well as an extensive number of available resources, will allow them to connect seamlessly to the best available access technology. On the other hand, the network operators will benefit from the lower cost and the efficient usage of the network resources.

1.1 Problem Statement

The communication technologies present in a heterogeneous network have different characteristics. These technologies have their offered bandwidths, coverage area, and operating frequencies. Their QoS characteristics, such as delay, throughput, and packet loss, as well as usage and implementation costs also differ from each other. As a result, the adaptation of heterogeneous network-based architecture for the provision of different applications especially multimedia applications faces significant challenges. Among these challenges, QoS-related issues such as the effective QoS evaluation, management, and monitoring still top the list [7].

Managing QoS for video or voice applications over heterogeneous networks is a challenging task. A research from Nemertes shows that the companies invest a significant amount of their budget to manage VoIP applications over these network architectures. For small enterprises, the annual costs range from $25,000, and for global enterprises this cost is around $2 million [8]. Therefore, the enterprises need to dedicate a lot of their effort to ensure service quality at every level of the network. System downtime is another challenge for businesses, which could often happen due to poor network management and monitoring. According to Gartner research, the hourly cost of system downtime for large enterprises was $42,000, with a typical business experiencing an average 87 hours of downtime a year [9]. As a result, the QoS of any service-based network should be monitored, managed, and evaluated on an ongoing basis.

The methods for QoS evaluation of heterogeneous wireless networks have been extensively studied. The motivations for these studies can be categorised as: Access Network Selection (ANS), Joint Admission Control (JAC), Joint Scheduling Control (JSC), and Vertical Handover (VHO) in heterogeneous networks. These studies mainly focus on two aspects. Firstly, the QoS evaluation of a single application or a
single radio access network in a heterogeneous environment to be able to handover to a better network. Secondly, the strategies to maintain the QoS of the current network while new call admission.

Most of these studies take into account the effects of a single access network on application performance rather than considering the impacts of the access networks on the other side of the connection. To measure the QoS in a heterogeneous environment, the conventional methods do not consider the performance of all the applications running on a network. For example, if there are voice and video conferencing applications running on a UMTS network, these methods do not include the performance analysis of both of these applications to quantify the network QoS. In addition, there is no unified metric to quantify the QoS of a network, which considers the performance of all the access networks present in it. For example, in a heterogeneous environment, there are three access networks, such as UMTS, WiMAX, and LTE. At present, no unified metric can represent the performance of this whole configuration using the QoS-related parameters of these access networks.

Many previous works in this context have also analysed traffic characteristics to recommend QoS evaluation processes for heterogeneous networks. However, precise evaluation of the overall characteristics of such network traffics is a challenging task. The presence of different types of communications technologies and a varying number of users, make that evaluation even more complex. The same user scenario can suggest varying performance results based on the measurement of packet loss and end-to-end delay. For example, with twenty voice clients and one streaming client, in some network, the end-to-end delay of voice calls shows an acceptable value. However, in terms of packet loss, the voice calls do not achieve an acceptable performance value for the same number of users [10]. Therefore, in some cases, even though, the end-to-end delay may be at an acceptable level for some applications, packet loss may simply be too high. In addition, the effects of different communication technologies on the performance of various applications, say, voice and video, must be accounted for in an efficient and methodical manner. In such situations, a unified QoS metric that would quantify the performance of the whole network configuration is helpful rather than analyse the resultant value of each QoS parameter separately.
The industrialised world has deployed several heterogeneous network-based socioeconomic service models such as ICTPD in New Zealand, Video conferencing (VC)-based distance education model in Alberta [11]. This type of network-oriented education and health service models can benefit the developing countries as they lack physical resources. For instance, the number of hospital beds in these countries is few in number, just fractionally more than two per one thousand people [12]. However, the QoS level of these models would vary between a developed and developing country. This is because the QoS parameters related to these models such as technological settings, numbers of users, the environmental settings, and user expectations are dynamic in nature. As a result, it is hard to assess the performance of such model prior to deployment [13]. The available studies in the literature in relation to QoS evaluation of such models have ignored this aspect. A unified QoS metric in this situation can help to measure the overall performance and facilitate better design and planning.

To evaluate the QoS of any heterogeneous network, the available studies have ranked each access network by combining different QoS metrics. The most common parameters, which are considered during this ranking process, are mainly service, network, and user related. For instance, Received Signal Strength (RSS), type of the service (e.g., conversational, streaming, interactive, background), minimum bandwidth, end-to-end delay, throughput, packet loss, bit error rate, cost, transmit power, traffic load, current battery status of the mobile terminal, and the user’s preferences. To combine these parameters into a single value, at first, a weight is assigned to each of these parameters according to its relative importance.

The weights are both subjective and objective in nature. For example, the importance of network-related parameters such as RSS and bandwidth are objective in nature. Application-related parameters such as end-to-end delay, packet loss, and jitter seems objective in nature, some studies reveal that they could be subjective in nature too. For example, a study conducted in Tanzania [14] shows that to evaluate the quality of a network, the users give moderate importance to end-to-end delay over packet loss. Another research study is conducted by the European Telecommunication Standards Institute (ETSI) [15], which reveals that the users give strong importance to end-to-end delay over packet loss. Therefore, the importance of
application parameters can vary based on the contexts, for example, between home and an industrial environment or between developed and developing countries.

The primary issue with the conventional weight calculation methods is that the weights have been assigned to QoS-related parameters in each application according to separate objective functions, and the context-based importance of these parameters has been ignored. Additionally, the application significance is not considered in these methods. However, same as the QoS-related parameters, the importance of any application can also vary with the changing context. For instance, an application for the education service would have more significance than an application for the entertainment service. Without considering this information, the weights will not reflect the proper importance attached to the considered parameter, application, or network.

1.2 Proposed Solutions

The objective of this thesis is to improve the QoS evaluation, management, and monitoring approach of heterogeneous networks. Even though, there are various methods studied in the literature in these relations, a comprehensive method, which can conduct these tasks, is still an ongoing research issue [16]. Therefore, the primary goal of this thesis is to derive a unified metric that will quantify the end-to-end QoS of a heterogeneous network configuration. This unified metric can help the network operators to design fault-tolerant systems and improve network and service quality through proper investigation, monitoring, and management of the systems. For instance, if a network operator wants to deploy a health-service centric network in a certain rural area he needs to consider all the relevant factors such as the most suitable technologies, the expected user, and traffic load in pick and off pick hours etc. The proposed QoS monitoring and management method can help the operators in such cases by facilitating the QoS level assessment before the implementation of that particular system.

To achieve this goal, in this thesis, the concept of unified metric measurement function for network QoS evaluation is introduced. This function considers all the relevant performance-related parameters to quantify the network and application QoS with a single numerical value. To implement this function, a progressive approach is adopted. At first, this approach uses a simple fixed weight-based method, which is
suitable for comparatively simple networks, then moves to a more complex dynamic weight-based method, and finally makes use of the concepts arising from fuzzy logic to take the network performance uncertainty into picture. This methodical approach considers the significance of the QoS-related parameters, the available network-based applications, and the available Radio Access Networks (RANs) to characterise the network performance with a set of three integrated QoS metrics. The first metric denotes the performance of each available application in the network, the second one determines the performance of each radio access network present in the network, and the third one characterise the QoS level of the entire network configuration.

The fixed weight-based method uses a performance related significant-based policy for the weight assignment of QoS-related parameters and an equal weight-based policy for application and networks. This method is suitable for simple networks with relatively fixed applications and technologies. However, it is unable to gather the changing dynamics of a more complex network. For example, if one of the applications is not present in a certain network, the weights will be subject to change. As a result, using the fixed weight-based method, it would be difficult to recalculate these weights. To make the weight calculation method more sophisticated and bring the context-based information into the picture, the dynamic weight-based evaluation is applied, which uses Fuzzy Analytical Hierarchy Process (FAHP) with extent analysis approach. The advantage of using FAHP is that it can combine several policies together while assigning weights. For instance, to determine the weights for any application in the network, the significance of that application relevant to usage purpose and the number of users in that application, these two criteria can be combined. Although, the dynamic weight-based evaluation can consider the active changes of the network, it has some limitations. For instance, it cannot capture the underlying uncertainty of network dynamics. In this relation, the concepts of fuzzy logic are adapted for further enhancement of the analysis method. It is noticeable that the more comprehensive the method is the more accurate and efficient the results would be however, with a higher implementation cost.

To evaluate QoS of any application or network, it is crucial to determine benchmarks for the relevant parameters. However, these benchmarks could be subjective and diverse in nature as stated in the previous section. Most current methods use the objective benchmarks of these parameters. To fill this gap, in this thesis, a context-
Chapter 1: Introduction

Based analysis is applied to determine a range of benchmarks for these parameters rather than choosing a fixed baseline. To define these ranges, various recommendations from the available literature have been analysed. For example, different benchmark ranges of packet loss for voice application have been chosen for an urban and a rural environment based on the recommendations from Cisco, ITU, and other available studies.

This unified metric can make the vertical handover process and call admission control algorithms to be more efficient as well. The VHO process has three phases; these are the system discovery phase, the decision phase, and the execution phase [17, 18]. In the system discovery phase, usually the available access networks are ranked according to a QoS value, which is calculated using several parameters. However, the conventional calculation methods do not calculate the QoS value based on the performance of all the applications present in a network. They are usually interested in the single application of use. In call admission control algorithms, a QoS value is calculated for the candidate network before entry of a new user to estimate the impact of the new load on the existing network QoS. However, this value is calculated based on the parameters of the application, which the new user is going to use, and its impact on the performance of other applications are overlooked.

The RAN QoS metric, which are derived using the proposed methods in this thesis, can deal with such limitations as the metric is derived based on the performance of all the active applications in the network. Using the application and radio access network QoS metric, it is possible to classify networks for their suitability in providing specific services. As a result, any network can be categorised as education or health service-oriented network, and so on. In this way, the new users can be admitted into the networks according to their service requirement.

This methodical approach is particularly useful in order to combine different configuration settings, such as various technologies supporting several applications. Therefore, it can quantify the performance of heterogeneous network-based service models by a unified QoS metric. This approach is also useful when some specific network-based service models are re-deployed from one region to another. Each area has its unique service requirements and technology availability. As a result, the service model, which demonstrates better performance in one context, is not necessarily going to have the same outcomes in another region. In such
circumstances, by applying this approach, it is possible to compare the resultant QoS level of any network-based service model with the expected one.

The efficiency and effectiveness of the proposed approach are assessed through a diverse range of simulation studies. The simulation studies have been categorised in the following fashion throughout the thesis:

1. UMTS based scenarios with different applications and traffic load
2. UMTS-WiMAX-based scenarios with different applications and traffic load
3. UMTS-WiMAX-WLAN based scenarios with different applications and traffic load
4. LTE based scenarios with different applications and traffic load
5. UMTS-LTE-WLAN based scenarios with different applications and traffic load

The reason behind this categorisation is to facilitate the performance comparisons of different technology, application, and traffic situations. For example, a set of simulation results of UMTS network have been compared with the set of results from LTE network. The simulation results indicate that the designed methodology facilitates the QoS evaluation, management, and monitoring of network configuration and applications running on them in a more efficient manner. The simulation studies also show that using the unified metrics, it is possible to choose a suitable network configuration for a particular application or service from among the range of available network configurations under investigation and classify them for their suitability to provide some specific services.

1.3 Thesis Outline

The remaining of the thesis is organised as follows:

Chapter 2 outlines the background and motivations of this thesis. This chapter presents the related works relevant to the major components of this work. It first gives an overview of the current states of the art of wireless and cellular technologies followed by the QoS requirement analysis of various communication technologies in terms of different network-based applications. It then discusses the current solutions and their existing gaps for QoS evaluation in heterogeneous networks. Then it presents a detailed analysis of fuzzy logic in relation to QoS analysis. Later in this chapter, the multi-criteria decision-making algorithms are discussed, which are
applied in the proposed QoS evaluation methodology. The chapter also presents a
detailed discussion on some QoS assessment tools such as E-model and illustrates
the investigation results for selecting an appropriate simulation tool for this thesis.

**Chapter 3** describes the idea of application-based QoS evaluation in heterogeneous
networks. It also introduces the concepts of unified QoS metrics. Detailed
simulation studies are carried out to investigate the impact of different environmental
and technological factors on the performance of network-based applications. The
simulations are conducted using OPNET software. The studies also take into account
other factors such as the number of active users in the network and the presence of
diverse traffics. Some analyses are also conducted to disclose the impact of dynamic
QoS changes on the user perception or Quality of Experience (QoE). The results
collected in this chapter are used to identify the crucial performance parameters for
each application and define the acceptable context-based ranges for these parameters.
These parameters and the ranges are used in the later chapters in designing the
proposed application-based QoS evaluation approach.

**Chapter 4** introduces a novel approach for QoS analysis of heterogeneous networks
referred to as fixed weight-based QoS evaluation. In this method, fixed weights are
assigned to the QoS-related parameters of a network-based application, based on
their importance. A fixed weight is also attached to the applications and the radio
access networks present in a network configuration using an equal importance
policy. An equal importance policy means an equal weight is allocated to each of
these entities. Then three sets of QoS metrics are calculated; these are the application
QoS metric, the radio access network QoS metric, and the network configuration
QoS metric. As the names suggest, the application QoS metric represents the
performance of each application present in a network. The radio access network QoS
metric or access network QoS metric characterises the performance metric for each
access network present in the network. Finally, the network configuration metric or
the configuration metric denotes the performance of the whole network
configuration. Diverse ranges of heterogeneous network-based simulation scenarios
are designed using the background case studies to evaluate the performance of the
proposed method. The communication technologies, which are used in simulations,
are IEEE 802.11 family, WiMAX, UMTS, and LTE. The detailed result analysis is
also presented.
Chapter 5 designs a dynamic weight-based evaluation method for the further enhancement of the QoS analysis method. The idea behind this approach is to consider the context-based information while weight calculation. For instance, a health service-related video conferencing session would have a higher significance than an entertainment movie. This weight assignment is implemented using the fuzzy-based analytical hierarchy process. Detailed simulation studies are carried out to evaluate the performance of this method. A comprehensive result analysis is presented, and the performance is also compared to other available methods in the literature and the fixed weight-based method presented in Chapter 4.

Chapter 6 introduces an entirely new evaluation method based on the fuzzy logic concepts. The importance and the acceptable values of the key QoS-related parameters which are identified in Chapter 3 can change based on the user and network contexts. For instance, the acceptable values of these parameters may vary between a developing country and an industrialised one. In addition, the key QoS evaluation parameters can also differ from a multimedia to a non-multimedia network. To deal with these uncertainties, fuzzy logic offers several accepted solutions. The proposed QoS evaluation method is further enhanced by incorporating techniques motivated by such solutions. The results are analysed and compared with the previously proposed QoS assessment methods.

Chapter 7 provides a summary of this thesis. This chapter mainly discusses the way this research has been developed throughout the end and highlights its contributions. Finally, the potential future directions for this research are illustrated.
CHAPTER 2

Background and Motivations

This chapter presents the background and motivations of this thesis. The aims and objectives of this chapter are to find the existing gaps in the available research work relevant to QoS analysis in heterogeneous networks. This review will also help to set up the groundwork for designing a methodical approach for QoS evaluation in this context. The current states of the art of wireless and cellular technologies are analysed in Section 2.1. In Section 2.2, various interworking architectures in heterogeneous networks are analysed. The QoS requirements of various technologies in terms of different network-based applications are illustrated in Section 2.3. The current solutions, which are proposed for QoS analysis of heterogeneous networks, are evaluated in Section 2.4. In Section 2.5, a detailed analysis of fuzzy logic in relation to QoS analysis is presented. In Section 2.6, some multi-criteria decision-making algorithms are discussed, which are applied in the QoS evaluation methodology of this thesis. Section 2.7 presents the description of the available QoS and QoE measurement tools such as E-model. Section 2.8 presents a detailed investigation to choose an appropriate simulation tool for this thesis. Finally, Section 2.9 concludes this chapter.

2.1 Wireless Communication Technologies

In this Section, an overview of the current states of the art of wireless technologies is presented.

2.1.1 Wireless Local Area Networks (WLANs)

In 1997, the Institute of Electrical and Electronics Engineers (IEEE) released Wireless Local Area Network (WLAN) standard [19]. The common standards for WLANs are 802.11a, 802.11b, 802.11g, 802.11n, 802.11ac, and 802.11ad. The available spectrums for WLANs are the industrial, scientific, and medical bands, 2.4 GHz to 5 GHz [20]. The Medium Access Control (MAC) and physical characteristics (PHY) for WLANs are specified in 802.11 standards. The MAC layer is the same
for all the standards; however, the physical layer differs. The physical layer has three different specifications; these are Frequency Hopping Spread Spectrum (FHSS), Direct Sequence Spread Spectrum (DSSS) and Infrared (IR). 802.11 WLANs are suitable for forming a local wireless community and share resources within them. They can be useful in creating local sharing models in health, commerce, and education. However, they have a limitation in the case of wider communication.

802.11b is the most popular as it is the cheapest one, and it has the best signal range. The frequency range it uses is between 2.4 GHz-2.4835 GHz [21]. It can support data rates of 1, 2, 5.5, and 11 Mbps. 802.11b uses DSSS mechanism for the physical layer. IEEE 802.11a uses orthogonal frequency-division multiplexing (OFDM) modulation technique, which helps to reduce multipath interferences. It operates in the 5 GHz band with up to 54 Mbps data rate within a range of 10 meters.

802.11g is an extension of 802.11b that uses 2.4 GHz and supports a data rate up to 54 Mbps. [22]. These WLAN standards do not have QoS support. As a result, 802.11e has come to fill that gap. 802.11e applies the priority mechanism to support QoS. Each type of data traffic is assigned a priority based on their QoS requirements.

802.11n is proposed to achieve higher throughput [23]. The breakthrough technologies for higher data throughput in this standard are Multiple Input Multiple Output (MIMO), frame aggregation, and channel bonding [24]. MIMO splits the data over a number of data stream through separate antennas while sending, and the receiver receives data through corresponding multiple antennas. Frame aggregation is used to reduce the physical layer overhead by reassembling traditional frames into a “super” frame. In this case, the mandatory break between each of the traditional frame knows as DCF (Distributed Coordination Function) Interframe Space (DIFS) occurs between each “super” frame [25]. Thirdly, the channel bonding is applied to combine two standard 20 MHz channels to a wideband 40 MHz channel. As a result, 802.11n devices can use twice the channel width than other standards available in 802.11 standard family. IEEE 802.11n allows for up to four data streams. However, the maximum performance of IEEE 802.11n is only possible in an IEEE 802.11n only network. If the network is mixed with IEEE 802.11a/b/g devices, the performance will be significantly lower.
IEEE 802.11ac is an enhancement of 802.11n standard to support very high data throughput in the 5 GHz bands. It adds 80 MHz, 160 MHz and non-contiguous 160 MHz channel bandwidth [26]. The other features added for throughput enhancement is multi-user MIMO (MU-MIMO). A more enhanced data coding scheme (256-QAM) is another key feature of this technology. IEEE 802.11ac MIMO works with a maximum of eight data streams to a single user. IEEE 802.11ad is another Gigabit standard, which operates in the unlicensed 60 GHz band [27]. It supports very high throughput up to 7 Gbit/s. Poorer path loss is one of the challenges in this standard. It uses some new technologies such as directional antennas and beamforming in order to improve link quality. Channel modification is also added to address directionality and spatial reuse [26]. The primary focus of IEEE 802.11ad is personal networking. It is mainly designed to deliver wireless connections between the computer and other devices in the network, e.g. a High-Definition Multimedia Interface (HDMI) video connection to a screen or a Universal Serial Bus (USB) connection to other terminal equipment. The main objective is of this standard is to replace cables to provide connectivity between devices. However, this standard will need to co-exist with other standards such as IEEE 802.11a/b/g/n/ac for the provisioning of TCP/IP connectivity within the network and to the Internet [28].

IEEE 802.11af is mainly a regulatory standard, which intends to operate WLAN in the TV white space [29]. To identify the usable white space, IEEE 802.11af will use cognitive radio technology. This cognitive technology will be based on an authorized geolocation database. This database will provide information on which frequency, at what time and under what conditions networks may operate. 802.11h standard has emerged to reduce interference and power consumption. It involves the extension of MAC and PHY layers for dynamic frequency selection (DFS) and Transmission Power Control (TPC) [30]. IEEE 802.11i enables the security feature in 802.11 family, which was previously missing. It combines IEEE 802.11, 802.1X, and Extensible Authentication Protocol (EAP) to provide security to Wireless LANs [31]. Figure 2.1 shows different wireless standards with their frequency and data rate.
2.1.2 WiMAX

Worldwide Interoperability for Microwave Access (WiMAX) is the commercialization of IEEE 802.16 standard, which is also known as Wireless Metropolitan Area Neatwork (WMAN) [32]. The IEEE 802.16 working group (WG) is responsible for developing WiMAX associated standards and its amendments. It is an OFDM-MIMO based high-performance technology [20]. As this technology is open Internet model-based, therefore it enables multiple device models. It can provide broadband wireless access in both fixed and mobile environments. Because of high bandwidth, it can reduce the transmission delay for quality images. The WG’s primary interest was to develop the standard in 10-66 GHz range, but later changed to 2-11 GHz. It can operate in both licensed and unlicensed bands. The
operating frequencies of 2.5 and 3.5 GHz require a license; however, 5.86 GHz frequency is an unlicensed band.

The coverage area of WiMAX is 30 to 50 kilometres with a data transmission rate of 100Mb/s in 20 MHz bandwidth [33]. The IEEE 802.16 has developed a series of guidelines to standardize fix and mobile broadband wireless access. WiMAX addresses the requirements of those users who want to use a broadband connection regardless of location which are not covered by the Digital Subscriber Line (DSL) and cable technologies. As a result, it is widely deployed in the regions, which have more developing countries [34]. The highest coverage it achieved is in the Asia-Pacific.

The WiMAX network has three major elements; these are Mobile Stations (MS), Access Service Network (ASN) and Connectivity Service Network (CSN). The ASN constitutes of base stations and ASN gateways. The CSN WiMAX network provides the IP connectivity and other IP core network functions. The major elements in WiMAX network constitutes of the following entities [35]:

**Subscriber station (SS) /Mobile Station (MS):** The Subscriber station (SS) is also termed as the Customer Premises Equipment (CPE). There are two types of SS referred to as "indoor CPE" and "outdoor CPE". Users can install the indoor CPE. This could be in the form of a dongle for using on the laptop or a computer. The outdoor CPE has a better antenna setup.

**Base Station (BS):** The base station provides the air to the subscriber and mobile stations. It also provides services such as mobility management in terms of handover execution and tunnel establishment. It is responsible for radio resource management, QoS policy management, traffic classification, Dynamic Host Control Protocol (DHCP) proxy, key management, session management, and multicast group management.

**ASN Gateway (ASN-GW):** The Access Service Network Gateway (ASN-GW) is responsible for a layer two traffic aggregation. It is mainly responsible for intra-ASN location management and paging, radio resource management and admission control, caching of subscriber profiles and encryption keys. The ASN-GW is also responsible for the Authentication, Authorisation, and Accounting (AAA) Server client functionality, establishment and management of mobility tunnel with base stations,
QoS policy management, foreign agent functionality management for mobile IP, and routing to the selected Connectivity Service Network (CSN).

**Home Agent (HA):** The HA is located inside the CSN. It works in conjunction with the ASN Gateway, to provide an efficient end-to-end Mobile IP solution for WiMAX network. It is as a connectivity point to provide secure roaming ensuring QoS capabilities for subscribers.

**Authentication, Authorisation and Accounting Server (AAA):** AAA server is located within CSN. As the name suggested, it is responsible for subscription services.

The standardized interfaces in the WiMAX are known as Reference Points (RPs)

termed from R1 to R8 [36]. Figure 2.2 shows the WiMAX network reference model.

This WMAN technology standard was first approved as the IEEE 806.16-2001 standard and was published in 2002. The microwave signals in the frequency range of 10-66 GHz have poor penetrability and rain attenuation easily affects the signals. As a result, this technology is only suitable for open areas. To make this standard work in areas with buildings 806.11a standard was published in 2003 [34].

The frequency range for 806.11a is 2-11 GHz [37]. The coverage area is up to 50 kilometres. It can operate in Non-Line of Sight (NLoS) environment and get less affected by rain. Thus, the installation cost is reduced as it requires fewer antennas.
also supports mesh topology and offers QoS guarantee. In 2004, the IEEE 802.16 WG integrated the IEEE 802.16-2001 and IEEE 802.16a standard together and issued the IEEE 802.16-2004 standard. In 2012, the latest amendment to this standard is published [38]. This 802.16-2004 is referred to fixed-WiMAX as well [39].

802.16-2012 operates within 2-11 GHz and in the fixed NLoS environment. The MAC architecture supports both point-to-multipoint and mesh. The data rate it achieves is between 1-75 Mbps. It employs three kinds of physical layer technologies, which are Single carrier (SC); OFDM 256 points in fixed wireless access and OFDMA 2048 points for long distance between operator point of presence and Wireless Local Area Network. In December 2005, an amendment to the existing 806.11-2004 was completed and approved which is known as 802.16e-2005. This standard forms the basis for the nomadic and mobile applications and known as mobile WiMAX. Although, it is referred to as Mobile WIMAX, it supports fixed application scenarios as well [40]. For fixed applications, it operates within the 2 GHz-11 GHz and for mobile applications, it operates within the 2 GHz-6 GHz. IEEE 802.16 is undergoing a continuous process of development, and the following 802.16 standards are still valid (May 2011) [37]:

- IEEE Std 802.16-2009 [41]
- IEEE Std 802.16j-2009 [42]
- IEEE Std 802.16h-2010 [43]
- IEEE Std 802.16m-2011 [44]
- IEEE Std 802.16k-2007 [45]
- IEEE Std 802.16.2-2004 [46]
- IEEE Std 802.16/Conformance04-2006 [47]

802.16j supports relay mode operation for 802.16 standards [48]. It is a Mobile Multi-hop Relay (MMR) network topology to increase transmission rates at the cell edge. The other ongoing projects for this technology are 802.16n for higher reliability network and 802.16p for enhancement to support machine-to-machine communication. WiMAX has the ability to interwork with satellite and terrestrial wireless existing as well as emerging technologies. It can also serve as a backbone network for WLAN hotspots for connecting to the broadband Internet. The WiMAX
network architecture has flexible features like interworking, roaming, and cost effectiveness. Figure 2.3 outlines characteristics of different WiMAX technologies.

![WiMAX Characteristics](image)

**Figure 2.3 WiMAX Characteristics**

### 2.1.3 Wireless Wide Area Network Technologies (WWANs)

Over the past two decades, cellular and wireless communication technologies have affected the way people communicate and work. Globally the cellular penetration rate is 96%, 128% of which are in developed countries and 89% in developing countries [49]. Cellular networks have started with 1G and now it is evolving towards 5G. Cellular telephone networks are one of the most important types of WWANs. Initially starting with voice services, now they are able to provide quality data services.

#### 2.1.3.1 First generation (1G)

The first generation of cellular technology, also known as 1G used analogue technology. It was mainly voice communication-based and did not allow roaming. It first started in Japan in 1979. The major 1G standards are Advanced Mobile Phone System (AMPS), Total Access Communication System (TACS), Japanese Total Access Communication System (JTACS), and Nordic Mobile Telephone (NMT). The speed was 2.4 Kbps.
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2.1.3.2 Second generation (2G)

The second generation (2G) of cellular technology uses digital transmission and allows roaming between different operators. This technology was primarily designed in the late 1980s and early 1990s before the privatization of the Internet and the advent of World Wide Web (WWW) [50]. With the advent of 3G, although 2G is declining; however, the penetration of 2G is still noticeable. In 2013, the world had 56% penetration if 2G and in 2014, it had 52% penetration [51].

Global System for Mobile Communications (GSM) in Europe and Integrated Digital Enhanced Network (iDEN), IS-136 and IS-95 in North America are examples of 2G [20]. The iDEN is a proprietary technology, which is developed by Motorola. IS-136 is the technology compatible with 1G AMPS standard. This is Time Division Multiple Access (TDMA) based. IS-95 is commercialised as cdmaOne, which is developed by Qualcomm. The speciality of this technology is the use of Code Division Multiple Access (CDMA) based on direct-sequence spread spectrum (DSSS) technique instead of TDMA.

GSM has become the world’s fastest-growing communications technology of all time and the leading global mobile standard. The technical work for maintenance and development of GSM has been transferred to 3GPP in 2000. Terrestrial GSM networks now cover 90% of the world’s population in 219 different countries [52]. GSM operates with a number of frequency bands: 900 and 1800 MHz bands in Europe, Asia-Pacific, and Africa and North America and parts of Central and South America use the 850 and 1900 MHz. It uses TDMA along with Frequency Division Duplex (FDD) to distribute physical resources among multiple users. It is deployed as paired spectrum between uplink and downlink using FDD. Each Radio frequency (RF) is divided into eight TDMA time slots. In this way, eight full-rate traffic channels are multiplexed onto a single RF carrier [53]. GSM uses Gaussian Minimum Shift Keying (GMSK) modulation technique. It can support data rates and voice calls up to 9.6 Kbps. IS-136 uses TDMA and Time Division Duplex (TDD). It allows data rates up to 30 Kbps. IS-95 has two versions, which are known as IS-95A and IS-95B. IS-95A employs FDD with a channel bandwidth of 1.25-MHz for each direction and supports data speeds of up to 14.4 Kbps. IS-95B can support data speeds of up to 115 Kbps by bundling up to eight channels. Due to its data speeds,
IS-95B is categorised as a 2.5G technology. Table 2.1 shows the development of 2G technology.

### Table 2.1 Development of 2G Technologies

<table>
<thead>
<tr>
<th>Standard</th>
<th>Place of Origin</th>
<th>Deployment year</th>
<th>Access Scheme</th>
<th>Carrier Spacing</th>
<th>Number of TDMA time slots</th>
</tr>
</thead>
<tbody>
<tr>
<td>GSM</td>
<td>Europe</td>
<td>1991</td>
<td>TDMA</td>
<td>200 KHz</td>
<td>8</td>
</tr>
<tr>
<td>iDEN</td>
<td>United States</td>
<td>1993</td>
<td>TDMA</td>
<td>25 KHz</td>
<td>6</td>
</tr>
<tr>
<td>IS-136</td>
<td>United States</td>
<td>1993</td>
<td>TDMA</td>
<td>30 KHz</td>
<td>6</td>
</tr>
<tr>
<td>IS-95</td>
<td>United States</td>
<td>1995</td>
<td>CDMA</td>
<td>1.25 MHz</td>
<td>-</td>
</tr>
<tr>
<td>PDC</td>
<td>Japan</td>
<td>1993</td>
<td>TDMA</td>
<td>25 KHz</td>
<td>3</td>
</tr>
<tr>
<td>PHS</td>
<td>Japan</td>
<td>1995</td>
<td>TDMA-TDD</td>
<td>300 KHz</td>
<td>4Tx/4 Rx</td>
</tr>
</tbody>
</table>

#### 2.1.3.3 2.5G

The evaluation of 2.5G happened to provide a data solution for customers along with voice solution in 2G networks. The primary 2.5G standards are General Packet Radio Service (GPRS), Enhanced Data rates for GSM Evolution (EDGE) and IS-95B. GPRS supports a data rate up to 115 Kbps with an average speed of 40 to 50 kbps. Like GSM, GPRS supports international roaming for its customers. GPRS provides Multimedia Messaging Service (MMS) and web browsing. GPRS users can simultaneously maintain a data session and a phone call. It is based on GSM network platform. Therefore, operators can build the infrastructure on the existing base stations (BSs) and Mobile Switching Centers (MSCs). The GPRS has attracted many device manufacturers due to its usability, adaptability, and flexibility. As its core network is based on Internet Protocol (IP) standards, therefore, can easily provide wireless access to Internet Service Providers (ISPs) and corporate Local Area Networks (LANs). While deploying EDGE, the operators reuse GPRS core elements such as Gateway GPRS Support Nodes (GGSNs). Therefore, it helps to make the migration to 3G cost-effective.
EDGE is an enhanced version of GPRS. It can provide a data rate up to 384 Kbps. This technology can handle three times more subscribers than GPRS. The subscriber gets triple data rate with an extra capacity to their voice communications. It is sometimes referred to as a 2.75G technology [54].

### 2.1.3.4 Third Generation (3G)

3G networks have higher data rates, significant system capacity, and improved spectrum efficiency compared to 2G. The 3G based technologies include Universal Mobile Telecommunications System (UMTS), Code Division Multiple Access 2000 (CDMA2000), and Time Division - Synchronous Code Division Multiple Access (TD-SCDMA). These technologies are based on CDMA. Most of the GSM/GPRS mobile operators use UMTS as the 3G technology because it has the built-in infrastructure to migrate from GSM/GPRS. The underlying air radio interface for UMTS is Wideband Code Division Multiple Access (WCDMA). UMTS is managed by the 3rd Generation Partnership Project (3GPP).

UMTS standard is available worldwide for use in the 850, 900, 1700, 1800, 1900, 2100, 1700/2100 and 2600 MHz bands. The original release (Release'99) supports a user date rate up to 384 Kbps. In many countries, the UMTS networks have been upgraded with High-Speed Downlink Packet Access (HSDPA) and High-Speed Uplink Packet Access (HSUPA) technologies to improve the data rate. These two technologies are together termed as HSPA. HSPA is usually referred to as 3.5 technology. The transmission speed for HSPA is increased to 14.4 Mbps.

According to the latest report of the Global mobile Suppliers Association (GSA), HSPA networks have commercially launched in 213 countries compared to 75 countries in 2008. According to the same report, most 3G/HSPA systems use the 2.1 GHz spectrum and 87 operators in 58 countries have commercially launched UMTS900 compared to 26 network operators in 2008. Over 67% of HSPA operators have commercially launched HSPA Evolved (HSPA+) systems and all of the world's WCDMA operators have commercially launched HSPA service. 384 HSPA+ networks are commercially launched in 164 countries and deployment of 42 Mbps Dual channel HSPA+ (DC-HSPA+) technology is the major trend in 2014. WCDMA has 1.627 billion subscriptions, including HSPA. Figure 2.3 shows the global growth of UMTS, and LTE according to Statista’s report.
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UMTS network constitutes of three main elements they are: User Equipment (UE), Radio Network Subsystem (RNS) and Core Network (CN).

**UE:** The UE contains two separate parts; these are Mobile Equipment (ME) and the UMTS Service Identity Module (USIM).

**RNS:** The subsystem that controls Wideband Radio access is usually called UMTS Terrestrial Access Network (UTRAN). UTRAN is divided into several RNSs. The radio element in UTRAN is called NodeB also referred to as Based Station (BS). The controlling part is called Radio Network Controller (RNC). The RNC is situated between the Iub and Iu interfaces. It also has a third interface called Iur to maintain RNS interconnections [55]. The open interface between UE and UTRAN is Uu and the open interface between Iu is between UTRAN and CN. The BS is situated between the Uu and UMTS interface. The Iub interface is situated between RNC and BS. The UTRAN’s primary task is to create and maintain Radio Access Bearers (RABs) for communication between UE and CN. The job of the RAB is to fulfil the QoS requirements required by the CN.
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**CN:** The core network manages all the network elements required for switching and subscriber control. CN manages both packet-switched (PS) and circuit-switched (CS) domains. Registers part in the CN is responsible for static subscription and security information. The RNC and the CN map the end-to-end QoS requirements of UMTS network using the Iu interface. Figure 2.5 shows the UMTS reference architecture.

CDMA2000 is another 3G mobile technology, which is the CDMA version of IMT-2000 standard. The technology is developed by the International Telecommunication Union (ITU). It supports both analogue and digital communication, as a result, suitable for rural areas. The rural areas in the US use this advantage of CDMA2000. The CDMA2000 standard family includes 3xRadio Transmission Technology (3xRTT), CDMA2000 Evolution-Data Optimized (CDMA2000 EV-DO) and CDMA2000 EV-DV (Evolution- Data and Voice). CDMA2000 uses multipath signal strategy. Therefore, the signal arrives in the receivers with different time delays, and the combination creates a stronger signal. The drop-offs in this technology occur when the mobile device goes two times further from the base station.

![Figure 2.5 UMTS Reference Architecture](image-url)
CDMA2000 EV-DO has undergone several releases since it is first launched. Table 2.2 shows the details. CDMA2000 EV-DO carries voice and data over the same 1.25MHz carrier. It offers a peak data rate of up to 4.8 Mbps downstream and up to 307 Kbps upstream. It was never deployed, as there was no requirement to carry data and voice using the same carrier. To improve the data rate of CDMA2000 a three times channel bandwidth is used than the standard 1.25 MHz channel carrier. For CDMA2000 1XRTT, a spreading rate of 1 was used and in 3XRTT this is 3.

### Beyond 3G Technology and 4G

**High-Speed Packet Access Plus (HSPA+):** HSPA+ is one of the prominent beyond 3G technology. It is also termed as HSPA Evolution and Evolved HSPA. HSPA+ enables upgraded support and performance for real-time conversational and interactive services such as Push-to-Talk over Cellular (PoC), picture and video sharing, and Video and Voice over Internet Protocol (VoIP) through the introduction of features like Multiple-Input Multiple-Output (MIMO) antennas, Continuous Packet Connectivity (CPC) and Higher Order Modulations [56]. According to the
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statistics collected on October 2012, 240 HSPA+ networks are deployed over the world [57]. HSPA+ provides a downlink throughput of 21 Mbps. With 64QAM and 2X2 MIMO technology, the theoretical capability is 42 Mbps with an uplink capability of 11.5 Mbps. It is a simple upgrade of HSPA network, which paves a way to bridge between 3G and 4G technologies.

**Long Term Evolution (LTE):** The first release of LTE, which is LTE release 8, is referred to as beyond 3G or sometimes as 3.9G [58]. In contrast to the circuit-switched cellular model, the LTE supports only packet-switched services. The aim of LTE is to provide seamless Internet Connectivity (IP) between the User Equipment (UE) and the Packet Data Network (PDN).

The LTE system consists of three main elements; these are the User Equipment (UE), the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and the Evolved Packet Core (EPC). The E-UTRAN and the EPC together form the Evolved Packet System (EPS). The high-level architecture of LTE is outlined in Figure 2.6.

![Figure 2.6 LTE High-level Architecture](image)

The E-UTRAN manages the radio communications between the UE and the EPC. It has only one component that is the evolved base stations also referred to eNodeB or eNB. Each eNodeB controls the mobiles in one or more cells. The eNodeB that is responsible for communicating with a mobile is known as its serving eNB. Each eNodeB connects with the EPC using of the S1 interface. It connects with the other nearby base stations using X2 interface. The X2 interface is mainly used for
signalling and packet forwarding during handover. A home eNodeB also referred to as HeNB is used to provide femtocell coverage within the home. It is usually purchased by a user privately. The eNodeB usually maintains all radio-related functionality of the overall network. This includes scheduling, radio-resource handling, retransmission protocols, coding, and various multi-antenna schemes.

The LTE core network referred to as EPC has the following component:[59]

- The Home Subscriber Server (HSS) component is from UMTS and GSM. It is a central database that contains information about all the network operator's subscribers.

- The Packet Data Network Gateway (PDN-GW) is responsible for communicating with the other packet data networks using SGi interface. Each packet data network is identified by an access point name (APN). The PDN-GW plays the same role as the GGSN and the SGSN nodes of the UMTS network.

- The serving gateway (S-GW) is as a router that forwards data between the eNodeB and the PDN-GW.

- The mobility management entity (MME) handles the high-level operation of the mobile using signalling messages and HSS.

- The Policy Control and Charging Rules Function (PCRF) is responsible for policy control decision-making, as well as for controlling the flow-based charging functionalities in the Policy Control Enforcement Function (PCEF). It is located in the PDN-GW.

The interface between the serving and PDN-GW is termed as S5/S8. The S5 interface is used if the two devices are on the same network, and S8 is used if they are in different networks.

The EPC is responsible for providing a complete mobile broadband network. This includes authentication, charging functionality and setup of end-to-end connections. Handling these functions separately, instead of integrating them into the RAN, is beneficial as it allows several radio access technologies to be served by the same core network. The EPS also interconnects with other RANs such as 3GPP (GSM/EDGE, UTRAN) and non-3GPP (CDMA, WiFi, WiMAX).
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LTE downlink and uplink transmission both use Orthogonal Frequency Division Multiplexing (OFDM) technology. However, in the uplink a different method is applied to reduce the cubic metric of the uplink transmission, which is known as DFTS-OFDM. One of the key features of LTE is the support for different multi-antenna transmission techniques. Another key feature is the reuse of frequency in neighbouring cells. LTE supports both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) modulation techniques.

LTE is commercially launched in 112 countries. Among those in 27 countries, 40 LTE TDD systems have been commercially launched. In the other 72 countries, 150 LTE1800 have been commercially launched. 79 LTE operators, i.e. almost 1 in 4, are deploying LTE-Advanced Technologies. 21 operators have commercially launched LTE-Advanced carrier aggregation systems [60].

The LTE physical layer specifications support a range of transmission bandwidths with a minimum value. It supports from roughly 1 MHz up to 20 MHz bandwidth. It supports peak data rates of 300 Mbit/s in the downlink and 75 Mbit/s in the uplink by applying spatial multiplexing of four layers (4 × 4 MIMO) in the downlink and 64QAM in both downlink and uplink.

The one of the main reasons for LTE Release 10 was to meet the set requirements by ITU for IMT-Advanced or 4G. Hence, LTE-Advanced is used for the LTE 10 release. Examples of these requirements are: the support for at least 40 MHz bandwidth, peak spectral efficiencies of 15 bit/s/Hz in downlink and 6.75 bit/s/Hz in uplink (corresponding to peak rates of at least 600 and 270 Mbit/s respectively), and control and user plane latency of less than 100 and 10 msec respectively [59].
The 3GPP TR 36.913 document has the design targets for the LTE release 10 or LTE-Advanced [61]. The LTE release 10 supports 30.6 bit/s/Hz in downlink and 16.8 bit/s/Hz in uplink for FDD. For TDD, this spectral efficiency is 30.0 bit/s/Hz and 16.0 bit/s/Hz in the downlink and the uplink respectively. The assumption for deriving the peak spectral efficiency numbers is a deployment with 20 MHz channel bandwidth, 8 × 8 MIMO in the downlink, and 4 × 4 MIMO in the uplink [59]. LTE release 10 has used Carrier Aggregation (CA), where multiple component carriers are aggregated and jointly used for transmission to/from a single terminal.

Five different UE categories have been specified for the LTE Release 8/9, ranging from the low-end category 1 not supporting spatial multiplexing to the high-end category 5 supporting the full set of features in the Release-8/9 physical layer specifications. In Release 10, there are three additional categories.

### Table 2.3 LTE UE Categories

<table>
<thead>
<tr>
<th>UE Category</th>
<th>Downlink Peak Rate (Mbit/s)</th>
<th>Uplink Peak Rate (Mbit/s)</th>
<th>Maximum Downlink Modulation</th>
<th>Maximum Uplink Modulation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Release 8/9/10</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>10</td>
<td>5</td>
<td>64QAM</td>
<td>16QAM</td>
</tr>
<tr>
<td>2</td>
<td>50</td>
<td>25</td>
<td>64QAM</td>
<td>16QAM</td>
</tr>
<tr>
<td>3</td>
<td>100</td>
<td>50</td>
<td>64QAM</td>
<td>16QAM</td>
</tr>
<tr>
<td>4</td>
<td>150</td>
<td>50</td>
<td>64QAM</td>
<td>16QAM</td>
</tr>
<tr>
<td>5</td>
<td>300</td>
<td>75</td>
<td>64QAM</td>
<td>64QAM</td>
</tr>
<tr>
<td>Release 10 only</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>300</td>
<td>50</td>
<td>64QAM</td>
<td>16QAM</td>
</tr>
<tr>
<td>7</td>
<td>300</td>
<td>150</td>
<td>64QAM</td>
<td>64QAM</td>
</tr>
<tr>
<td>8</td>
<td>3000</td>
<td>1500</td>
<td>64QAM</td>
<td></td>
</tr>
</tbody>
</table>

2.2 Interworking Architectures in Heterogeneous Networks

Interworking mechanisms are very important to achieve anytime, anywhere access and seamless mobility in heterogeneous wireless networks [62]. The development of
interworking solutions for heterogeneous wireless networks is widely studied in the literature, especially in the context of IEEE 802.11 WLAN and cellular network integration. There are two main interworking architectures that are defined by 3GPP for WLAN and cellular integration, these are: loose-coupling or interworking WLAN (I-WAN) [63] and tight-coupling also referred to as generic access network (GAN) [64]. The other commonly used interworking solutions are interworking between Mobile WiMAX and 3GPP Long Term Evolution/System Architecture Evolution (LTE/SAE) [65-67].

![Figure 2.7 Loose-coupling and Tight-coupling Architecture](image)

In I-WAN interworking architecture, the two access networks do not have anything in common; however, their core networks are connected. This is mainly an IP layer interconnection. The two networks operate independently, and they communicate through the Internet. The IP address changes when the mobile terminal moves from one access network to another and such heterogeneity is covered by Mobile Internet Protocol (MIP). The integration point is the Home Agent (HA) of the MIP mechanism implemented in the external PDN. In this case, the interworking point is placed after the GGSN.

One important aspect of this architecture is the combination of MIP and AAA functionalities. In this way, the service continuity with roaming capabilities is ensured. The existing AAA server is responsible for exchanging information and
credentials between 3GPP and non-3GPP networks. In loose-coupling architecture, the WLAN and 3GPP data path are separate. The Packet Data Gateway (PDG) is used to interface directly to the GGSN if a single operator deploys two access technologies.

In the GAN architecture, the WLAN access network is deployed as a new radio access technology within the cellular network. As a result, two radio access networks use a single core network. The interworking point is at the 3GPP core network or the UTRAN. In this architecture, the WLAN data traffic uses the 3GPP core network. The handover, in this case, does not involve the change of remote IP address and the AAA policies. The RNC and the SGSN emulator provide functionalities that are equivalent to those of an RNC/SGSN in the UMTS network to hide WLAN access network specifications from UMTS. Its main function is to provide a standardised interface to the UMTS core network. Figure 2.7 shows the loose and the tight-coupling architectures.

The integration between WiMAX and 3GPP networks is termed as WiMAX-3GPP Interworking. There are three scenarios, which are considered for this interworking architecture by the WiMAX forum [68]. The first scenario does not affect either 3GPP or WiMAX architecture. In this case, the user will be charged on the same bill for both 3GPP and WiMAX usage. The Common customer care handles the billing. The second scenario uses direct IP address. In this case, the subscriber may use the WiMAX to access services such as Internet; however, the AAA operations are handled by the 3GPP system.

The scenario 3 or 3GPP IP access allows the operators to extend 3GPP packet switched-based services to the WiMAX network. In this scenario, there are two cases; these are roaming and non-roaming. For a roaming case, an authenticated 3GPP subscriber can access to 3GPP PS services through a WiMAX access network interworking with a visited 3GPP Public Land Mobile Network (PLMN). In the case of non-roaming, the subscriber can access this service through a 3GPP PLMN. In the interworking architecture, the WiMAX ASN is connected to the 3GPP network through a WiMAX CSN that provides IP connectivity.

LTE 3GPP interworks with non-3GPP access systems using a loose-coupling architecture [5]. The 3GPP AAA Server is located in the 3GPP LTE/EPC network,
and the AAA proxy function of ASN-GW is used. The LTE and WiMAX are integrated at the EPC anchor entity. The handover between 3GPP LTE and non-3GPP access systems are managed by EPC.

In the simulations in the later chapters, some of the simulation scenarios are designed based on these interworking architectures.

2.3 QoS Model

According to the four-layered QoS model proposed by [69], the QoS paradigm in the network starts with the identification of a client’s expectations from the service providers, or more precisely from the underlying network they are using. These expectations in turn help service providers in defining the network performance parameters, including packet loss, end-to-end delay, delay variation or jitter, throughput and the like. To achieve the values identified for these parameters, a range of mechanisms is available from the application layer to different network layers. The last part of the model deals with the client’s experience in terms of network performance, relating the model to Quality of Experience (QoE). Figure 2.8 illustrates the four-layered QoS model.
As this work is related to intrinsic QoS, the QoS-related mechanisms in the context of different network-based layers and connections need to be studied in more details. Each network layer employs its own mechanisms to achieve certain QoS levels in terms of the key performance parameters. Physical layer uses adaptive modulation and coding scheme to improve the bandwidth allocation and interference management depending on the network conditions. Network layer uses path and access selection and queuing algorithms to improve the network performance. The QoS mechanisms in the transport and the application layer involve various compression techniques and coding schemes. For example, the robust header compression technique is used to compress the 40 byte overhead of RTP/UDP/IP to a smaller number [70].

Connection management mechanisms are engaged to secure the application-level QoS up to a certain level. Call Admission Control (CAC) schemes are defined to control the maximum number of users in the network to maintain the minimal level
of QoS required to support the ongoing transmissions [71]. For instance, if the entry of a new user in the network causes the interference level to be raised to a value greater than the threshold value, the new user is not admitted. In a heterogeneous network as multiple radio access networks are involved, CAC schemes are termed as Joint Call Admission Control (JCAC) [72].

CAC schemes are classified according to different design options [73]. Based on the centralization of decision-making, these schemes can be divided into centralised and distributed scheme. In the centralised scheme, the Mobile Switching Centre (MSC) controls the admission and in the distributed schemes, the admission is performed in each cell controlled by BS [74, 75]. CAC schemes are also classified based on the number of services. As the networks are now mostly multi-service based, the primary interest in these types of the scheme is now on multiple-service based scheme [76]. CAC algorithms can be also classified as parameter-based or proactive and measurement-based or reactive [77]. Figure 2.9 shows the workflow for the proactive CAC schemes. In this type of scheme, a QoS metric of the current network configuration is calculated if a new user wants to join the network. This QoS metric is then compared against the expected QoS value of the network to take a decision whether the new resource in would be admitted into the network or not. The second type of algorithm computes the resource if a new connection is going to be established. With the new one, if the resource is adequate, then the connection is admitted otherwise it is rejected. The QoS evaluation method that is proposed in the later chapters can be associated with the parameter-based schemes in a sense that it uses a unified metric to evaluate the performance of any access network. Nevertheless, in the traditional parameter-based algorithms the impact of a new resource for the performance of the active applications in the network is ignored.
All these mechanisms aim to achieve timeliness, precision, and accuracy of transmission in relation to network performance. These translate to delay, delay variation or jitter, throughput and packet loss. According to [78] the QoS building blocks are as follows:

- Resource Allocation
- Routing
- Resource Reservation
- Shaping
- Policing

**Figure 2.9 Proactive CAC Scheme**
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- Admission Control
- Classification and Marking
- Queue Management
- Scheduling

Figure 2.10 shows this in details.

QoS and QoE are interlinked and highly related concepts [79, 80]. QoE reflects the perception of the user from the provided service. QoS relates to both application and network level parameters, which in turn dictates QoE, at least to a certain level. Figure 2.11 shows the mapping between network the QoS and the QoE parameters. Accessibility, retention, and integrity of service are the three important concepts.
relevant to QoE. Accessibility covers the issues relevant to unavailability, security, activation, access, coverage, blocking, and call setup time, which is of paramount importance in establishing service guarantees. Retention refers to avoiding the connection loss. The integrity of service relates to the important network performance parameters, including throughput, delay, jitter, and packet loss.

Figure 2.11 Relationship between QoS and QoE

QoS can be broken down into several other quality-related concepts, for instance into, Quality of Resilience (QoR), Class of Service (CoS) and Grade of Service (GoS). CoS is discussed in detail in the next chapter in relation to application-based QoS evaluation. GoS encompasses connection setup time, call blocking probability, and the probability of maintaining a connection. These parameters are mainly related to the accessibility of QoE parameters. For example, in the next-generation network, the call control mechanisms based on the probability of rejection of a request is a GoS parameter. QoR deals with the operational matters of a service [81].

Many researchers have studied the integrity of service from a technical point of view. In some of these works, subjective performance measurements through usability testing are carried out and are then matched to objective performance parameters [82]. It has been noted that the type of the user environment, the type of device, and the type of service can influence QoE. For example, due to a smaller screen size of a mobile device, users may tend to be satisfied with the low video quality of a movie when viewing it on such a device [83]. Similarly, users in a business environment may expect higher QoS compared to a home user. All such user QoE variables must be taken into account when dealing with network and application-based QoS
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analysis. The traditional QoS evaluation methods in this relation tend to ignore this sort of context-based information. To fill this gap, in this thesis this information are considered while QoS evaluation of any network.

2.4 QoS Analysis of Heterogeneous Networks

QoS evaluation in heterogeneous networks has been an active area of research. Different studies have taken it from different perspectives. The major studies can be divided into a few groups; these are roaming across heterogeneous networks, resource utilisation, adoption, resource allocation, and multimedia QoS issues across heterogeneous networks. Roaming across heterogeneous networks can be further divided into call admission control and vertical handover (VHO). The studies related to VHO are discussed in Section 2.6.

An end-to-end QoS framework for heterogeneous wireless networks is proposed in [84]. The authors integrate a three-pane network architecture of a unified based hierarchical policy management framework. A game-theoretic framework for radio resource management is proposed based on allocated bandwidth and the capacity reservation threshold to maintain a certain level of QoS performance of the ongoing connections in the heterogeneous networks [85].

One of the key focuses of many research works related to QoS provision for heterogeneous networks is about resource allocation to mobile users on a call or connection level. The probabilities of the new call blocking and existing call dropping are two main QoS metrics that are widely studied [86, 87]. The call admission algorithm as stated in the previous Section, determines whether an incoming call can be admitted or not in the network, based on the efficient allocation of the limited bandwidth maintaining the guaranteed users’ QoS requirements at the same time. A call is forced to terminate if the target cell does not have enough bandwidth to support the connection while opting for a handover. Therefore, a network that has some buffer bandwidth for the expected handovers is likely to have lower connection dropping probability and hence better QoS [16]. A distributed resource application algorithm for heterogeneous networks is proposed in [88]. The algorithm enables each network base station or access point to perform its own resource allocation rather than the central resource manager. Most of the other
available resource management algorithms, in this case, use a central resource manager [89].

Bandwidth reservation techniques are another widely studied topic for QoS provision in heterogeneous wireless networks. Some studies have suggested to use the expert knowledge of the traffic pattern of a certain area and the estimation of channel occupancy time distribution for bandwidth reservation [90]. To reduce the call dropping probability and the call blocking probability, and to maximise the bandwidth utilisation, the previous movements of the mobile users are used for bandwidth reservation [91]. The local and global mobility profiles for mobile users are generated to predict the future path of a mobile user.

Due to the unpredictable nature of wireless networks, the mobile applications, especially the multimedia applications should be able to adapt to changing QoS condition. As a result, the QoS provision of multimedia applications over heterogeneous networks is another extensively researched topic. A novel quality-aware the adaptive concurrent multipath transfer solution is proposed in [92]. The main idea behind this algorithm is the continuous observation and analyse of each path’s data handling capability to take a decision on the data delivery adaptation. Some of the studies also consider QoE-driven adoption scheme for multimedia applications [93].

A QoS-aware routing algorithm is proposed in [94]. The authors propose this algorithm according to the analysis of a policy-based QoS supporting infrastructure. In a dynamic network environment, the policy-based QoS supporting system enables the operators to accommodate policies to meet ever-changing requirements.

A mobile QoS framework for heterogeneous IP multimedia subsystem (IMS) is presented in [95]. The framework supports Session Initiation Protocol (SIP)-based IMS mobility, where the UE is modelled as a transition in the multicast group membership. To reduce mobility impact on service guarantees, UEs make QoS reservations in advance at the neighbouring IMS networks to ensure QoS guarantee during the lifetime of their sessions.

Many other aspects of QoS in heterogeneous wireless networks have been studied. The potential of seamless roaming in 3G/4G networks has led to the study of QoS
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oriented vertical handovers [96-98]. The main idea behind these studies is to evaluate the situation whether or not the current network is able to satisfy the QoS requirements of the considered application. If the current network is unable to satisfy the QoS requirements of the considered application, then the Mobile Terminal (MT) moves to a suitable network.

Resource management solutions in integrated networking architecture comprising various wireless communication networks such as GPRS and UMTS, IEEE 802.11 and WiMAX are also widely studied. An efficient load balancing-based policy framework for resource management in a loosely-coupled cellular/WLAN integrated is presented in [99]. The policy uses a two-phase control strategy. It uses call assignment to provide a statistical quality of service guarantee during the admission phase. Then, the dynamic vertical handover during the traffic service phase is used to minimise the performance variations. A generic reservation-based QoS model for the integrated cellular and WLAN networks is proposed in [100]. The model ensures the delivery of adaptive real-time flows for end users by taking the advantage of high data-rate WLAN systems and the wide coverage area of cellular networks.

Other topics that are studied related to QoS in heterogeneous networks are signalling protocol, hardware robustness, and energy efficiency. Effective signalling protocols ensure the seamless exchange of information between wireless terminals and various components of the integrated network [62, 101]. The optimisation of power control and base station assignment can affect the resource management and QoS in wireless networks [102, 103].

A network architecture oriented, energy-efficient VHO process is illustrated in [104]. The study has introduced the concept of a central management entity called Virtual Domain Controller (VDC) in a 3G-WLAN-based tight-couple architecture. Figure 2.12 shows the workflow for this algorithm. The MT detects the more power efficient interface for the selected activity, for instance, between 3G and WLAN interface of an MT. Then it sends a handover request to the VDC. The VDC analyses the effect of this handover on the overall network performance. It accepts or rejects the request according to this analysis.
Figure 2.12 Tight-couple-based Energy-efficient VHO

It considers the uplink and downlink load independently. An MT might use 3G access technology to upload and WLAN technology to download according to the selection criteria. The idle time in the uplink and the downlink might cause high-energy consumption in this manner.

An energy-efficient integrated framework to facilitate VHO, based on collected context information is presented in [105]. Figure 2.13 shows this process in detail. They propose to include this multi-module framework on the MT side. The Real-time Monitoring Module (RTMM) is responsible for collecting energy-related parameters in real-time. This set of data is used to build up an auxiliary VHO context, which takes place in the Context Fusion Module (CFM). The Network Discovery Module (NDM) is responsible for maintaining a list of available RANs list. The VHO decision module combines energy-related information and RAN information to take the VHO decision. This algorithm assigns comparatively more burden on the MT side, which leads to more battery consumption.
The QoS analysis that considers different aspects of a heterogeneous network is necessary to evaluate and enhance the application-based performance. To address this necessity, several studies [6, 106, 107] have examined the application-centric performance of these networks. Nevertheless, the establishment of an integrated QoS metric for this type of configuration as a whole is still a challenging task [108]. Most of the existing research discussed above focus on the partial QoS evaluation of a heterogeneous network by deriving the QoS level of a single access network and a single application present within a heterogeneous environment. In addition, different studies have come up with different performance metrics for QoS evaluation of these networks. In this thesis, the focus is on combining the different performance metrics together to come up with a unified QoS metric to evaluate the QoS of heterogeneous network configuration as a whole. The proposed method in this thesis integrates the application QoS level and the network QoS level in heterogeneous wireless networks under various traffic and mobility scenarios.

### 2.5 Applications of Fuzzy Logic

Fuzzy Logic (FL) is viewed as a multi-valued logic, which deals with the approximate mode of reasoning rather than the precise mode [109]. It has been described as computing with words [110]. Primarily, it was introduced to model the uncertainty of natural language, and since then has been widely used as a mean to
support intelligent systems [111]. Although, the history of fuzzy logic goes back to 1965, the use of fuzzy logic in computer networking had started to become popular during the 1990s [112]. It has been extensively used in QoS evaluation of networks and web-based services [113-115].

2.5.1 Fuzzy Logic Principles
A brief overview of the basic principles of fuzzy logic is outlined in this Section. Using FL it is possible to map the imprecise values or the word expressions to a crisp number. It also provides the options to use natural language for rule classification. As a result, complex algorithm of a mathematical model could be easily explained and understandable by non-experts [116]

2.5.1.1 Fuzzy Versus non-Fuzzy Set Theory
The traditional sets are defined by crisp boundaries fuzzy sets, on the other hand, are defined by the ambiguous boundaries [117]. For example, in today’s world to classify whether a certain product is made in Australia or imported would be hard using the classical set. Because these days it is hard to classify which product is locally made as some parts of that product may be imported from outside. There is no clear boundary in such cases. Therefore, the fuzzy theory can be used in such situations, as they do not have clear boundaries. The age of people is also an example of fuzzy sets. Using the crisp set theory, a person between 50 and above is regarded as old. It is hard to put a person of 49 years in a group. On his 50th birthday, he will suddenly become old. However, such a sudden change is against common sense.

Another example could be given in relation to the application QoS evaluation. For example, 400 msec is regarded as the upper boundary of end-to-end delay for a quality voice call. Now, if any voice call experiences a 395 msec delay, in this case, how the quality of this voice call should be evaluated? Would this call be evaluated as poor or acceptable? Hence, in many situations, when a person analyses a statement, there is always natural fuzzification. Therefore, a membership curve can describe to what degree an element does or does not belong to a specific set.

As stated earlier, a fuzzy set has uncertain boundaries. A set is represented using a membership function defined on a space called the universe of discourse. The fuzzy sets apply the idea of the degree of membership. Therefore, the membership function
maps the elements of the universe of discourse onto numerical values between 0 and 1. A membership function value of zero implies that the corresponding element is not an element of the fuzzy set; while a value of one means that, the element fully belongs to the set. A value close to 0 means it has less possibility to become a member of the set and a value close to 1 means it has more possibility to become the member of that set.

The advantage of fuzzy set theory over crisp set theory is that a fuzzy set can constitute of elements with only a partial degree of membership (a membership degree value between 0 and 1). In the example of the end-to-end delay stated previously, it was hard to classify the delay such as 395 msec. By applying fuzzy theory, it would be easier to classify these types of value, as fuzzy sets do not have sharply fixed boundaries. Therefore, a delay of 395 msec can be an acceptable delay with a degree of 0.3. As it has a lower degree of membership, the evaluated QoS level of the voice call will show the impact.

### 2.5.1.2 Basic Operations and Definitions

Some of the basic operations and definitions related to the fuzzy set theory are discussed here.

**Definition 1** If $X$ is a collection of objects which are denoted graphically by $x$, then a fuzzy set $A$ in $X$ is a set of ordered pairs [118]:

$$A = \{(x, \mu_A(x)) \mid x \in X\}$$

(2.1)

$\mu_A(x)$ is called the membership function, which maps $X$ to the membership space $M$.

**Definition 2** A linguistic variable is defined through a quintuple $(x, T(x), U, G, M)$, in which $x$ is the name of the variable; $T(x)$ is the term set of $x$, that is, the set of names of linguistic values of $x$. Each of these values is a fuzzy variable, denoted graphically by $X$ and ranging over a universe of discourse $U$; $G$ is the rule defined in the form of a grammar; $M$ is the semantic rule which maps the terms in $T(X)$ to the fuzzy sets in $U$ [119].

For example, the temperature is a linguistic variable, and it can take cold, warm, and hot as linguistic terms.
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**Definition 3** The membership functions of the intersection of two fuzzy sets \( A \) and \( B \) is defined as:

\[
\mu_{A \cap B}(X) = \min(\mu_A(X), \mu_B(X)) \forall x \in X
\]  
(2.2)

**Definition 4** The membership functions of the union of two fuzzy sets \( A \) and \( B \) is defined as:

\[
\mu_{A \cup B}(X) = \max(\mu_A(X), \mu_B(X)) \forall x \in X
\]  
(2.3)

**Definition 5** The membership function of the complement of a fuzzy set \( A \) is defined as:

\[
\mu_A(X) = 1 - \mu_A(X) \forall x \in X
\]  
(2.4)

The concepts of fuzzy membership functions and linguistic variables are extended in the following Sections.

2.5.1.3 Fuzzy Membership Functions

A membership function is the graphical representation of a fuzzy set. A membership function for a fuzzy set \( X \) in the universe of discourse \( U \) is defined as \( \mu_X : U \rightarrow [0,1] \), where each element of \( U \) is mapped to a value between zero and one. This value, called membership value or degree of membership, quantifies the grade of membership of the element in \( U \) to the fuzzy set \( X \). The universe of discourse for the example of locally made product is the list of all products. The term membership function would correspond to a graphical curve that would define the degree to which any product would be considered as local made.

There are different types of membership functions [120]. These functions are illustrated as follows:

1. **Linear Functions**: The linear are the simplest types of membership functions.

   There are mainly two types of linear membership functions. They are:

   **Triangular Function**: A triangular membership function for \( m \) is defined by three points, a lower limit \( a \), an upper limit \( b \) and a value \( c \), where \( a < c < b \). It can be defined as:

   \[
   f(m,a,b,c) = \max \left\{ \min \left( \frac{m-a}{c-a}, \frac{b-m}{b-c} \right), 0 \right\}
   \]  
   (2.5)
**Trapezoidal Function**: A trapezoidal function has a flat top. Four points are used to define this function; these are a lower limit \(a\), an upper limit \(b\), a lower support limit \(b_1\), and an upper support limit \(c\).

A trapezoid function for a value \(m\) is defined as:

\[
f(m,a,b,c,d) = \max \left\{ \min \left( \frac{m-a}{b-a}, 1, \frac{d-m}{d-c} \right), 0 \right\}
\]

(2.6)

There are two special cases for a trapezoidal function, which are called \(R\) function and \(L\) function. In the case of \(R\) function: \(a = b = -\infty\) and for \(L\) function this is \(c = d = +\infty\)

A triangular and trapezoidal function for delay is defined in Figure 2.14.

![Figure 2.14 Example of Triangular and Trapezoidal Function](image)

**Gaussian Functions**: A Gaussian function is defined by a central value \(m\) and a standard deviation \(\sigma > 0\). A Gaussian function is expressed as:

\[
f(m,\sigma,c) = e^{-\frac{(m-c)^2}{2\sigma^2}}
\]

(2.7)

where the parameter \(c\) is the distance from the origin and \(\sigma\) is the standard deviation or indicates the width of the curve. The Gaussian membership functions have the features of being smooth and non-zero at all points.
3. **Bell-shaped Functions:** The Bell-shaped functions have symmetrical shapes. They can be expressed as:

\[
f(m,a,b,c) = -\frac{1}{1 + \left|\frac{m-c}{a}\right|^{2b}}
\]

(2.8)

where the parameter \(b\) is usually positive, the parameter \(c\) indicates the centre of the curve and \(a\) represents the width of the curve. Bell-shaped functions also have the feature of being smooth and non-zero at all points like Gaussian functions. Figure 2.15 and 2.16 show the example of a Gaussian and bell-shaped membership function.

![Figure 2.15 Example of Gaussian Function](image1)

![Figure 2.16 Example of Bell-shaped Function](image2)

4. **Sigmoidal Function:** A sigmoidal function is usually open to the right or left depending on the sign of the steepness of the function. It is expressed as:

\[
f(m,a,c) = -\frac{1}{1 + e^{-a(m-c)}}
\]

(2.9)

where \(c\) locates the distance from the origin and \(a\) determines the steepness of the function. If \(a\) is positive, the membership function is open to the right, and if it is negative, it is open to the left. This type of construction gives them the advantage to represent “very large” or “very negative”.
5. **Polynomial-based Functions:** The polynomial functions can be categorised as polynomial-Z (zmf), polynomial-S (smf) and polynomial-PI (pimf). Polynomial-Z and polynomial-S are always asymmetric.

\[ y = zmf(m, [a, b]) \]  
(2.10)

where the parameters \( a \) and \( b \) locate the extremes of the sloped portion of the curve.

\[ y = smf(m, [a, b]) \]  
(2.11)

This spline-based curve is a mapping on the vector \( x \), and is named because of its S-shape. The parameters \( a \) and \( b \) locate the extremes of the sloped portion of the curve

\[ y = pimf(m, [a, b, c, d]) \]  
(2.12)

The membership function is evaluated at the points determined by the vector \( x \). The parameters \( a \) and \( d \) locate the "feet" of the curve, while \( b \) and \( c \) locate its "shoulders".

### 2.5.1.4 Linguistic Variables

If a variable takes word in natural languages as its values instead of numerical values, it is called a linguistic variable [118]. Linguistic variables are used to define the uncertainty. For example, the speed of a car can be defined as “fast”, “slow”, etc. Hedges are the fuzzy set qualifiers associated with the linguistic variable. They modify the shape of fuzzy sets. They include adverbs such as very, somewhat, quite, more or less and slightly. For instance, the speed of a car can be defined as “very fast”, “slightly slow” and so on.

It can be defined using a context-free grammar. The semantic of the linguistic terms is illustrated using fuzzy numbers, which are represented by membership functions. For example, a context-free grammar \( G \) is a 4-tuple structure \((V_N, V_T, I, P)\), where \( V_N \) is non-terminal, \( V_T \) is the terminal, \( I \) is the starting rule, and \( P \) is the production rule. The second approach, on the other hand, defines the linguistic term set using an ordered structure and the semantic of this linguistic terms is derived from the ordered
structure, which may be either symmetrically distributed on an interval of [0,1] or not. For instance, a set of five terms \( T \) is defined as:

\[
T = \{ t_0 = \text{none}, t_1 = \text{very low}, t_2 = \text{low}, t_3 = \text{medium}, t_4 = \text{high} \}
\]

### 2.5.1.5 Fuzzy Rules

A fuzzy rule takes the form of IF-THEN conditional statements [121]. It is expressed as:

\[
\text{IF } X \text{ Then } Y
\]

where \( X \) and \( Y \) are termed as fuzzy propositions, which contain linguistic variables. \( X \) is called the premise and \( Y \) is the consequence of the rule. Fuzzy propositions can be two types, these are: atomic fuzzy propositions and compound fuzzy propositions. An atomic fuzzy proposition is a single fuzzy statement. For example:

**Delay is Low**

where delay is the linguistic variable and low is the linguistic value of delay. A compound fuzzy proposition on the other hand consists of more than one atomic fuzzy proposition and uses the connectives “and”, “or” and “not” representing fuzzy intersection, fuzzy union and fuzzy complement respectively. For example,

**IF Delay is Low and Jitter is Low and Packet Loss is not Low**

There are many different ways to interpret the fuzzy rules. These are Denies-Rescher implication, Lukasiewicz implication, Zadeh implication, Gödel implication and Mamdani implication. In this thesis, Mamdani implication method is used for its simplicity and efficiency. In Mamdani implication, the fuzzy IF-THEN rules are interpreted as a fuzzy relation \( R_{MM} \) in \( U \times V \) with the membership function:

\[
\mu_{R_{MM}}(x,y) = \min \left[ \mu_{F_P}(x), \mu_{F_Y}(y) \right]
\]

(2.13)

The above rule can be explained using the following example.

Let delay be a fuzzy variable and the linguistic variables for delay is low, medium, and good. The network QoS has the linguistic variables as poor, medium and good. One of the rules could be:

**If Delay is low then the network QoS is Good.**
Now, the membership functions for the delay in the universe of discourse $U$ is given by:

$$
\mu_{\text{low}}(d) = \begin{cases} 
0, & \text{if } d \geq 150 \\
\frac{d}{150}, & \text{if } 0 \leq d \leq 150 \\
1, & \text{if } d \leq 0 
\end{cases} 
$$

(2.14)

$$
\mu_{\text{medium}}(d) = \begin{cases} 
0, & \text{if } d \geq 400 \\
\frac{d-150}{250}, & \text{if } 150 \leq d \leq 400 \\
1, & \text{if } d \leq 150 
\end{cases} 
$$

(2.15)

$$
\mu_{\text{high}}(d) = \begin{cases} 
0, & \text{if } d \geq 600 \\
\frac{d-400}{200}, & \text{if } 400 \leq d \leq 600 \\
1, & \text{if } d \leq 400 
\end{cases} 
$$

(2.16)

Now, the membership function for the network QoS in the universe of discourse $V$ is given by:

$$
\mu_{\text{poor}}(\text{netQoS}) = \begin{cases} 
0, & \text{if } \text{netQoS} \geq 2 \\
\frac{\text{netQoS}}{2}, & \text{if } 0 \leq \text{netQoS} \leq 2 \\
1, & \text{if } \text{netQoS} \leq 2 
\end{cases} 
$$

(2.17)

$$
\mu_{\text{medium}}(\text{netQoS}) = \begin{cases} 
0, & \text{if } \text{netQoS} \geq 3 \\
\frac{\text{netQoS} - 1.5}{1.5}, & \text{if } 1.5 \leq \text{netQoS} \leq 3 \\
1, & \text{if } \text{netQoS} \leq 1.5 
\end{cases} 
$$

(2.18)

$$
\mu_{\text{good}}(\text{netQoS}) = \begin{cases} 
0, & \text{if } \text{netQoS} \geq 5 \\
\frac{\text{netQoS} - 3}{2}, & \text{if } 3 \leq \text{netQoS} \leq 5 \\
1, & \text{if } \text{netQoS} \leq 3 
\end{cases} 
$$

(2.19)
where netQoS denotes the network QoS and the measurement scale for network QoS are between zero and five. If the Mamdani implication operator of the equation (2.13) is applied to the rule antecedent and the rule consequent, the result is:

\[
\mu_{\text{netQoS}}(d, \text{netQoS}) = \min \left[ \mu_d(100), \mu_{\text{netQoS}}(4) \right] = \min[0.67, 0.5] = 0.67 \tag{2.20}
\]

### 2.5.1.6 Fuzzy Inference System

A Fuzzy Inference System (FIS) maps the fuzzy input parameters to the output parameters using fuzzy theory. This mapping provides the ability to decision-making or pattern detection [122]. The strength of FIS depends on their twofold identity. Firstly, they have the ability to handle linguistic concepts. Secondly, their universal approximators are able to perform non-linear mappings between inputs and outputs [123]. There are mainly two types of fuzzy inference systems: Mamdani Fuzzy Inference Method and Sugeno Inference Method.

The most commonly used fuzzy inference systems are the Mamdani method that was introduced in 1975. Professor Ebrahim Mamdani of London University built one of the first fuzzy systems to control a steam engine and boiler combination [124]. The Sugeno Inference system was introduced by Takagi-Sugeno-Kang in 1985. The first two parts of the fuzzy inference process (fuzzifying the inputs and applying the fuzzy operator) of the Sugeno method are the same as the Mamdani method [125]. The main difference between these two systems is regard to the consequent of rules [126]. The Mamdani fuzzy system uses fuzzy sets as the rule consequent. On the other hand, the Sugeno system uses linear functions of input variables as the fuzzy consequent. The Sugeno system works well with linear techniques and optimisation and adaptive techniques [127].

In this thesis, the Mamdani FIS is used due to its simplicity, widespread acceptance, and the vibrancy of output membership functions compared to Sugeno systems. A Mamdani inference system has the following steps:

- **Fuzzification of the input variables:** The first step is to take the crisp inputs and determine the degree to which these inputs belong to each of the appropriate fuzzy sets. This step involves simple calculations by applying graphical analysis to a crisp input value (X axis) versus its membership
value (Y-axis). In other words, fuzzification is the process of taking actual real-world data and converting them into the fuzzy input. For example, an end-to-end delay input value of 400 msec may be converted to a fuzzy input value of high. This process produces multiple fuzzy outputs for every real-world input based on the number of membership functions defined for the system. The goal of the fuzzification step is to produce inputs that can be processed by the rule evaluation step.

- **Rule evaluation:** The second step is to take the fuzzified inputs, and apply them to the antecedents of the fuzzy rules. If a given fuzzy rule has multiple antecedents, AND or OR are used to obtain a single number that represents the result of the antecedent evaluation. This number is then applied to the consequent membership function. For example, IF delay is low, and jitter is low, and packet loss is low then QoS is good. The end-result of the rule evaluation step is a set of rule strength. The rule strength is determined from the numeric values of its input labels.

- **Aggregation of the rule outputs:** Aggregation is the process of unification of the outputs of all rules. In this step, the membership functions of all rule consequents previously scaled are combined into a single fuzzy set. In other words, the input of the aggregation process is the list of scaled consequent membership functions, and the output is one fuzzy set for each output variable.

- **Defuzzification:** Fuzziness helps to evaluate the rules, but the final output of a fuzzy system has to be a crisp number. The input for the defuzzification process is the aggregate output fuzzy set. The output is a single number. There are several defuzzification methods, but probably the most popular one is the centroid technique. This technique works by finding the point where a vertical line would divide the aggregate set into two equal masses.

### 2.6 Multi-criteria/Multi-Attribute Decision Making Algorithms

Decision-making processes related to telecommunication networks is becoming extremely complex due to the increased environmental complexity, fast growing
technological evolution, rapid changes in service availability, market structure, and growing customer demands [128]. Involvement of multiple communication technologies in a single network is making the QoS evaluation of any network complex more than ever as each technology has its own characteristics and the applications utilising them have their own QoS requirements. These multidimensional QoS issues in the new technological platforms lay down the ground for multi-criteria analysis method.

Multi-criteria decision-making (MCDM) or Multi-Attribute Decision Making (MADM) algorithms have been widely used in the area of the heterogeneous network from vertical handover perspectives [6, 94, 106, 107, 129-133]. The most common criteria, which are considered during this ranking process are service, network, and user related. For instance, received signal strength, type of the service (e.g., conversational, streaming, interactive, background), minimum bandwidth, delay, throughput, packet loss, bit error rate, cost, transmit power, traffic load, current battery status of the mobile terminal, and the user’s preferences. To combine these attributes into a single value, a weight is assigned to each of this attribute according to its relative importance.

The weights for QoS-related parameters have both subjective and objective elements in it [134]. The network attributes, for example, the importance of received signal strength and bandwidth are objective in nature. Application related attributes such as delay, packet loss and jitter although seems objective in nature, some studies reveal that they have a potential subjective nature. For example, a study conducted in Tanzania [14] shows that users give moderate importance to delay over packet loss. Another research study is conducted by the European Telecommunication Standards Institute (ETSI) [15], which reveals that users give strong importance to delay over packet loss. Therefore, the importance of service attributes can vary according to context, for example, between a home and industry environment or between developed and developing countries.

VHO process usually has three phases: system discovery phase, the decision phase and the execution phase [17, 18]. A QoS-aware fuzzy rule base algorithm is proposed for vertical handover in [94]. The algorithm takes a multi-criteria-based decision considering various parameters of different traffic classes such as end-to-end delay,
jitter, and packet loss. The weights for each of these parameters are considered based on their relative importance in relation to that specific traffic class.

A network selection method established on Analytical Hierarchy Process (AHP) and fuzzy Total Order Preference by Similarity to the Ideal Solution (TOPSIS) is proposed in [129]. The fuzzy TOPSIS method is used to rank each access network, according to application performance. Table 2.4 shows the details of the results from this method.

**Table 2.4 Network Ranking Using Fuzzy TOPSIS Method**

<table>
<thead>
<tr>
<th>Applications</th>
<th>Technology</th>
<th>Ranking Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Web Browsing</td>
<td>UMTS</td>
<td>0.62</td>
</tr>
<tr>
<td></td>
<td>WiMAX</td>
<td>0.67</td>
</tr>
<tr>
<td></td>
<td>WLAN</td>
<td>0.65</td>
</tr>
<tr>
<td>VOIP</td>
<td>UMTS</td>
<td>0.6</td>
</tr>
<tr>
<td></td>
<td>WiMAX</td>
<td>0.61</td>
</tr>
<tr>
<td></td>
<td>WLAN</td>
<td>0.615</td>
</tr>
<tr>
<td>Streaming media</td>
<td>UMTS</td>
<td>0.58</td>
</tr>
<tr>
<td></td>
<td>WiMAX</td>
<td>0.66</td>
</tr>
<tr>
<td></td>
<td>WLAN</td>
<td>0.62</td>
</tr>
</tbody>
</table>

The AHP and the Grey Relational Analysis (GRA) are combined to calculate the final network QoS in [130]. The AHP breaks down the network selection criteria into several sub-criteria and assigns a weight to each of these criteria, and sub-criteria. Then GRA is applied to rank the candidate access networks. Then MADM algorithm is applied again to rank the candidate networks.

Some studies have applied Elimination and Choice Translating Reality (ELECTRE) for network selection [135]. A cost-function-based access network selection model is proposed in [136]. The study considered the available bandwidth and received signal strength for network selection.

An application-based vertical handover algorithm is proposed in [137]. A QoS level for each candidate network is quantified by the application performance. Then this QoS value is compared with the current QoS value for that specific application before a mobile terminal initiates any handover.

An access network selection scheme is proposed based on fuzzy logic, Genetic Algorithms (GAs) and MCDM algorithm in [138]. The scheme has used GA to
assign suitable values for weights of the considered criteria. Three parallel FL subsystems are used to score each of the available RAN. The scores and the weights are then passed on to the MCDM system to take the final decision for VHO.

For weight assignment, the available literature on QoS evaluation in network selection has mostly used the AHP method, which is primarily developed by Saaty [94, 129, 130, 139-141]. Some studies have also assigned fixed weights to these parameters based on their importance to service performance [142].

Both AHP and fixed weight methods are unable to handle the subjective and ambiguous factors related to weight determination such as context-based significance. To deal with this situation, in [143], the authors have used Fuzzy AHP (FAHP) method to assign these weights and ELECTRE method to in order to select the most suitable network from a range of available candidate networks.

However, for weight calculation, the network-related technical parameters, such as delay, jitter, packet loss, and throughput are compared against the non-technical parameters such as cost. This can cause an imbalance in the weight assignment. In this thesis, although FAHP is used for weight calculation, it is used in a different manner. The criteria that are related to each other are grouped together for creating a pairwise comparison matrix. For example, the weights are assigned in a hierarchical manner starting from QoS-related parameters, applications running on a network and each active radio access network in a network configuration. Some of the multi-criteria algorithms, which are used for weight calculation, are discussed here:

**Analytic Hierarchy Process (AHP) Method:** The AHP method was introduced by Saaty as stated earlier. This method makes decisions about complicated problems by dividing them into a hierarchy of decision factors. In this way, the analysis becomes simple and easy. This method has the following steps:

1. **Determination of the objective and the decision factors:** This step involves the analysis of the final objective of the problem. The problem is analysed as a number of decision factors until it acquires a hierarchical structure. The lowest level is the alternative solution to the problem.

2. **Determination of the relative importance of the decision factors:** This step includes the pairwise comparisons between the attributes in each hierarchy. These
comparisons are generally based on how influential one attribute compared to another attribute.

3. Normalisation and calculation of the relative weights: In this step, the relative weights are calculated for each attribute. The largest Eigenvector method is used if the matrix is inconsistent.

**Fuzzy AHP with Extent Analysis (FAHP):** In FAHP, the scales that are used in the AHP is expressed via Triangular Fuzzy Numbers (TFN). The classical FAHP method has the following steps:

1. The first step is the formulation of fuzzy judgement matrix. This matrix contains the pairwise comparisons of the attributes of the considered alternatives. It is expressed as:

\[
R^a = \begin{bmatrix}
    r_{11}^a & r_{12}^a & \cdots & r_{1n}^a \\
    r_{21}^a & r_{22}^a & \cdots & r_{2n}^a \\
    \vdots & \vdots & \ddots & \vdots \\
    r_{n1}^a & r_{n2}^a & \cdots & r_{nn}^a
\end{bmatrix}
\]  

(2.21)

where \(a\) represents the number of alternatives, and \(i,j=1,2,\ldots,n\)

2. In the second step, TFNs are obtained that represents the grades of alternatives

\[
l^a_{ij} = \min_a \left( r^a_{ij} \right), m^a_{ij} = \frac{\sum_i r^a_{ij}}{a}, u^a_{ij} = \max_a \left( r^a_{ij} \right)
\]

(2.22)

where \(a=1,2,\ldots,t\) and \(i,j=1,2,3,\ldots,n\)

A TFN is a special type of fuzzy number which is denoted as \(\tilde{m} = (k, l, m, u)\). Its membership function is:

\[
\mu_m(k) = \begin{cases} 
  \frac{k-l}{m-l}, & \text{if } l < k \leq m \\
  \frac{u-k}{u-m}, & \text{if } m < k \leq u \\
  0, & \text{otherwise}
\end{cases}
\]

(2.23)
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The addition operations for two TFN numbers \( k_1 \) and \( k_2 \) are as follows:

\[
 k_1 \oplus k_2 = (l_1, m_1, u_1) \oplus (l_2, m_2, u_2) = (l_1 + l_2, m_1 + m_2, u_1 + u_2)
\]  

(2.24)

The multiplication operation for these two TFN numbers is expressed as:

\[
 k_1 \otimes k_2 = (l_1, m_1, u_1) \otimes (l_2, m_2, u_2)
\]

(2.25)

3. In the third step, the FAHP matrix is established that contains TFNs representing pairwise comparisons between the attributes for a certain alternative. This matrix can be expressed as:

\[
 \tilde{R}_{ij} = \begin{bmatrix}
 (1,1,1) & (l_{i2}, m_{i2}, u_{i2}) & (l_{i3}, m_{i3}, u_{i3}) \\
 (l_{i1}, m_{i1}, u_{i1}) & (1,1,1) & (l_{i2}, m_{i2}, u_{i2}) \\
 & & (1,1,1)
\end{bmatrix}
\]

(2.26)

where \( \tilde{R}_{ij} = ij \cdot \tilde{u}_{ij} \) and \( \tilde{u}_{ij} = \left( \frac{1}{m_{ij}}, \frac{1}{l_{ij}} \right) \)

4. In this step, the weights are calculated using the fuzzy extent analysis method [144]. Fuzzy extent analysis method considers an object set and a goal set. Each object is analysed to find out to which extent an object can satisfy a respective goal. Let \( X = \{x_1, x_2, \ldots, x_n\} \) be an object set, and \( U = \{u_1, u_2, \ldots, u_m\} \) be a goal set. In extent analysis method, each object is taken to perform extent analysis of each goal respectively. Hence, \( m \) extent analysis values for each object could be attained as follows [145]:

\[
 M_{g11}^1, M_{g12}^1, \ldots, M_{g1m}^1, \ldots, M_{g21}^1, \ldots, M_{g2m}^1, \ldots, M_{gn1}^1, \ldots, M_{gnm}^1, i=1,2,\ldots,n,
\]

(2.27)

where all the \( M_{gij}^j (j = 1,2, \ldots, m) \) are TFNs.

The steps of the extent analysis methods are as follows:

Step 1. The value of fuzzy synthetic degree with respect to the \( i \)th object is defined as:
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\[ D_l = \sum_{i=1}^{m} M_{gi}^j \otimes \left[ \sum_{i=1}^{n} \sum_{j=1}^{m} M_{gi}^j \right]^{-1} \]  

(2.28)

\[ \sum_{j=1}^{m} M_{gi}^j \] can be obtained employing the fuzzy addition operation of \( m \) extent analysis values for a particular matrix such that

\[ \sum_{j=1}^{m} M_{gi}^j = (\sum_{j=1}^{m} l_j, \sum_{j=1}^{m} m_j, \sum_{j=1}^{m} u_j) \]  

(2.29)

To obtain \( \left[ \sum_{i=1}^{n} \sum_{j=1}^{m} M_{gi}^j \right]^{-1} \), a fuzzy edition operation is performed for \( m \) extent analysis values for a particular matrix such that

\[ \sum_{i=1}^{n} \sum_{j=1}^{m} M_{gi}^j = (\sum_{i=1}^{n} l_i, \sum_{i=1}^{n} m_i, \sum_{i=1}^{n} u_i) \]  

(2.30)

Then compute the inverse of the vector in Equation (2.19) such that

\[ \left[ \sum_{i=1}^{n} \sum_{j=1}^{m} M_{gi}^j \right]^{-1} = \left( \frac{1}{\sum_{i=1}^{n} u_i}, \frac{1}{\sum_{i=1}^{n} m_i}, \frac{1}{\sum_{i=1}^{n} l_i} \right) \]  

(2.31)

Step 2: The degree of possibility of \( M_2 = (l_2, m_2, u_2) \geq M_1 = (l_1, m_1, u_1) \) is defined as

\[ V(M_2 \geq M_1) = \sup [\min(\mu_{M_1}(x), \mu_{M_2}(y))] \]  

(2.32)

In addition, can be equivalently expressed as follows:

\[ V(M_2 \geq M_1) = \text{hgt}(M_1 \cap M_2) = \mu_{M_2}(d) \]

\[ = \begin{cases} 
1 & \text{if } m_2 \geq m_1, \\
0 & \text{if } l_2 \geq u_1, \\
\frac{l_2 - u_2}{(m_2 - u_2) - (m_2 - l_1)} & \text{otherwise}
\end{cases} \]  

(2.33)

where \( d \) is the ordinate if the highest intersection point \( D \) between \( \mu_{M_1} \) and \( \mu_{M_2} \).

To compare \( M_1 \) and \( M_2 \) both the values of \( V(M_1 \geq M_2) \) and \( V(M_2 \geq M_1) \) are needed.

Step 3: The degree possibility for a convex fuzzy number to be greater than convex fuzzy numbers \( M_i (i = 1, 2, \ldots, k) \) can be defined by
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\[ V(M \geq M_1, M_2, \ldots, M_k) = V[(M \geq M_1) \text{ and } (M \geq M_2)] \text{ and \ldots} \]

\[ (M_1 \geq M_k)] \]

\[ = \min V(M \geq M_i), i = 1, 2, \ldots, k \quad (2.34) \]

Assume that

\[ d'(A_i) = \min V(d_i \geq d_k) \quad (2.35) \]

For \( k = 1, 2, \ldots, n; k \neq i \). Then the weight factor is given by

\[ W' = (d'(A_1), d'(A_2), \ldots, d'(A_n))^T \quad (2.36) \]

Where \( A_i (i = 1, 2, \ldots, n) \) are \( n \) elements.

Step 4: Via normalisation, the normalised weight vectors are

\[ W = (d(A_1), d(A_2), \ldots, d(A_n))^T \quad (2.37) \]

where \( W \) is a non-fuzzy number.

There are several issues encountered while using AHP weight-based method. One of the requirements of AHP is that it assumes any two attributes at the same level of the hierarchy are independent of each other. However, in network QoS evaluation, the application related QoS parameters such as delay or jitter have interrelationships. As a result, weight calculation of QoS-related parameters using this method is not accurate enough. Another issue with the conventional AHP methodology is its failure to effectively handle the intrinsic imprecision and fuzziness associated with the mapping of the end user’s preferences to crisp numbers. As a result, in thesis FAHP method is used for weight calculations.

2.7 E-model

The E-model is a well-known transmission-planning tool that provides a forecast of the expected voice quality of the considered connection. [146]. It is applicable for quality assessment of both the voice quality of wire-line and wireless scenarios, based on circuit-switched and packet-switched technology. The quality that is predicted represents the perception of a typical telephone user, for a complete end-to-
end (i.e. Mouth-to-ear) telephone connection under different conversational conditions. To evaluate the user satisfaction and communication quality the model uses several parameters, these are: basic signal to noise ratio, delay impairment factor, equipment impairment factor, and an advantage factor. The equipment impairment includes the influence of packet loss. The advantage factor represents an advantage of access which certain systems may provide in comparison to conventional systems. The primary output of the E-model is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other quality measures such as Mean Opinion Score (MOS), Percentage Good or Better (GoB) or Percentage Poor or Worse (PoW).

The E-model is established on an impairment factor-based mathematical algorithm. This algorithm is responsible for the transformation of the individual transmission parameters into different individual "impairment factors". These impairment factors are assumed to be additive on a psychological scale [147]. The algorithm also takes into account the combined effects of the simultaneous occurrence of those impairment factors at the connection level and the masking effects. E-model predictions may be inaccurate when the combined effect is not considered proper.

The relation between the different impairment factors and R is given by the following equation:

$$ R = R_0 - I_s - I_d - I_{s,\text{eff}} + A $$

where:

The term $R_0$ expresses the basic signal-to-noise ratio (received speech level relative to the circuit and acoustic noise).

The term $I_s$ represents all impairments that occur more or less simultaneously with the voice signal, such as: too loud speech level (non-optimum OLR), non-optimum sidetone (STMR), quantization noise (qdu), etc.

The term $I_d$ sums all impairments which take place due to delay and echo effects.

The term $I_{s,\text{eff}}$ is an "effective equipment impairment factor", which represents impairments caused by low bit-rate codecs. It also includes impairment due to packet losses of random distribution. The values of $I_s$ are transformed to $I_{s,\text{eff}}$ in case of
random packet loss, using the E-model algorithm. The values of $I_e$ is taken from [148].

The term A is an "advantage factor", which allows for an "advantage of access" for certain systems relative to conventional systems, trading voice quality for convenience. While all other impairment factors are subtracted from the basic signal-to-noise ratio $R_o$, $A$ is added to compensate other impairments to a certain amount. It is used to take into account the fact that the user will tolerate some decrease in transmission quality in exchange for the "advantage of access". Examples of such advantages are cordless and mobile systems or connections into hard-to-reach regions via multi-satellite hops.

### Table 2.5 E-model Details

<table>
<thead>
<tr>
<th>$R$</th>
<th>MOS</th>
<th>User Satisfaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>90 &lt; R &lt; 100</td>
<td>4.34 &lt; MOS &lt; 4.5</td>
<td>Very satisfied</td>
</tr>
<tr>
<td>80 &lt; R &lt; 90</td>
<td>4.03 &lt; MOS &lt; 4.34</td>
<td>Satisfied</td>
</tr>
<tr>
<td>70 &lt; R &lt; 80</td>
<td>3.60 &lt; MOS &lt; 4.03</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>60 &lt; R &lt; 70</td>
<td>3.10 &lt; MOS &lt; 3.60</td>
<td>Many users dissatisfied</td>
</tr>
<tr>
<td>50 &lt; R &lt; 60</td>
<td>2.58 &lt; MOS &lt; 3.10</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>

A value between 90 and 100 for $R$ factor leads to a higher user satisfaction. Table 2.5 shows the interpretations of $R$ factor for different values, the interpretations of Mean Opinion Score (MOS) and the satisfaction measurement based on MOS.

### 2.8 Simulation Tools

Network simulation is a powerful tool to study any complex system, to verify and evaluate the performance of newly developed protocols and analyse the existing protocols in different circumstances. These simulators try to model the real-world networks. However, the simulation tools need to be credible enough to generate accurate results. A number of available network simulators have been studied for this thesis and finally, the Optimized Network Engineering Tool (OPNET) modeller version 17.5 is chosen.
The OPNET has been chosen for several reasons. Using OPNET it is possible to simulate a large, complex network consisting of different radio access networks, applications and protocols. The OPNET libraries provide the option of choosing from a diverse library of the existing network components. These models are much more reliable as these models have been developed by the industry experts.

It is a discrete event-based simulator. Discrete event simulators define the behaviour of a complex system as an ordered sequence of well-defined events. An event in this context is a specific change in the state of the modelled system at a specific point in time [149]. It has a hierarchical design with three separate levels; these are the Network Editor, the Node Editor and the Process Editor. These three levels together are called the OPNET modelling domain. The Network Editor is mainly the graphical representation of the modelled network, where devices and linking mediums can be intuitively combined [150].

The node editor displays the internal structure of network objects and graphically describes the data flow between functional elements of protocol layers, radio links, and buffers. Using the node editor, the users can create a new network entity or modify the internal structure of an existing network device. A node is composed of several modules. Each module represents a particular functionality of a specific node. These modules are usually separated by logical functionalities. They can communicate with each other through packet streams and statistic streams. Packet streams are responsible for packet flows through these modules. Modules are responsible for transmitting and receiving packets, processing and storing data, routing packets, and so, on [151].

The process editor is the place to implement the actual algorithms and protocols for node modules. Each node module has specific process models assigned, which consist of finite state machines and transitions. These state machines and transitions can be accessed and modified in the Process Editor.

Objects are the building blocks of the OPNET model. These objects are present in each modelling domain. Objects are created to specify behaviours of a node or create, store and manage information. They are also responsible for processing, modifying, or relaying information and responding to events. One object can contain other objects. There are different mechanisms to create objects in the OPNET
modeller. They could be created by the explicit specification by the user via graphical or programmatic methods. For instance, UMTS UE could be created by simply dragging it from the Project Editor’s object palette. Some objects are also created automatically by the system.

The objects provide attributes to control their behaviour. The attribute is part of the object’s interface. Each attribute consists of a name, a value, and properties. Properties to specify the rules governing the use of the attribute, including its data type, allowable values, suggested values, and documentation. The OPNET models consist of objects and, on the other hand, most objects have a model attributed to them. The behaviour and structure of the OPNET objects are specified by a model. The interface of objects is specified by model attributes. Some objects can be customised via their model attribute. However, some objects cannot be modified using their model attributes.

Packets are the most commonly used mechanism for information exchange between simulation entities. Packets in the OPNET represent different types of information-carrying structures in communications and systems architectures that include packets, messages, frames, cells, datagrams, and transactions. In stream-oriented communications, packets are used to represent key signalling information, bandwidth requirements, etc. Packets typically travel over streams in the Node domain and the Network Domain; they are usually transferred over links.

The OPNET simulations can produce different types of output, which are important for analysing any system behaviour. It supports a set of key statistics, which are usually important in simulation studies. At the same time, it enables the users to customise the statistics according to the project requirements. The support for application-specific statistics in relation to an object attributed as a local and overall system related referred to as global are both available. Simulations can potentially generate large amounts of output data for efficient analysis of system behaviour. Results for collection are defined by probes, which also specify options that affect the manner in which results are collected or displayed.
2.9 Summary

In this chapter, the background information on various wireless technologies, the interworking architectures in heterogeneous networks and the available QoS evaluation methods in the literature in the context of heterogeneous networks have been discussed. A detailed review of the theoretical concepts related to this thesis, such as fuzzy logic, and multi-criteria decision-making algorithms have been presented. The gaps in the current literature in relation to QoS evaluation in heterogeneous networks have been illustrated. To fill these gaps, in the next chapter, an application-based approach is outlined to analyse the QoS evaluation in heterogeneous wireless networks. The QoS requirements of different applications over various access technologies are discussed. The impacts of different factors, such as environment, technology, traffic type, and numbers of active users on the application performance are also examined.
CHAPTER 3

Application-based QoS Evaluation

The increasing popularity of multimedia traffic over wireless and cellular networks among the consumers has raised new demands for the investigation of their underlying Quality of Service (QoS) and Quality of experience (QoE) requirements. In order to assess QoS and QoE of any network efficiently, network and service-related performance metrics should be identified carefully. Many parts of the developing world are highly dependent on wireless and cellular technologies while the provision of socio-economic services in industrialised countries is based on conventional broadband and advanced cellular systems. Clearly, the variations of the underlying networking technologies perturb the QoS and QoE.

To discuss these ideas, in this chapter, an application-based approach is introduced for QoS evaluation in heterogeneous wireless networks. The QoS requirements of different applications and access technologies are studied. The influence of various factors such as environment, technology, traffic types, and the number of active users on the application performance is investigated. Some analyses are also conducted to examine the impact of QoS on the user perception or QoE. This chapter sets the groundwork for the later chapters in designing the methodical approach for QoS evaluation. The general concepts of this chapter are presented in Section 3.1. Section 3.2 outlines the ideas behind the unified QoS metrics. The QoS requirement analyses for different applications are described in Section 3.3. The simulation studies are presented in Section 3.4. Section 3.5 presents the result analysis related to these simulation studies. Then Section 3.6 summarises the outcomes of this chapter and gives the idea of the overall design of the proposed methodical approach presented in the following chapters.

3.1 Introduction

In order to assess network QoS efficiently, network and service-related performance parameters should be identified carefully. The importance of these parameters can
vary based on the user contexts. For instance, a user in any industrial settings would be interrupted even with a smaller amount of network delay. On the other hand, a user in a rural area might be satisfied just with the availability of connection regardless of high delay. Although, this may not directly affect the roles of these parameters in determining the QoS level, it can affect the QoE perceived by the end-users. As a result, it is crucial to consider this type of context-based factors when QoS is analysed for any network.

A typical heterogeneous network, usually consists of different communication technologies. Each of this technology has its own offered bandwidth, operating frequency, coverage area, delay, and packet loss requirements. Therefore, it is hard to assess the QoS of such a network as it involves a diverse range of parameters. Although, the communication technologies have different performance assessment parameters, the applications within these radio access networks have the same QoS requirements. As a result, it would be easier to evaluate the QoS of the access networks and the overall network configuration depending on the performance of applications running on them. Using such application-based QoS evaluation approach, the heterogeneous nature of the underlying networks and the diversity of their traffic can be adequately taken into account.

### 3.2 Unified QoS Metric

The quality of service on any network or application is usually evaluated through a set of specific metrics. For example, to assess the performance of any voice application, the delay, jitter and packet loss are measured and compared with the acceptable values of these parameters. The QoS of a network is evaluated through parameters such as delay, packet loss, throughput, and available bandwidth. Due to the presence of different types of communication technologies and applications, the QoS assessment of heterogeneous network is a challenging task. To deal with these challenges, in this thesis, the concept of unified metric measurement function for network QoS evaluation is introduced. This function considers all the critical performance parameters to quantify the network and application QoS with a single numerical value.

The QoS evaluation approaches proposed in the later chapters consider any heterogeneous network as a set of three layers; these are the application layer, the
radio access network layer and the network configuration layer. Each of these layers uses a function to quantify a unified QoS metric, which flows to the next layer and derive the combined metric of that layer. Figure 3.1 shows the general flowchart of this approach.

![Figure 3.1 Workflow of the Proposed Approaches](image)

**Figure 3.1 Workflow of the Proposed Approaches**

In the application layer, a function is defined to derive the QoS of each application through a unified application QoS metric. This function combines the values of several application performance metrics in order to do that. Therefore, the QoS of a network-based application is treated as a function of QoS-related parameters. This can be expressed as:

\[
QoSAM_A = f (Q_{P_1}, Q_{P_2}, ..., Q_{P_p})
\]

(3.1)
where \( A \) denotes network-based application and \( QP \) refers to the QoS-related parameters. Then in the radio access network layer or RAN layer, the QoS of each access network, which is present in the network, is evaluated. This evaluation is conducted based on the performances of the active applications in those access networks. Hence, the QoS of an access network is viewed as a function of the application QoS metrics. It can be expressed as:

\[
QoSRM_R = f \left( QoSAM_{i=0,1,2,...,n} \right)
\]  

(3.2)

where \( R \) denotes any radio access network, and \( i \) refers to the number of active applications present on a network. Finally, to evaluate the QoS of the overall network configuration another function is defined, which uses the radio access network metrics as its input. This is expressed as:

\[
QoSCM_N = f \left( QoSRM_{j=1,2,...,n} \right)
\]  

(3.3)

where \( N \) denotes the whole network configuration, and \( j \) refers to the number of radio access networks present on a network. All these functions discussed above are used in the later chapters to formulate the comprehensive QoS evaluation methods to deal with complex networks.

### 3.3 Application-based QoS Requirement Analysis

In this section, QoS requirements of different network-based applications are outlined. Application categorisation is important in defining the required QoS levels of network-based applications [152]. There are various criteria proposed by researchers to accomplish such categorisation, though. For instance, applications can be categorised depending on the underlying transport protocol, time dependency, and resource allocations. Voice and video applications, mainly use the unreliable User Datagram Protocol (UDP) for data transfer, whereas Transmission Control Protocol (TCP) is the protocol of choice for transferring of files and web-browsing applications. Applications are also characterised according to service classes referred to as Class of Service (CoS). Most of the advanced wireless and cellular technologies have their own service classes. These classes resolve the behaviour and the acceptable values for the network performance parameters or the integrity of service parameters.
To satisfy the QoS requirements of different applications, UMTS has defined four service classes; these are the conversational class, the streaming class, the interactive class and the background class. The conversational class deals with the applications, which are delay-sensitive and designed for human-to-human interactions. Highly interactive voice and video applications, such as video conferencing (VC), falls into this category. The applications in the streaming class are mainly machine-to-machine interaction-based applications, which can tolerate some delay. Live streaming is an example of such applications. The interactive class supports applications, which are intended for human-machine interactions. The most popular applications in this category are perhaps web browsing, online gaming, and Telnet. The background class deals with what can be considered as traditional Internet applications, including email, file sharing, and file download.

The service classes are treated in a slightly different manner under WiMAX technology. The service classes in the WiMAX technology are: the Unsolicited Grant Service (UGS) class, the Real-time Polling Service (rtPS) class, the extended Real-time Polling Service (ertPS), the non-Real-time Polling Service (nrtPS) class and the best effort (BE) service class. The UGS class deals with delay-intolerant applications such as Voice over Internet Protocol (VoIP) and VC. The rtPS class deals with streaming applications such as audio and video streaming. The ertPS class covers delay-sensitive variable bit rate applications such as VoIP with silence suppression. The applications with variable bit rate such as File Transfer Protocol (FTP) are part of the nrtPS class. The BE service class supports email and web browsing applications.

Time dependency is another essential criterion for application classification [153]. The applications that require the output performance resembling the real-time situations are classified as Real-time applications. Those that do not need such performances are classified as non-Real-time applications. Real-time applications are in general delay-sensitive, but loss-tolerant. On the other hand, non-Real-time applications are delay-tolerant but sensitive to packet loss. Applications may also be classified based on the resource allocation, such as symmetric and asymmetric [153]. For example, video telephony is a symmetric application, requiring similar resources at both sender and receiver ends. Whereas video-on-demand and live streaming are
asymmetric applications, requiring more resources on the server side compared to that of the client side.

No matter how the applications are classified, the aim is to establish the acceptable levels for the QoS-related parameters, such as packet loss, delay, and delay variation for various categories. This will also depend on the infrastructure constraints and the underlying networking technologies in use. The levels may, for instance, need to be considered differently for conventional, wireless, or cellular networks. While there has been substantial research in this area, interestingly enough, there are clear discrepancies between the published works on the acceptable values of QoS-related parameters [15, 154-157].

According to the Ping End-to-end Reporting (PingER), for a VC session, packet loss below 1% is regarded as satisfactory, 1%-2.5% of loss are acceptable, 2.5%-5% are poor, 5%-12% are very poor, and loss above 12% is unacceptable [156]. The observations indicate that when a VC session experiences 4%-6% of packet loss, it is hard for non-native speakers to communicate properly. This analysis demonstrates that user’s background can affect the assessment of application QoS level. On the other hand, according to Cisco, the acceptable packet loss for a VC session should be limited to 1% [155]. As reported by ITU-T, the acceptable level of packet loss for VC is 3% [15] Figure 3.2 compares accepted end-to-end delay values for different applications reported by 3GPP, ITU-T and Cisco [15, 154, 155, 158].

For conversational and videophone applications, the Cisco, ITU-T and 3GPP agree that it is preferred to have an end-to-end delay of no more than 150 msec. However, according to ITU-T, this value refers to a long-term achievable value. Given the current technology, an end-to-end delay value of around 400 msec is considered acceptable by both ITU-T and 3GPP. For real-time games, 3GPP recommends a delay of no more than 75 msec.

In contrast, ITU-T recommends a value no more than 200 msec. The requirements for two-way asymmetric data application such as Telnet also vary from one source to another. ITU-T recommends that the end-to-end delay should be less than 200 msec. However, 3GPP considers 250 msec as an acceptable value. Different sources also give different benchmarks for jitter or delay variations. For instance, Cisco requires
the delay variation for an audio call to be less than 30 msec, while ITU-T and 3GPP say it is preferable to have delay variation less than 1 msec [15, 154].

![Figure 3.2 Comparison of End-to-end Delay](image)

Figure 3.3 shows the recommended values of packet loss for different applications by three key studies [15, 154, 155]. For the loss of audio calls, the sources provide different benchmarks. According to ITU-T and 3GPP, it is preferred to have less than 3% packet loss, and Cisco sets it to no more than 1%. According to another resource, the highest amount of tolerable packet loss for audio calls could be even 20%, depending on the codec and the proper error correction technique [159].
The VS application also shows different recommendations for its QoS-related parameters. According to ITU-T, end-to-end delays of less than 10 second are preferred for VS applications. However, Cisco recommends a value of less than 4-5 seconds. Cisco recommends a value of no more than 5% for packet loss, where only 1% of packet loss is recommended by ITU-T. These can be looked at in light of another study conducted in Tanzania [14]. That study shows that 74% of users are satisfied with 1%-5% of packet loss. ITU-T Recommendations Y.541 suggests slightly different values for packet loss, delay, and delay variation compared to other sources [81]. Table 3.1 shows the values relevant to multimedia applications.

**Figure 3.3 Comparison of Acceptable Packet loss in different Applications**
Additionally, performance parameters can vary according to different environmental settings. For example, the vehicular and the static mobile environments experience different values for QoS-related parameters, such as packet loss and jitter. Table 3.2 and 3.3 summarise the packet loss and mean jitter for stationary and vehicular conditions reported in [160]. The results show that the vehicular environment experiences more packet loss in mobile networks than that of stationary situations. For instance, when the operator 2 changes the cell, it experiences 3.68% of packet loss in the UMTS network, which is slightly higher than the previously stated accepted values from ITU-T, 3GPP and Cisco. The experimental results also show that the packet loss varies according to the type of radio access technology. For example, under certain circumstances the calls in the GPRS network undergo more loss and delay variations compared to the calls in the UMTS and HSDPA network [160].

It is clear from the above analysis that various studies have suggested slightly different acceptable values for the critical QoS-related parameters of network-based applications. The environmental factors also affect the results of the performance evaluation as the values of these QoS-related parameters increases or decreases due to environmental settings. Therefore, to define acceptable ranges for these parameters, various simulation studies are conducted in the later sections.

<table>
<thead>
<tr>
<th>Applications</th>
<th>IP packet delay variation</th>
<th>IP packet Error Ratio</th>
<th>IP packet Loss Ratio</th>
<th>IP Packet Transfer Delay</th>
</tr>
</thead>
<tbody>
<tr>
<td>Highly interactive (VoIP, VC)</td>
<td>50 msec</td>
<td>1x10-4</td>
<td>1x10-3</td>
<td>100 msec</td>
</tr>
<tr>
<td>Interactive (VoIP, VC)</td>
<td>50 msec</td>
<td>1x10-4</td>
<td>1x10-3</td>
<td>400 msec</td>
</tr>
<tr>
<td>Video Streaming (VS)</td>
<td>Undecided (U)</td>
<td>U</td>
<td>U</td>
<td>1 second</td>
</tr>
</tbody>
</table>
### Table 3.2 Packet Loss in Different Cellular Networks

<table>
<thead>
<tr>
<th>Network type</th>
<th>Static operator one</th>
<th>Static operator two</th>
<th>Vehicular operator 1 no cell changes</th>
<th>Vehicular operator 2 no cell changes</th>
<th>Vehicular Operator 1 cell changes</th>
<th>Vehicular operator 2 cell changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>GPRS</td>
<td>9.66</td>
<td>13</td>
<td>n/a</td>
<td>n/a</td>
<td>32.35</td>
<td>18.20</td>
</tr>
<tr>
<td>UMTS</td>
<td>0.23</td>
<td>0.122</td>
<td>0.2</td>
<td>1.78</td>
<td>2.16</td>
<td>3.68</td>
</tr>
<tr>
<td>HSDPA</td>
<td>0</td>
<td>0</td>
<td>1.05</td>
<td>1.14</td>
<td>5.74</td>
<td>4.18</td>
</tr>
</tbody>
</table>

### Table 3.3 Mean Delay Variation in Different Cellular Networks

<table>
<thead>
<tr>
<th>Network type</th>
<th>Operator 1 static</th>
<th>Operator 2 static</th>
<th>Operator 1 vehicular no cell changes</th>
<th>Operator 2 vehicular no cell changes</th>
<th>Operator 1 vehicular cell changes</th>
<th>Operator 2 vehicular cell changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>GPRS</td>
<td>13</td>
<td>107.78</td>
<td>n/a</td>
<td>n/a</td>
<td>213.59</td>
<td>159.02</td>
</tr>
<tr>
<td>UMTS</td>
<td>0.122</td>
<td>54.63</td>
<td>50.21</td>
<td>61.76</td>
<td>70.42</td>
<td>70.42</td>
</tr>
<tr>
<td>HSDPA</td>
<td>0</td>
<td>32.95</td>
<td>52.01</td>
<td>55.48</td>
<td>69.81</td>
<td>71.92</td>
</tr>
</tbody>
</table>

### 3.4 Simulation Studies

Simulation is regarded as one of the modern technologies to conduct network-based research. It is proven to be an effective technique in terms of studying large and complex networks. Path loss models, which are considered as one of the dominant factors in simulation modelling, can be configured to fit real-world behaviours [161]. In most of the cases, it is more expensive and time-consuming to build a large-scale network and test it in different environments. Network simulators provide time and...
cost effectiveness in simulating a large-scale network. This section presents the simulation studies, which are carried out to analyse the performance of network-based applications in different contexts. The simulations are conducted using OPNET. Results for different targeted environments and technologies are collected.

The goal of these simulation studies is to analysis the impact of different environmental factors, diverse traffics, and variant numbers of users on the performance of network-based applications. A primary overview of the simulation setups is presented here. The simulation models are validated against the different real-time scenario experiments available in the literature. To analyse the environmental effects on application performance, simulations are conducted in the following environments:

- Outdoor Rural Area
- Outdoor Urban environment
- Indoor Urban environment
- Pedestrian Urban environment
- Mixed Urban environment
- Suburban area

A rural area is considered to have smaller houses, large gardens, hills and no tall buildings. The outdoor urban environments have the tall buildings and fewer trees. The indoor urban environment has mainly urban office settings with several floors. Pedestrian environment refers to the typical pedestrian environment in the urban city with tall buildings. The suburban area has a mix of the urban and rural environment.

To simulate different environments, various path loss models are selected. The ITU path loss models such as indoor office, outdoor-to-indoor pedestrian model are used to simulate indoor office, pedestrian and mixed urban environments. For outdoor rural environments, the Hata model for small city and the Hata model for small and medium cities are used. For the rural area and suburban area, the free space path loss model and Hata path loss models are used.

The technologies that are considered for simulations are:

- UMTS
- LTE
The applications that are considered:
- Video Conferencing (VC)
- Video Streaming (VS)
- Voice

Table 3.4 shows the simulation parameters used for the RLC layer in the UMTS network setup. Two UMTS-based QoS classes are used; these are the background and the conversational classes. For VC and voice calls, the conversational class is used as these two applications have more stringent QoS requirements. The background class is used for the VS applications. In the UMTS technology, there are three modes available in the RLC layer; these are the Unacknowledged Mode (UM), the Acknowledged Mode (AM) and the Transparent Mode (TM). The AM is used for the VS applications, as video applications tend to have better performance in the AM mode [162].
However, for the voice calls and VC sessions, the UM is used as these applications are delay-sensitive in nature. In the RLC layer, for each QoS class, the data channel is configured separately. The transmission time interval is set to 10 msec and 20 msec for the conversational and background QoS classes respectively. This is because the voice and VC applications are delay-sensitive, and reduced TTI helps to reduce the E2E delay.

Table 3.5 shows the settings of various QoS classes for different applications under the UMTS network. The QoS classes for voice and VC applications are set as conversational because these applications have higher priority. The QoS class for VS application is set as background as the application has been assigned a lower priority in this scenario. For voice application, the uplink and downlink data rate is set to 64

---

**Table 3.4 UMTS RLC Layer Parameter Details**

<table>
<thead>
<tr>
<th>RLC Layer</th>
<th>Parameters</th>
<th>Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Conversational</td>
</tr>
<tr>
<td>Uplink Transport Channel (TrChnl)</td>
<td>Transmission Time Interval (TTI)</td>
<td>10</td>
</tr>
<tr>
<td></td>
<td>Type of Channel Coding</td>
<td>Convolutional</td>
</tr>
<tr>
<td></td>
<td>Coding Rate</td>
<td>Rate 1/2</td>
</tr>
<tr>
<td></td>
<td>Cyclic Redundancy Check (CRC) size (bits)</td>
<td>16</td>
</tr>
<tr>
<td>Downlink Transport Channel (TrChnl)</td>
<td>Transmission Time Interval (TTI)</td>
<td>10</td>
</tr>
<tr>
<td></td>
<td>Type of Channel Coding</td>
<td>Convolutional</td>
</tr>
<tr>
<td></td>
<td>Coding Rate</td>
<td>Rate 1/2</td>
</tr>
<tr>
<td></td>
<td>Cyclic Redundancy Check (CRC) size (bits)</td>
<td>16</td>
</tr>
<tr>
<td></td>
<td>Transmission Window Size</td>
<td>32</td>
</tr>
<tr>
<td></td>
<td>Receiving Window Size</td>
<td>32</td>
</tr>
<tr>
<td></td>
<td>RLC Mode</td>
<td>UM</td>
</tr>
<tr>
<td>RLC Discard</td>
<td>Move Receiving Window (MRW) (milliseconds)</td>
<td>900</td>
</tr>
<tr>
<td></td>
<td>Timer Discard (milliseconds)</td>
<td>7500</td>
</tr>
<tr>
<td></td>
<td>MAX MRW</td>
<td>6</td>
</tr>
<tr>
<td></td>
<td>MAX Number of retransmissions of a PU before a SDU is discarded</td>
<td>4</td>
</tr>
</tbody>
</table>
kbps. This is set according to the voice encoder setting requirements. For, the VC session, the data rate is set according to the frame rates per second. Table 3.6 shows the admission control parameters for the conversational and background classes. The uplink other-cell interference is modelled with a Frequency Reuse Factor Fe. Fe is defined as the ratio of the same-cell received power at the base station to the total received power (same-cell + other-cell), i.e. \( Fe = 1 / (1 + f) \). \( f \) is defined as the ratio of other-cell interference to same-cell interference. Typical values of \( f \): \( f = 0.75 \) (vehicular & pedestrian) \( f = 1.17 \) (indoor). Voice activity factor is set to 0.5. For background class, this value is set to 1.0 as per the standard requirement. Table 3.7 shows the DCH channel factors.

**Table 3.5 QoS Class Configuration for Different Applications**

<table>
<thead>
<tr>
<th>Client type</th>
<th>Parameter name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice clients</td>
<td>QoS class</td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>Downlink bit rate (kbps)</td>
<td>64</td>
</tr>
<tr>
<td></td>
<td>Uplink bit rate (kbps)</td>
<td>64</td>
</tr>
<tr>
<td>VS</td>
<td>QoS class</td>
<td>Best Effort/Background</td>
</tr>
<tr>
<td></td>
<td>Downlink bit rate (kbps)</td>
<td>144</td>
</tr>
<tr>
<td></td>
<td>Uplink bit rate (kbps)</td>
<td>144</td>
</tr>
<tr>
<td>VC</td>
<td>QoS class</td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>Downlink bit rate (kbps)</td>
<td>144,250</td>
</tr>
<tr>
<td></td>
<td>Uplink bit rate (kbps)</td>
<td>144,250</td>
</tr>
<tr>
<td>General configuration</td>
<td>Maximum SDU size (bytes)</td>
<td>1500</td>
</tr>
<tr>
<td></td>
<td>Block error ratio</td>
<td>1/100</td>
</tr>
</tbody>
</table>
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Table 3.8 shows the configuration settings used in the WiMAX network for different service classes. For the Gold (UGS) class, the maximum sustained traffic rate is 5 Mbps and for the Gold (ertPS) class this rate is 96 Kbps second. In the simulations, the gold (UGS) class is used for video and silver class for voice. The maximum sustained rate for the silver class is 1 Mbps. The physical profile (PHY) parameters used for the simulations are stated in Table 3.9. These are set according to the standard service requirements. Table 3.10 shows the WLAN parameters used for the simulations. Table 3.11 shows the LTE parameters used for the simulations. The QoS class identified for voice and video applications are set as Guaranteed Bit Rate (GBR) with a priority 1. For the streaming application, this is set as non-Guaranteed Bit Rate (non-GBR) with priority 6.

### Table 3.6 Uplink and Downlink Configuration

<table>
<thead>
<tr>
<th>Client type</th>
<th>Parameter name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice clients</td>
<td>QoS class</td>
<td>Best effort/Conversational</td>
</tr>
<tr>
<td></td>
<td>Downlink bit rate (kbps)</td>
<td>64</td>
</tr>
<tr>
<td></td>
<td>Uplink bit rate (kbps)</td>
<td>64</td>
</tr>
<tr>
<td>Video clients</td>
<td>QoS class</td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>Downlink bit rate (kbps)</td>
<td>144, 250, 250</td>
</tr>
<tr>
<td></td>
<td>Uplink bit rate (kbps)</td>
<td>64, 64, 120</td>
</tr>
<tr>
<td>General configuration</td>
<td>Maximum SDU size (bytes)</td>
<td>1500</td>
</tr>
<tr>
<td></td>
<td>Block error ratio</td>
<td>1/100</td>
</tr>
</tbody>
</table>
### Table 3.7 DCH Channel Factors

<table>
<thead>
<tr>
<th>DCH</th>
<th>Conversational</th>
<th>Background</th>
</tr>
</thead>
<tbody>
<tr>
<td>Uplink Reuse Efficiency factor</td>
<td>1.17/0.75</td>
<td>1.17/0.75</td>
</tr>
<tr>
<td>Voice Activity factor</td>
<td>0.5</td>
<td>1.0</td>
</tr>
<tr>
<td>Average sector links per user</td>
<td>2.0</td>
<td>1.3</td>
</tr>
</tbody>
</table>

### Table 3.8 Wimax Service Class Parameters

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Service Classes and Parameter value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service Classes</td>
<td>Gold Gold (ertPS) Silver Bronze Basic Control Class</td>
</tr>
<tr>
<td>Scheduling Type</td>
<td>UGS ertPS rtPS Best Effort (BE) BE ErtPS</td>
</tr>
<tr>
<td>Maximum Sustained Traffic Rate</td>
<td>5 Mbps 96 Kbps 1 Mbps 384 Kbps 32 Kbps 32 Kbps</td>
</tr>
<tr>
<td>Minimum Reserved Traffic Rate</td>
<td>1 Mbps 96 Kbps 0.5 Mbps 384 Kbps 32 Kbps 32 Kbps</td>
</tr>
<tr>
<td>Maximum Latency (milliseconds)</td>
<td>30.0 10.0 30.0 30.0 30.0 30.0</td>
</tr>
<tr>
<td>Parameter</td>
<td>Value</td>
</tr>
<tr>
<td>-----------------------------------------------</td>
<td>----------------</td>
</tr>
<tr>
<td>Frame Duration (milliseconds)</td>
<td>5</td>
</tr>
<tr>
<td>Symbol Duration (microseconds)</td>
<td>100.8</td>
</tr>
<tr>
<td>Number of Subcarriers</td>
<td>2048</td>
</tr>
<tr>
<td>Frame Preambles</td>
<td>1</td>
</tr>
<tr>
<td>Transmit/Receive Transition Gap (TRG) (microseconds)</td>
<td>106</td>
</tr>
<tr>
<td>Receive/transmit transition gap (RTG) (microseconds)</td>
<td>60</td>
</tr>
<tr>
<td>Boundary Position</td>
<td>Fixed</td>
</tr>
<tr>
<td>UL Subframe Size</td>
<td>12</td>
</tr>
<tr>
<td>Duplexing Technique</td>
<td>Time Division</td>
</tr>
<tr>
<td></td>
<td>Multiplexing (TDD)</td>
</tr>
<tr>
<td>Base Frequency</td>
<td>5 GHz</td>
</tr>
</tbody>
</table>

**Table 3.10 WLAN Parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Physical characteristics</td>
<td>Direct Sequence</td>
</tr>
<tr>
<td>Data Rate (bps)</td>
<td>11</td>
</tr>
<tr>
<td>Transmit Power</td>
<td>0.005</td>
</tr>
<tr>
<td>Short Retry Limit</td>
<td>7</td>
</tr>
<tr>
<td>Long Try Limit</td>
<td>4</td>
</tr>
<tr>
<td>AP Beacon Interval (secs)</td>
<td>0.02</td>
</tr>
<tr>
<td>Max Receive Lifetime (secs)</td>
<td>0.5</td>
</tr>
<tr>
<td>Buffer Size (bits)</td>
<td>256000</td>
</tr>
</tbody>
</table>

**Table 3.11 LTE Profile Configuration**

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>QoS Class Identifier (Voice/Video)</td>
<td>1 (GBR)</td>
</tr>
<tr>
<td>QoS Class Identifier (Streaming)</td>
<td>6 (non-GBR)</td>
</tr>
<tr>
<td>Uplink Guaranteed Bit Rate (bps)</td>
<td>1 Mbps</td>
</tr>
<tr>
<td>Downlink Guaranteed Bit Rate (bps)</td>
<td>1 Mbps</td>
</tr>
<tr>
<td>Uplink Maximum Bit Rate (bps)</td>
<td>1 Mbps</td>
</tr>
<tr>
<td>Downlink Maximum Bit Rate (bps)</td>
<td>1 Mbps</td>
</tr>
<tr>
<td>UL Base Frequency (GHz)</td>
<td>1920 MHz</td>
</tr>
<tr>
<td>UL Bandwidth (MHz)</td>
<td>20 MHz</td>
</tr>
<tr>
<td>UL Cyclic Prefix Type</td>
<td>7 symbols per slot</td>
</tr>
<tr>
<td>DL Base Frequency (GHz)</td>
<td>2110 MHz</td>
</tr>
<tr>
<td>DL Bandwidth (MHz)</td>
<td>20 MHz</td>
</tr>
<tr>
<td>DL Cyclic Prefix Type</td>
<td>7 symbols per slot</td>
</tr>
</tbody>
</table>
3.5 Simulation Result Analysis

The simulation scenarios are designed to investigate the impact of various factors on the performance of network-based applications. The factors, which are considered, are different environments, the presence of different traffic types in each RAN, and the number of active users in each RAN. Thus, the designed scenarios can be divided into the following groups:

**Group A:** This group of scenarios is designed in order to study the effect of environmental factors on the performance of different applications. To achieve this, the following scenarios are simulated:

- Different numbers of voice calls are simulated in the rural outdoor, the urban outdoor, the urban indoor and the urban pedestrian environments. For the urban outdoor environment, the scenarios are simulated based on small and medium city environments. The technologies, which are chosen for these scenarios, are UMTS, WiMAX, and LTE.

- In the second phase of simulations, some scenarios are simulated with a different number of voice calls and VS clients in the contexts mentioned above. The same communication technologies are used for these simulations.

**Group B:** The focus of these scenarios is to investigate the impact of different traffic types and the number of users on the application and network performance. To achieve this goal, UMTS, LTE, WLAN, and WiMAX-based homogeneous and heterogeneous networks with a different number of voice, VC, and VS users are examined.

**Group C:** The focus of this group of simulation scenarios is to find out the influence of different communication technologies on the performance of the overall network and applications. To achieve this, different communication technology-based networks are simulated with the same type of applications and number of users.
3.5.1 Environmental Factors

This section presents the simulation result analysis from the environmental factors perspectives. In the first stage of the simulations, a UMTS network with 12 simultaneous voice calls is simulated. The scenario is simulated in a different type of environment. Figure 3.4 shows the average packet loss of voice calls in various contexts. The calls for the outdoor rural area experience the highest amount of packet loss due to the presence of fewer base stations resulting in poor signal coverage. The lowest amount of loss is experienced in the urban indoor as the coverage is comparatively better in this type of settings. The callers in the medium and the small
urban cities experience almost the same amount of packet loss as expected. The calls on the urban pedestrian environment experience almost 2.67% more average packet loss than the calls in the indoor environment. This is due to the presence of fewer base stations compared to high mobile settings in an urban pedestrian environment.

![Average Packet Loss for 20 Voice Calls in the UMTS Network](image)

**Figure 3.6** Average Packet loss for 20 Voice Calls in the UMTS Network

![Average End-to-end Delay for 12 Voice Calls in the UMTS Network](image)

**Figure 3.7** Average End-to-end delay for 12 Voice Calls in the UMTS Network
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The calls in the rural outdoor experience almost 9.81% more average packet loss than the calls in the small city environment. This is due to poor signal coverage in the rural outdoor environment.

To investigate this huge gap, a call-by-call analysis is conducted for the calls in the rural outdoor. Figure 3.5 shows this analysis. It is apparent that the sixth call experiences the highest amount of packet loss that makes the average packet loss higher for all the calls. The reason behind this call to experience a large amount of packet loss is due to the callers having poor signal coverage.

Figure 3.6 shows the average E2E delay that the same callers experience in different environments. In this case, the medium city outdoor calls experience the highest amount of delay, and the small city outdoor calls experience the lowest amount of delay. The rural outdoor calls experience 76.32% more delay than the calls in the small city outdoor environment. The calls in the urban indoor and the pedestrian environment experience the same amount of delay. These results vary due to signal coverage, high mobile environment, and volume of traffic.

Figure 3.7 shows the average packet loss for 20 simultaneous voice calls in the UMTS network for different environments. In this case, the rural outdoor calls experience the highest amount of packet loss and the medium city outdoor calls experience the lowest amount of average packet loss. The small city outdoor calls experience 0.47% more packet loss than the medium city outdoor calls. In the mixed small city urban environment, when the calls take place between the outdoor and indoor office environment experience 5.68 % and 6.61% more packet loss than the urban indoor calls and the urban outdoor calls respectively. The rural outdoor calls experience 15.55% more packet loss than the small city outdoor calls.
A call-by-call analysis is also conducted in this environment to investigate the reason for this huge packet loss. The analysis shows that few calls experience a large amount of packet loss, which takes the average to a higher value. The reason behind this huge loss is due to some mobile stations being away from the base station. Figure 3.8 shows the average E2E delay experienced by these calls. The rural outdoor calls experience the highest amount of delay, and the urban mixed environment calls experience the lowest amount of delay. The E2E delay experienced by the urban

![Figure 3.8 Average E2E delay for 20 Voice Calls in the UMTS Network](chart)

**Figure 3.8 Average E2E delay for 20 Voice Calls in the UMTS Network**

Figure 3.8 shows the average E2E delay experienced by these calls. The rural outdoor calls experience the highest amount of delay, and the urban mixed environment calls experience the lowest amount of delay. The E2E delay experienced by the urban

![Figure 3.9 Average Packet loss for 20 Simultaneous Calls in Urban Environments](chart)

**Figure 3.9 Average Packet Loss for 20 Simultaneous Calls in Urban Environments**

![Average Packet Loss (%)](chart)
mixed environment calls show a contradictory behaviour to packet loss for the same calls.

Figure 3.9 shows the average packet loss experienced by calls in different urban environments. The calls in the medium city (MC) outdoor environment undergo the lowest amount of packet loss. The pedestrian environment calls experience around 0.21% more packet loss than the calls in the urban indoor environment. The urban indoor calls experience around 1.4% and 0.93% more packet loss than the small and the medium city outdoor calls respectively.

In the next stage of the simulations, the values of QoS-related parameters are analysed for the VS clients. The VS clients are placed on a UMTS network in the rural outdoor area, and the server is on a WiMAX network in an urban outdoor area. Figure 3.10 shows the packet loss that the clients experience in different environments. The second VS client in the rural outdoor environment experiences a 0.22% more packet loss than the VS client in the urban outdoor. The third VS client in the rural outdoor environment experiences a 0.31% packet loss than the client in the urban outdoor. These clients experience almost the same packet loss for the urban indoor and the urban outdoor environments.

Figure 3.10 Packet loss for VS Clients in the UMTS-WiMAX Network
Then a UMTS network is simulated with 20 voice calls and one VS client. The VS client and the VS server are placed in different environments to analyse their performance. Table 3.12 shows these results for the mixed environments. When the server is placed in the urban outdoor environment, the users in the rural outdoor environment experience a 0.11% more packet loss than the urban outdoor users. When the server is in the rural outdoor environment, and the client is in the urban indoor environment, the client experiences a 0.54% more packet loss than when the server was in the urban outdoor environment. When the server is placed in a suburban outdoor environment and the clients are in the rural outdoor environment, the client experience 0.49% more packet loss than when the server was in the urban outdoor environment.

**Table 3.12 QoS Parameters for VS Clients**

<table>
<thead>
<tr>
<th>Path loss model</th>
<th>Average packet loss (%)</th>
<th>Average end-to-end delay (msec)</th>
<th>Average delay variation (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Client side</td>
<td>Server side</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rural outdoor</td>
<td>Urban outdoor</td>
<td>5.72</td>
<td>314.5</td>
</tr>
<tr>
<td>Urban outdoor</td>
<td>Urban outdoor</td>
<td>5.61</td>
<td>314.2</td>
</tr>
<tr>
<td>Urban Indoor</td>
<td>Rural outdoor</td>
<td>6.15</td>
<td>314.5</td>
</tr>
<tr>
<td>Rural outdoor</td>
<td>Suburban outdoor</td>
<td>6.21</td>
<td>314.4</td>
</tr>
</tbody>
</table>
In the next scenario, a UMTS only network is simulated with three VS clients and one VS server in the mixed environments. Figure 3.11 shows the packet loss for these VS sessions. The VS clients in the rural outdoor environment experience the highest amount of packet loss and the calls in the medium city outdoor show the lowest amount of packet loss as expected. The VS users in the rural outdoor environment face a 2.65% more packet loss than the clients in the small city outdoor environment. On the other hand, for the urban indoor environment users, this amount is a 0.45% and 1.42% less than the medium and small city outdoor users respectively. In the case of E2Edelay, the users in the urban small city outdoor experience a 13% less delay than the rural outdoor users. The urban indoor users experience a 140% less E2Edelay than the medium city outdoor clients. Figure 3.12 shows the data of E2Edelay for these VS sessions. As stated earlier, the quality of the calls and video sessions vary due to variant signal coverage, geographical settings and volume of traffic.

**Figure 3.11 Packet loss for VS Clients in the UMTS Network**
Then, LTE networks in a rural area with hills and trees and an urban outdoor environment have been simulated. The applications, which are considered in those networks, are the voice, VC and VS. The VS server is placed in an urban environment. Figure 3.13 outlines the packet loss for VC in the LTE rural and urban network. The VC participants in the rural environment have a 0.92% more packet loss than the urban participants. In the case of delay, they experience 15.38% more delay than the urban participants. Figure 3.14 shows this data. Then the average packet losses for VS clients in those two environments are compared. The users in the rural outdoor settings experience a 0.07% more packet loss than the urban users as expected. In the case of voice calls, the calls in the urban outdoor experience a 0.61% more packet loss than the calls in the rural outdoor. This was an unexpected result. Then the investigations of these results show that the more loss happened because the callers in the urban environment were in a moving vehicle. Figure 3.15 and 3.16 show the data from these simulations.

**Figure 3.12 Average E2E delay for VS Clients in the UMTS Network**
3.5.2 Technology and Service-related Factors
In this section, the impacts of technology, service, and the number of user-related factors on the application and network performance are analysed. Figure 3.17 shows the comparison of average packet loss for different number of voice calls. For eight new calls in the network, the callers in the rural outdoor environment experience 8.91% more average packet loss. In the small city outdoor environments, this loss is increased by 2.54%. Table 3.13 presents the percentage of increased packet loss for each environment.

![Bar chart showing the impact of number of users on average packet loss in different environments.](image)

**Figure 3.16 Impact of Number of Users on Average Packet loss in Different Environments**

<table>
<thead>
<tr>
<th>Environment</th>
<th>Increase of Packet loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rural Outdoor</td>
<td>8.91</td>
</tr>
<tr>
<td>Urban SC: Outdoor</td>
<td>2.54</td>
</tr>
<tr>
<td>Urban: Indoor</td>
<td>3.99</td>
</tr>
<tr>
<td>Urban:Pedestrian</td>
<td>1.53</td>
</tr>
<tr>
<td>Urban MC: Outdoor</td>
<td>2.17</td>
</tr>
</tbody>
</table>

A few scenarios are simulated with different number of users in the context of rural area. Figure 3.18 shows the average packet loss for a range of users. When the number of users is 14, the average packet loss is increased by 0.88% compared to when the number of users is eight. For ten more users, this packet loss is increased...
by 1.17% and with 14 more users, this packet loss is increased by 3.24%. For 24 more new users, the packet loss is increased by 13.78%.

![Figure 3.17 Impact of Number of Users on Average Packet loss](image)

**Figure 3.17** Impact of Number of Users on Average Packet loss
Table 3.14 shows the effect of VS traffic on the performance of voice calls. When there are 14 voice clients and one VS client in the network, the voice calls experience 1.46% more packet loss compared to when there were no VS client in the network. When there are 18 voice users and one VS client, the voice users experience a 4.83% more packet loss compared to the presence of only 18 voice clients in the network. In this simulation, the VS clients use the MPEG-4 video codec.

**Table 3.14 Impact of VS Clients on the Performance of Voice Clients**

<table>
<thead>
<tr>
<th>Number of Voice Clients</th>
<th>Number of VS clients</th>
<th>Average Packet Loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>0</td>
<td>2.06</td>
</tr>
<tr>
<td>8</td>
<td>1</td>
<td>2.64</td>
</tr>
<tr>
<td>14</td>
<td>0</td>
<td>2.94</td>
</tr>
<tr>
<td>14</td>
<td>1</td>
<td>4.4</td>
</tr>
<tr>
<td>18</td>
<td>0</td>
<td>3.23</td>
</tr>
<tr>
<td>18</td>
<td>1</td>
<td>8.06</td>
</tr>
</tbody>
</table>
Then, the impact of different codecs on the performance of voice and video applications are analysed. In order to do that the scenarios with voice calls using the same voice codec and the different voice codecs are simulated. Some scenarios are simulated with the VS clients using different video codecs. In this way, the effects of various video codecs on the performance of voice calls are investigated. Figure 3.19 shows the number of voice clients vs. the average packet loss for two separate codecs used for VS client. The voice calls experience more packet loss in the presence of MPEG-4-based VS client. This is because the MPEG-4 codec consumes more bandwidth than H.263 codec. In addition, it is noticeable that packet loss has nearly become double when the number of calls has been increased from 16 to 18. This is due to the mobility of some of the users in that specific scenario. During the ongoing calls, some of the callers moved far away from the base station and as a result the statistics indicate more packet loss. Table 3.15 shows the packet loss experienced by VS clients for the MPEG-4 and the H.263 video codec for a different number of users.

In the next stage, the effects of various communication technologies on the performance of different applications are investigated. In order to do that, a diverse range of scenarios based on UMTS, WiMAX, and LTE access technologies are compared. Figure 3.20 shows the comparison results of one such scenario. It shows the average packet loss experienced by the VS client in different environments and communication technologies. The UMTS-WiMAX urban outdoor environment
experiences the highest amount of average packet loss. The LTE urban outdoor VS client experiences a 0.61% less packet loss than the UMTS outdoor VS client. The UMTS rural outdoor VS client experience 5.06% more average packet loss than the LTE rural outdoor client does.

![Average Packet Loss (%)](image)

**Figure 3.19 Average Packet loss for VS Clients in various RANs**

In the next set of simulations, the values of QoS performance parameters are compared for 32 voice users in two different cases, these are: the voice calls having the same voice codec vs. the voice calls having different voice codec. Table 3.16 shows the results. Using GSM Full Rate (FR) codec the average packet loss for the voice calls is 15.84%. The parallel use of GSM FR and IS-641 codec decreases the packet loss by 3.06%.

Then, the performances of different applications in various network architectures are

<table>
<thead>
<tr>
<th>Codec</th>
<th>Average Packet Loss (%)</th>
<th>Average E2EDelay (msec)</th>
<th>Average Jitter (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Different Voice Codec (GSM FER, IS-641)</td>
<td>12.78</td>
<td>208</td>
<td>50</td>
</tr>
<tr>
<td>Same Voice Codec (GSM FER)</td>
<td>15.84</td>
<td>210</td>
<td>56</td>
</tr>
</tbody>
</table>
compared. In the first phase of these simulations, a UMTS network with three hexagonal cells, each covering one kilometre of the area is simulated. A three-sector base station is placed in the joint of the three cells to cover this three-kilometre area.

There are eight voice clients, which are communicating with each other within the UMTS cells. There are three VC participants, which are communicating through the backbone network with a VC participant, which is in the WiMAX access network. There are three WLAN hotspots in each of the UMTS cells. Each of the hotspot has one VC participant. The UMTS-WLAN clients are in a tight-coupling architecture (WLAN gateway-GGSN). Different uplink and downlink bit rate have been configured for the present clients in the network that are stated in Table 3.6 in section 3.4. For voice clients, Best Effort (BE) service class is used for local voice (within UMTS cells) transmissions, and interactive voice has been setup for the voice transmissions in VC sessions. These application configurations are illustrated in Table 3.16.

<table>
<thead>
<tr>
<th>Client type</th>
<th>Parameter name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice</td>
<td>QoS class</td>
<td>Interactive voice/Best effort</td>
</tr>
<tr>
<td></td>
<td>Encoder scheme</td>
<td>G.729 A (silence)</td>
</tr>
<tr>
<td></td>
<td>Silence length</td>
<td>0.65 seconds</td>
</tr>
<tr>
<td></td>
<td>Talk spurt length</td>
<td>0.352 seconds</td>
</tr>
<tr>
<td>Video</td>
<td>QoS class</td>
<td>Interactive multimedia</td>
</tr>
<tr>
<td></td>
<td>Frame size (bytes)</td>
<td>600</td>
</tr>
<tr>
<td></td>
<td>Frame rate (frames/Sec)</td>
<td>10</td>
</tr>
<tr>
<td></td>
<td>Block error ratio</td>
<td>1/100</td>
</tr>
</tbody>
</table>

Followed by the previous one, another scenario is simulated with one WLAN hotspot, which has one VC participant instead of three WLAN hotspots and three VC participants. Figure 3.21 shows the comparisons of packet loss for video transmissions when the transmission goes through the UMTS network (no WLAN
hotspot), one WLAN hotspot and three WLAN hotspots. The results show that the video transmissions experience more packet loss when the traffic goes through the UMTS network rather than the WLAN.

![Packet loss for Video Transmissions](image)

**Figure 3.20 Packet loss for Video Transmissions**

### 3.5.3 Simulation Results and the Benchmark Values

In this section, the values of QoS-related parameters, which are received from simulations, are compared with the benchmark values previously discussed in Section 3.3. Figure 3.22 shows the packet loss from 12 voice calls placed in different environments. The figure also demonstrates the comparison of performance results to various recommendations [15, 154, 155, 159]. Figure 3.23 shows the performance of 12 voice calls based on the evaluation of E2E delay. The calls in the medium city outdoor experience a larger delay than the other environments. This was unexpected. However, the call-by-call analysis shows that this is due to a single call experiencing a larger delay, which takes the average delay to a higher value. The user related to that call was away from the base station.
Chapter 3: Application-based QoS Evaluation

Figure 3.21 Performance of Voice Calls based on Packet loss

![Packet Loss Chart]

<table>
<thead>
<tr>
<th>Environment</th>
<th>Packet Loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Urban: Small City</td>
<td>3.2%</td>
</tr>
<tr>
<td>Urban: Pedestrian</td>
<td>5.35%</td>
</tr>
<tr>
<td>Urban Medium City: Outdoor</td>
<td>3.1%</td>
</tr>
<tr>
<td>Urban: Indoor</td>
<td></td>
</tr>
<tr>
<td>Rural: Outdoor</td>
<td>12.38%</td>
</tr>
</tbody>
</table>

Acceptable values:
- ITU: 3%
- 3GPP: 3%
- Cisco: 1%
- [9]: 1%

The ITU E-model is used to calculate the mean opinion score (MOS) and R factor, which express the overall transmission quality rating for the voice calls. The E-model does not have any option to use jitter for the calculations. Therefore, the values of packet loss and delay are used for these calculations. Table 3.18 shows the analysis. The QoE derived from the MOS and R factor shows that in the rural outdoor environment, nearly all users are dissatisfied with the transmission quality. However,
the call-by-call analysis demonstrates a slightly different overview. According to QoE analysis of call-by-call, three calls out of six indicate that the users are satisfied. The QoE analyses of two calls out of six show that nearly all users are dissatisfied (e.g.: 1st and 6th call). The QoE analysis of the fourth call indicates that some users are dissatisfied. The reasons for the poor quality of the 1st and 6th calls are a high delay and packet loss. The callers were away from the base station and experienced poor signal. On the other hand, the 2nd and 3rd call experience a lower delays and packet losses. Table 3.19 shows this analysis.

**Table 3.18 MOS and R-Factor Analysis for Voice Calls in the UMTS Network**

<table>
<thead>
<tr>
<th>Environment</th>
<th>Delay</th>
<th>Packet loss</th>
<th>MOS</th>
<th>R Factor</th>
<th>QoE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Urban: Small City</td>
<td>190</td>
<td>3.2</td>
<td>4.16</td>
<td>83.7</td>
<td>Satisfied</td>
</tr>
<tr>
<td>Urban: Pedestrian</td>
<td>200</td>
<td>5.35</td>
<td>2.42</td>
<td>47.1</td>
<td>Nearly all users dissatisfied</td>
</tr>
<tr>
<td>Urban: Medium City</td>
<td>585</td>
<td>3.1</td>
<td>4.04</td>
<td>80.5</td>
<td>Satisfied</td>
</tr>
<tr>
<td>Urban: Indoor</td>
<td>200</td>
<td>2.68</td>
<td>4.14</td>
<td>83.1</td>
<td>Satisfied</td>
</tr>
<tr>
<td>Rural: Outdoor</td>
<td>335</td>
<td>12.38</td>
<td>2.9</td>
<td>56.2</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>

**Table 3.19 Call-by-Call Analysis in the UMTS Network**

<table>
<thead>
<tr>
<th>Call Number</th>
<th>Delay</th>
<th>Packet loss</th>
<th>MOS</th>
<th>R Factor</th>
<th>QoE</th>
</tr>
</thead>
<tbody>
<tr>
<td>1st call</td>
<td>455</td>
<td>6.05</td>
<td>2.72</td>
<td>52.8</td>
<td>Nearly all users dissatisfied</td>
</tr>
<tr>
<td>2nd call</td>
<td>190</td>
<td>2.005</td>
<td>4.23</td>
<td>85.9</td>
<td>Satisfied</td>
</tr>
<tr>
<td>3rd call</td>
<td>190</td>
<td>2.005</td>
<td>4.23</td>
<td>85.9</td>
<td>Satisfied</td>
</tr>
<tr>
<td>4th call</td>
<td>240</td>
<td>2.065</td>
<td>4.00</td>
<td>79.5</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>5th call</td>
<td>155</td>
<td>2.01</td>
<td>4.30</td>
<td>88.3</td>
<td>Satisfied</td>
</tr>
<tr>
<td>6th call</td>
<td>710</td>
<td>16.55</td>
<td>1.60</td>
<td>29.8</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>

3.6 Summary

In this chapter, the concepts of application-based QoS evaluation approach for heterogeneous networks have been discussed. A heterogeneous network involves a diverse range of radio access networks. Each technology has its own QoS characteristics. One way to deal with this situation is to evaluate the performance of
any access network according to the performance of the applications utilising it. Because the applications have the same QoS requirement regardless of the access network it is using. The idea of a unified QoS metric for performance evaluation of these networks is also outlined. At first, the key QoS-related parameters of each application have been selected through available recommendations. The simulation results analysis shows that in the wireless networks the application performance is affected by different factors. These factors could be the environment, technology, network architecture, or traffic related. The results also illustrate that it is difficult to define any acceptable fixed value for these parameters due to the involvement of multiple factors. Therefore, an acceptable range is more suitable which would consider all relevant factors.

In Section 3.3, some of the experimental data and recommendations regarding QoS-related parameters of different applications have been discussed. Based on those discussions and the simulation results presented in Section 3.4, the acceptable ranges

<table>
<thead>
<tr>
<th>Applications</th>
<th>Packet loss (%)</th>
<th>Delay (msec/sec)</th>
<th>Jitter (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>LB</td>
<td>UB</td>
<td>LB</td>
</tr>
<tr>
<td>Voice</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rural</td>
<td>3</td>
<td>20</td>
<td>150 msec</td>
</tr>
<tr>
<td>Urban</td>
<td>1</td>
<td>3</td>
<td>100 msec</td>
</tr>
<tr>
<td>VS</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rural</td>
<td>3</td>
<td>5</td>
<td>5 sec</td>
</tr>
<tr>
<td>Urban</td>
<td>1</td>
<td>3</td>
<td>1 sec</td>
</tr>
<tr>
<td>VC</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rural</td>
<td>2.5</td>
<td>5</td>
<td>100 msec</td>
</tr>
<tr>
<td>Urban</td>
<td>1</td>
<td>2.5</td>
<td>100 msec</td>
</tr>
</tbody>
</table>

for these parameters in each application are defined. These are given in the Table 3.20 for each parameter with an Upper Bound (UB) and a Lower Bound (LB) value. Lower Bound (LB) means the minimum acceptable value for a certain metric under normal conditions. For example, the performance metric will be the same for packet loss being anything below or equal 3%. These ranges are used in the later chapters to design the proposed application-based QoS evaluation methods.

In the fourth and the fifth chapter of this thesis, a performance metric value is calculated for the QoS-related parameters using these ranges. These ranges could be
also divided into certain categories, such as high, average, poor, etc. Using this concept, in Chapter 6, these ranges are expressed as fuzzy membership functions. As a continuation of this work, in the next chapter, a fixed weight-based method for QoS evaluation of heterogeneous networks is proposed and evaluated.
CHAPTER 4

Fixed Weight-based QoS Analysis

A typical heterogeneous wireless network consists of different types of wireless communication technologies, which have diverse characteristics. Generally speaking, the key performance parameters for these technologies are not directly comparable. For example, the delay ranges of UMTS and WLAN are entirely different. Therefore, a high value of delay measured from a WLAN may not be considered high in a UMTS environment. On the other hand, the applications running over them have the same QoS characteristics regardless of the communication technology they are utilising. Hence, in this context, an application-based QoS evaluation approach is more applicable than a communication technology-based performance evaluation approach.

In the previous chapter, a detailed analysis has been carried out, to select the key QoS-related parameters for each application and to define the acceptable ranges of these parameters. The concepts of application-based QoS evaluation approach and unified QoS metrics are also introduced in the previous chapter. In this chapter, using those concepts a fixed weight-based application-oriented QoS evaluation method is proposed which derives the unified QoS metric to assess network performance. This mechanism is evaluated using various simulation scenarios. The particular focus is on multimedia-based applications as these applications have distinct characteristics, which make their QoS evaluation method unique. The introductory concepts of this chapter are presented in Section 4.1. Section 4.2 discusses the detail steps of the proposed Fixed Weight-based QoS Evaluation Method (FWQEM). Section 4.3 illustrates the simulation scenarios in detail. Section 4.4 presents the simulation result analysis. The QoS analysis using the fixed weight-based method is outlined in Section 4.5. Finally, Section 4.6 summarises this chapter.
4.1 Introduction
Multimedia applications such as video streaming (VS), video conferencing (VC), and Internet Protocol television (IPTV) are widely used in different socioeconomic remote services. Because of this growing demand for multimedia-based services, the communication networks are experiencing a massive increase in the traffic they carry. According to Cisco forecasts, 66 percent of the global mobile data flow will consist of video traffic by 2015 [1]. To deal with the increased demand for throughput, the network operators are usually deploying heterogeneous networks as they can expand the network capacity in that way. Setting up WLAN hotspots on a 3G network is such an example. However, QoS provisions over this type of networks are still challenging due to the underlying characteristics, for instance, the channel capacity variations, or area coverage. The coverage issues are mostly experienced in rural areas and developing regions.

Hence, from network designing, planning and troubleshooting perspectives, a QoS evaluation method is required that can bring these domains into a common platform. From the analysis of Chapter 2, it is evident that the most current studies aim to evaluate the QoS of each application or radio access network individually. This approach is useful in case of homogeneous networks. However, for heterogeneous networks, the presence of multiple types of applications and access technologies...
make the assessment of the overall performance challenging. To deal with this situation, in this chapter, a QoS evaluation method is designed that can unify multiple performance evaluation parameters into a single measurement metric. The unified metrics measure the performance of different entities in a network such as applications and access networks. The method is suitable for simple networks where the applications and technologies are relatively fixed. This method does not consider application importance while QoS evaluation. This method can be used in several situations, they are discussed as follows.

A conventional heterogeneous network constitutes of different communication technologies and diverse types of applications. Figure 4.1 shows a typical heterogeneous network scenario. In this network, there are several radio access networks, such as UMTS, LTE, and WLAN. Each of these access networks has several active applications; for example, UMTS has a few active voice calls and a VC session. Some users of VC session are in the WLAN and LTE network. The QoS evaluation of this type of network can be very complex in some situations. For example, to measure the QoS of the UMTS network considering the performance of both voice calls and VC session is a challenging task. The QoS evaluation of the

Figure 4.2 Deploying the Network-based Service Model of a Developed Country in a Developing Country

![Diagram showing a network scenario with various access networks and applications.]
overall network configuration involving the performance of UMTS, LTE, and WLAN will be a difficult case. The method proposed in this chapter evaluates the QoS of the applications, and the radio access networks individually and derives a unified network configuration QoS metric for the whole network. Therefore, this QoS evaluation method can handle the above-discussed situation efficiently.

Figure 4.2 shows a scenario of a heterogeneous network-based service model. For example, there is a network-based distance education model in a developed country A. Another developing country B wants to deploy the same service model. The question is in the network designing and planning stage, how to analyse the satisfactory QoS level for the users in country B, for such a model. A unified metric can facilitate the evaluation in such cases. The goal of this unified metric is to include various constraints of the network to measure the network QoS with a single numerical value.

![Figure 4.3 Deploying a New Network](image)

The third case demonstrates a scenario where service operators want to deploy a health service-centric network in a rural area B. They are considering several available technologies, such as 3G and WiMAX. While planning the configuration, they would consider the number of active users for the applications, relevant to
health service and non-relevant to health service. It would be better from the planning perspective if they the achievable QoS level with such constraints and the ideal QoS level for this type of network beforehand. To analyse these situations, an analytical method that can evaluate the QoS level with a unified metric would be useful.

The unified QoS metric can also be useful to select a suitable access network. Although some classical access selection algorithms use an integrated QoS metric for this purpose, in this case, the considered situations are different though. Figure 4.3 shows the scenario of such an access network selection. For example, there are three networks: X, Y, and Z. These networks have the following specifications:

**Network X:**
Radio Access Network: UMTS, WLAN
Active users:
- Numbers of VC users: 5
- Numbers of VS clients: 25
- Numbers of Voice calls: 50

**Network Y:**
Radio Access Network: LTE, WLAN

![Figure 4.4 Access Network Selection](image)
Chapter 4: Fixed Weight-based QoS Analysis

Active users:

Numbers of VC users: 10
Numbers of VS clients: 20
Numbers of Voice calls: 40

Network Z:
Radio Access Network: LTE, WLAN, UMTS
Active users:

Numbers of VC users: 30
Numbers of VS clients: 40
Numbers of Voice calls: 40

Now if a user $A$ wants to join a video lecture, which network is the most suitable from his perspective? Or another user $B$ in a rural area $K$ wants to join a VC session with a doctor in an urban area $J$, and then which is the most suitable network from his perspective? Here arises the question of measuring QoE as well. To deal with these situations, a QoS evaluation method that can evaluate the overall network, RAN and application performance individually with single QoS metrics would be useful.

4.2 The Fixed Weight-based QoS Analysis

In this section, the steps of the proposed Fixed Weight-based QoS Evaluation Method (FWQEM) are discussed in detail. For assessment purpose, the heterogeneous network is divided into two separate entities: they are the active applications and RANs. The overall network configuration consists of both of these entities and works as an overlaying layer of these two entities. Figure 4.5 shows the conceptual diagram. A heterogeneous wireless network $N$ is considered, which has $j (j = 1, 2, 3, ..., n)$ number of radio access networks. Each of this access networks contains $i (i = 1, 2, 3, ..., m)$ number of applications.
In Chapter 3, the key QoS-related parameters of each application and their acceptable ranges have been identified. In FWQEM, those ranges are utilised to derive separate functions to calculate the application, the RAN and the network configuration QoS metric.
The variables, which are used for this method, are specified as follows:

QoS-related parameter: $QP$

Application: $A$

Radio Access Network: $R$

Network Configuration: $N$

A QoS-related parameter $QP_k$ in an Application $A$, in a RAN, $R$: $QP_{k,A,R}$, where $k$ is the index for the QoS-related parameters and $k = \{1,2,\ldots,p\}$.

The upper bound for the benchmark range of a $QP_k$ in any application $A$: $UB_{QP_k,A}$

The lower bound for the benchmark range of a $QP_k$ in an application $A$: $LB_{QP_k,A}$

The weight of a QoS-related parameter $QP_k$ in any application $A$: $W_{QP_k,A}$

The weight of any application $A$, in a RAN $R$: $W_A^R$

The weight of any RAN, $R$ in a network $N$: $W_R^N$

The performance metric for a QoS-related parameter $QP_k$: $P_{QP_k,A}$
The real-time performance metric for a QoS-related parameter $QP_k : RTP_{QP_{k,A}}$.

The application QoS metric for an application $A$, in a RAN, $R$: $QoSAM_{QA}^R$

The radio access network QoS metric for any RAN, $R$ in a Network $N$: $QoSRM_{R}^N$

Network Configuration QoS Metric: $QoSCM_N$

The steps of the proposed FWQEM method are presented in Figure 4.6.

- In this step, the weights for the key QoS-related parameters are determined. These weights are defined based on their impacts on the application performance. For example, jitter plays a crucial role in the performance evaluation of voice and VC applications; however, it does not affect the performance of VS applications. As a result, while assigning weight, the jitter is assigned a higher value for the VC and voice applications.

- In the second step, a performance metric is calculated for each application-related QoS parameter.

- These performance metrics and the weights of the QoS-related parameters are used to compute the QoS metric for each application.

- Then the RAN QoS metric is derived using the application QoS metrics and the weight of each application.

- The final network configuration QoS metric is calculated using the RAN QoS metrics and their weights.

The following sections present each of these steps in detail.

### 4.2.1 QoS-related Parameter Weights

The weights are determined based on the performance-related impacts that each QoS-related parameter has on any application. In Chapter 3, the key QoS-related parameters for the performance evaluation of the considered applications in this thesis have been discussed. In this chapter, to establish the weights of those parameters, a scale ranges between 0 and 1 is defined, 1 being the highest importance level and 0 being the lowest. The more impact the parameter has on any application performance, the more weight it is assigned. For example, to evaluate the performance of a voice application in a UMTS network, three QoS-related parameters, namely delay, jitter and packet loss are considered. The weights are defined in such a way that $W_{QP_{Delay, Voice}} + W_{QP_{Jitter, Voice}} + W_{QP_{Packet Loss, Voice}} = 1$. 

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Table 4.1 shows the weights for the QoS-related parameters of each considered application in this work. Jitter is not a significant parameter for the performance evaluation of VS applications. Therefore, a zero weight is assigned to jitter in VS applications. Delay and Jitter both equally affect the performance of voice and VC applications. As a result, a weight of 0.4 is assigned to these two parameters. On the other hand, packet loss has a minimal impact on the performance evaluation of these two applications as compared to delay and jitter. Therefore, a 0.2 weight is assigned to packet loss. The packet loss has the highest impact on the performance of VS application. Hence, a weight of 0.8 is assigned to packet loss in VS application.

### 4.2.2 The Performance Metric

In this step, a performance metric is calculated for each QoS-related parameter. In order to do that the context-based benchmark ranges presented in Chapter 3 are applied. Each range has an Upper Bound (UB) and a Lower Bound (LB). Table 4.2 shows these values. The performance evaluation parameter of a network can be divided into two categories; these are the benefit and the cost category. The parameters, which are preferred to have a maximum value, such as throughput, falls within the benefit category. On the other hand, the parameters, which are preferred to have minimal costs, are regarded as the cost type parameter. Delay, jitter, and packet loss all these QoS-related parameters fall within the cost category. In the proposed method, only the cost-based parameters are used for the performance evaluation of any application.
The function to derive the performance metric uses three parameters, which are: a real-time performance metric, the upper and lower bounds of the cost-based performance parameter that is to be evaluated. The real-time performance metric value is referred to as $RTP_{QP_{k,a,R}}$. This value is collected for each $QP_{k,a,R}$ from an ongoing session of an application $A$ in a RAN $R$ at some time interval. The function is as follows:

$$P_{QP_{k,a}} = \begin{cases} 
0, & RTP_{QP_{k,a}} \geq UB_{QP_{k,a}} \\
1, & RTP_{QP_{k,a}} \leq LB_{QP_{k,a}} \\
\frac{UB_{QP_{k,a}} - RTP_{QP_{k,a}}}{UB_{QP_{k,a}} - LB_{QP_{k,a}}}, & UB_{QP_{k,a}} > RTP_{QP_{k,a}} > LB_{QP_{k,a}} 
\end{cases} \quad (4.1)$$

For example, to evaluate the performance of an ongoing VC session in the UMTS access network, three QoS-related parameters, namely delay, jitter and packet loss are considered. The real-time performance metric for each of these parameters is expressed as $RTP_{QP_{k,a}}$, $RTP_{QP_{k,a}}$, and $RTP_{QP_{k,a}}$, where D, J, and PL refer to as delay, jitter, and packet loss respectively, and U refers to as UMTS network.

If $RTP_{QP_{k,a}} > UB_{QP_{k,a}}$, then $P_{QP_{k,a}} = 0$.
If $RTP_{Q_{P_{vc}}} < LB_{Q_{P_{vc}}}$ then $P_{Q_{P_{vc}}} = 1$

If $UB_{Q_{P_{vc}}} > RTP_{Q_{P_{vc}}} > LB_{Q_{P_{vc}}}$ then

$$P_{Q_{P_{pl,vc}}} = \frac{UB_{Q_{P_{vc}}}-RTP_{Q_{P_{vc}}}}{UB_{Q_{P_{vc}}}-LB_{Q_{P_{vc}}}}$$

### 4.2.3 The Application QoS Metric

In this step, a function is defined to derive an application QoS Metric for each active application in a specific RAN. This function combines the weights and the performance metric of the QoS-related parameters together. With $W_{Q_{P_{k,A}}}$ corresponding to the weight of the QoS-related parameter, $QP_k$ in an application $A$ and $P_{Q_{P_{k,A}}}$ representing the performance metric for the same, this will be:

$$QoSAM^R_A = \sum_{k=1}^{P} P_{Q_{P_{k,A}}} W_{Q_{P_{k,A}}}$$ (4.2)

where $k = \{1,2,...,p\}$ and 1 = Delay, 2 = Jitter etc. For example, there is a VC session running in a UMTS network. The application QoS metric for this VC session is evaluated as:

$$QoSAM^\text{UMTS}_{\text{VC}} = P_{Q_{P_{\text{D,vc}}}} W_{Q_{P_{\text{D,vc}}}} + P_{Q_{P_{\text{J,vc}}}} W_{Q_{P_{\text{J,vc}}}} + P_{Q_{P_{\text{PL,vc}}}} W_{Q_{P_{\text{PL,vc}}}}$$

where D = Delay, J = Jitter and PL = Packet Loss.

### 4.2.4 The RAN QoS Metric

In this step, a function is defined to calculate the QoS metric for each available RAN in the network. The function combines the values of the application QoS metrics and application weights to derive the QoS metric for a specific RAN. Suppose, there are $m$ numbers of applications ($A_1, A_2, ...., A_m$) in a RAN $R$ and the weight for each of this application is defined as ($W_{R_1, A_1}, W_{R_2, A_2}, ...., W_{R_m, A_m}$). An equal weight is distributed for the applications present in the access network, and these weights are assigned in such a way that: $\sum W_{R_1, A_i} = 1$. As stated earlier, the fixed weight-based method is designed
to handle the QoS evaluation of simple network scenarios. In this method, the priority of applications and radio access networks are not considered. Therefore, an equal weight is used for applications and radio access networks, assuming they all carry the same priority. The usage of priority-based weights for applications and radio access networks are discussed in the later chapter to handle the QoS evaluation of more complex network scenarios. The function in this case is expressed as follows:

\[
QoS_{RM}^N = \sum_{i=1}^{m} QoS_{AM}^R W^R_A
\]  \hspace{1cm} (4.3)

where \( i = \{1,2, \ldots, m\} \). For example, in LTE-based access network, there are two applications VC and voice. In this case, the RAN QoS metric in the LTE network is derived as:

\[
QoS_{RM}^{LTE} = QoS_{AM}^{VC} W^{LTE}_{VC} + QoS_{AM}^{Voice} W^{LTE}_{Voice}
\]

where \( A_1 = VC, A_2 = Voice \) and \( R=LTE \).

4.2.5 The Network Configuration QoS Metric

In this step, another function is defined to calculate the network configuration QoS metric. This function amalgamates the values of RAN QoS metrics to derive the unified QoS metric for the network configuration. Each RAN is assigned an equal weight in order to normalise the network configuration metric. Suppose, there are \( n \) number of radio access networks \( (R_1, R_2, \ldots, R_n) \) in a network \( N \), and the weight for each of this RAN is defined as \( (W_{R_1}^N, W_{R_2}^N, \ldots, W_{R_n}^N) \). In this case, the weights are assigned in such a way that \( \sum_{j=1}^{n} W_{R_j}^N = 1 \). The function is expressed as follows:

\[
QoS_{CM}^N = \sum_{j=1}^{n} QoS_{RM}^N W_{R_j}^N
\]  \hspace{1cm} (4.4)

where \( j = \{1,2,3, \ldots, n\} \). For example, in a network \( N \), there are two RANs: UMTS and LTE denoted as \( U \) and \( L \) respectively, the network configuration QoS Metric is calculated as:

\[
QoS_{CM}^N = QoS_{RM}^U W_U^N + QoS_{RM}^L W_L^N
\]

where \( R_1 = UMTS \) and \( R_2 = LTE \).
Chapter 4: Fixed Weight-based QoS Analysis

4.3 Simulation Scenarios

In this section, the setups for the simulation scenarios, which have been designed to evaluate the performance of the proposed QoS analysis method, are discussed. The simulation scenarios are divided into three major categories, and they are:

- UMTS/UMTS-WiMAX-based network scenarios
- LTE-UMTS/LTE-based network scenarios
- Other Heterogeneous network-based scenarios

Each of these categories has several sub-scenarios. The simulation setups for these technologies have been mostly discussed in Chapter 3. In this chapter, the details of some other parameters such as the number of active users, the number of applications, and the application setups are presented. It is assumed for each scenario that there is a network $N$ with $j$ numbers of RANs, and each RAN has $i$ numbers of active applications.

4.3.1 UMTS-WiMAX-based Heterogeneous Network Scenarios

A UMTS-WiMAX and a UMTS only network are considered with Voice, VC, and VS applications in these scenarios. The UMTS and the WiMAX model of OPNET are used for the simulations as these models are well established. The rural outdoor environment is chosen for these simulations. Some of the simulation parameters are stated in Table 4.3.
The highest bitrate chosen is 144 Kbps as in a rural area usually the bandwidth is lower than the UMTS standard bandwidth of 2 Mbps. In the first phase of the simulations, a UMTS network in the context of a rural area is simulated with various numbers of voice application users starting from 8 to 20. The values of packet loss, delay, and jitter are evaluated in each scenario. The scenarios are run for 13 times with different seed values and 95% confidence interval. In the second phase of the simulations, a VS application is added to the network. Two separate video codecs, H.263 and MPEG-4, have been used for the streaming application. The reasons for using two separate video codecs are twofold. Firstly, it will help to evaluate the performance of voice calls in combination with different codec based streaming applications. Secondly, the usage of different video codec will help to study the most suitable settings under such network scenarios. The VS server is placed in the WiMAX environment. The performance of the voice application is evaluated with each type of the streaming codec running. This assessment is conducted to identify

<table>
<thead>
<tr>
<th>Layers</th>
<th>Parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>QoS class</td>
</tr>
<tr>
<td></td>
<td>Payload size</td>
</tr>
<tr>
<td>Application</td>
<td></td>
</tr>
<tr>
<td>Voice calls</td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>224 bits</td>
</tr>
<tr>
<td>VS</td>
<td>Background</td>
</tr>
<tr>
<td>VC</td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>576 bytes</td>
</tr>
<tr>
<td>RLC</td>
<td>Mode</td>
</tr>
<tr>
<td></td>
<td>Unacknowledged</td>
</tr>
<tr>
<td>Timer MRW (msec)</td>
<td>900</td>
</tr>
<tr>
<td>Timer Discard (msec)</td>
<td>7500</td>
</tr>
<tr>
<td>MAX MRW</td>
<td>6</td>
</tr>
<tr>
<td>MAX DAT</td>
<td>4</td>
</tr>
<tr>
<td>PHY</td>
<td>Channel type</td>
</tr>
<tr>
<td></td>
<td>DCH</td>
</tr>
<tr>
<td></td>
<td>Conversational</td>
</tr>
<tr>
<td></td>
<td>Background</td>
</tr>
<tr>
<td>Bit rate (Kbps)</td>
<td>TTI (msec)</td>
</tr>
<tr>
<td></td>
<td>Bit rate (Kbps)</td>
</tr>
<tr>
<td></td>
<td>TTI (msec)</td>
</tr>
<tr>
<td>64</td>
<td>10</td>
</tr>
<tr>
<td>144</td>
<td>20</td>
</tr>
</tbody>
</table>

The highest bitrate chosen is 144 Kbps as in a rural area usually the bandwidth is lower than the UMTS standard bandwidth of 2 Mbps. In the first phase of the simulations, a UMTS network in the context of a rural area is simulated with various numbers of voice application users starting from 8 to 20. The values of packet loss, delay, and jitter are evaluated in each scenario. The scenarios are run for 13 times with different seed values and 95% confidence interval. In the second phase of the simulations, a VS application is added to the network. Two separate video codecs, H.263 and MPEG-4, have been used for the streaming application. The reasons for using two separate video codecs are twofold. Firstly, it will help to evaluate the performance of voice calls in combination with different codec based streaming applications. Secondly, the usage of different video codec will help to study the most suitable settings under such network scenarios. The VS server is placed in the WiMAX environment. The performance of the voice application is evaluated with each type of the streaming codec running. This assessment is conducted to identify.
the optimal number of voice and streaming users for this sort of network settings. In the third stage of simulations, a VC session is added to the voice and VS sessions.

IP-based calls are used for voice communications. 7.4 kbit/s mode Adaptive Multi-Rate (AMR) speech codec is used for these calls with an activity factor of 0.5. A typical AMR packet runs for 20 msec with a payload size of 224 bits. VS applications are the traditional mediums for distance education. The video codec used, in this case, are the H.263 and MPEG-4 as these two are the most popular video codec for mobile devices [163]. Two video trace files encoded with H.263 and MPEG-4 are used for simulations. The trace files are collected from [164]. These trace files have come with the parameters presented in Table 4.4. The parameters such as the resolution, frame size, and number of frames have been set up in the simulation configuration file accordingly. These Group of Picture (GoP) structure is denoted using GgBb where g defines the total number of frames in a GoP and b indicates the number of B frames between successive I or P frames. B frames are the bi-directionally predictive coded frames, I frames are intra-coded frames and P are predictive coded frames. The VC session uses a frame rate of 15 fps. The packet size

<table>
<thead>
<tr>
<th>Codec</th>
<th>Performance metrics</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Metric</td>
</tr>
<tr>
<td>H.263</td>
<td>QP</td>
</tr>
<tr>
<td></td>
<td>Resolution</td>
</tr>
<tr>
<td></td>
<td>Encoder</td>
</tr>
<tr>
<td></td>
<td>Bit rate</td>
</tr>
<tr>
<td></td>
<td>USE of PB-frames</td>
</tr>
<tr>
<td></td>
<td>Length</td>
</tr>
<tr>
<td></td>
<td>No. Frames</td>
</tr>
<tr>
<td>MPEG-4</td>
<td>Layer</td>
</tr>
<tr>
<td></td>
<td>Encoder</td>
</tr>
<tr>
<td></td>
<td>Frame Size</td>
</tr>
<tr>
<td></td>
<td>No. Frames</td>
</tr>
<tr>
<td></td>
<td>Number of GoP</td>
</tr>
<tr>
<td></td>
<td>INTRA PERIOD</td>
</tr>
<tr>
<td></td>
<td>QP</td>
</tr>
</tbody>
</table>
used for the VC application is 576 bytes. The FDD version of UMTS is considered in these simulation scenarios. A free space path loss model is used, which is the most suitable model for a rural environment. The throughput-based admission control algorithm is used and in the downlink, the other-cell interference factor is set to 0.65. Figure 4.7 shows the simulation setup scenario.

![Figure 4.7 UMTS-based Network Scenario](image)

### 4.3.2 LTE-UMTS-based Heterogeneous Network Scenarios

In these scenarios, LTE-UMTS and LTE only network are considered with Voice, VC, and VS applications. In this case, also the LTE model of OPNET is used for the simulations. The environments chosen for these simulations are also rural outdoor environment. Some of the simulation parameters are stated in Table 4.5.

**Table 4.5 Simulation Parameters for LTE**

<table>
<thead>
<tr>
<th>Layers</th>
<th>Parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Application</strong></td>
<td></td>
</tr>
<tr>
<td>Voice calls</td>
<td>Interactive Voice 224 bits</td>
</tr>
<tr>
<td>VS</td>
<td>Best Effort Variant</td>
</tr>
<tr>
<td>VC</td>
<td>Interactive Voice 576 bytes</td>
</tr>
<tr>
<td><strong>PHY -UL</strong></td>
<td></td>
</tr>
<tr>
<td>Modulation and Coding Scheme</td>
<td>9</td>
</tr>
<tr>
<td>Timer MRW (msec)</td>
<td></td>
</tr>
<tr>
<td>Base Frequency</td>
<td>1920MHz</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>20 MHz</td>
</tr>
</tbody>
</table>
### Chapter 4: Fixed Weight-based QoS Analysis

<table>
<thead>
<tr>
<th>PHY-DL</th>
<th>Base Frequency</th>
<th>2110 MHz</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Bandwidth</td>
<td>20 MHz</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>EPS Bearer</th>
<th>QoS Class Identifier</th>
<th>GOLD</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Uplink Guaranteed Bit Rate (bps)</td>
<td>64 Kbps</td>
</tr>
<tr>
<td></td>
<td>Downlink Guaranteed Bit Rate (BPS)</td>
<td>64 Kbps</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>UL</th>
<th>Maximum Bit Rate</th>
<th>64 Kbps</th>
</tr>
</thead>
<tbody>
<tr>
<td>DL</td>
<td>Maximum Bit Rate</td>
<td>64 Kbps</td>
</tr>
<tr>
<td>Allocation Retention Priority</td>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>QoS Class Identifier</th>
<th>Silver</th>
</tr>
</thead>
<tbody>
<tr>
<td>UL Guaranteed Bit Rate (bps)</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>DL Guaranteed Bit Rate (bps)</td>
<td>384 Kbps</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>UL</th>
<th>Maximum Bit Rate</th>
<th>384 Kbps</th>
</tr>
</thead>
<tbody>
<tr>
<td>DL</td>
<td>Maximum Bit Rate</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>Allocation Retention Priority</td>
<td>4</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>QoS Class Identifier</th>
<th>Bronze</th>
</tr>
</thead>
<tbody>
<tr>
<td>UL Guaranteed Bit Rate (bps)</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>DL Guaranteed Bit Rate (bps)</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>UL Maximum Bit Rate (bps)</td>
<td>384 Kbps</td>
</tr>
</tbody>
</table>
Chapter 4: Fixed Weight-based QoS Analysis

In the first phase of the simulations, LTE-based network in the context of a rural area is simulated with various numbers of voice application users starting from 8 to 20. The values of packet loss, delay, and jitter are evaluated in each scenario. The scenarios have been run for 13 times with different seed values and 95% confidence interval. The application settings in the LTE network are same as the UMTS network.

In the second phase of the simulations, LTE-UMTS-based networks in the context of a rural-urban area is simulated with various numbers of voice calls, VS, and VC sessions. In some of the scenarios, the VS server is placed in the LTE RAN and in some of them, the voice calls are initiated between the UMTS and LTE network.

4.3.3 Other Heterogeneous Network-based Scenarios
These scenarios involve UMTS, WiMAX, WLAN, LTE and Ethernet technologies. The scenarios are designed in the context of rural-urban communication. The core architecture of this scenario is divided into two main segments; these are the Rural

![Figure 4.8 Heterogeneous Network Scenario](image-url)
Area Segment (RAS) and the Urban Area Segment (UAS). Figure 4.8 depicts the architecture of these scenarios.

The conference participants of rural area are considered to be participating through 3G (UMTS), WLAN, or Ethernet. The Session Initiation Protocol (SIP)-based proxy server sits between the urban area and the backbone network. All video and audio data from both parties are transmitted via this server. The overall architecture of the rural area segment can be divided into three technology-based clusters. These are the UMTS/3G cluster, the WLAN cluster, and the Ethernet cluster. The Urban area segment may also be categorised into two of such clusters, namely the WiMAX and the Ethernet clusters. These urban and rural clusters are connected to a backbone network. The conference participants in the urban area are considered to be participating using the WiMAX or Ethernet network.

Each cluster in the RAS has gateway that connects it to the backbone network, which in turn connects it to the proxy server of the UAS. The connection between the wireless access points and their wireless gateway nodes are considered to be provided by 45 Mbps Digital Signal 3 (DS3) links. The same links connect the wireless gateway nodes and the backbone network. A UMTS gateway node connects the UMTS GPRS support node (GGSN) to the backbone network through a DS3 link. The gateway nodes connect UAS clusters to RAS through a backbone network.

Several architectures are suitable for VC applications. Peer-to-Peer (P2P) architecture can be used in a two-party VC session. In such an architecture, both participants can send data to each other directly. For a multi-party VC session, both P2P and server/client (S/C)-based architectures are suitable. In a multipoint P2P VC session, users relay videos to each other. On the other hand, in an S/C-based architecture, at first, the participant uploads a video to a server and then the server sends the video to the receiver. The VC applications over this scenario are modelled over S/C-based architecture. Figure 4.9 depicts this architecture.
A number of VC applications are available depending on different architectures, bandwidth, and data size. Bandwidth threshold and suitable video/audio packet size for several VC applications are analysed in [165]. Vsee is selected for this model, as, in terms of bandwidth and available user data rate, this application appears to be the most suitable one. G.723.1.5.3k, which is the recommended codec for VC, is used for audio transmission. This codec has a frame size of 30 msec and coding rate of 5.3 Kbps, with a payload (PL) size of 159 bytes. Table 4.6 summarises the VC specifications. In some scenarios, a composition of LTE, UMTS and Ethernet networks are considered. The VC and VS servers are placed in the Ethernet technology cluster. The VC and VS clients are placed in the UMTS and LTE technology. A number of active voice calls are simulated in these two technology clusters.

**TABLE 4.6 VC SPECIFICATIONS**

<table>
<thead>
<tr>
<th>Attributes</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video frame rate per second</td>
<td>30</td>
</tr>
<tr>
<td>Video frame size (byte)</td>
<td>600</td>
</tr>
<tr>
<td>Video type of service</td>
<td>Differentiated service for Interactive multimedia (EF)</td>
</tr>
<tr>
<td>Audio codec</td>
<td>G.723.1.5.3k</td>
</tr>
</tbody>
</table>
4.4 Simulation Result Analysis

This section presents the result analysis from the above-discussed simulation scenarios.

4.4.1 UMTS/UMTS-WiMAX-based Networks

Figure 4.10 shows the end-to-end delay experienced by voice users after adding an MPEG-4 or H.263 codec-based VS session on the network. The VS server is in the WiMAX RAN, and the VS client is placed in the UMTS RAN. It shows that an H.263 codec-based VS client has less effect on the performance of voice clients compared to an MPEG-4 VS client. Figure 4.11 demonstrates the percentage of packet loss experienced by voice calls after adding an MPEG-4 or H.263-based VS client. The simulation results clearly indicate that the voice calls experience less packet loss in the presence of an H.263-based VS client. The figure also shows that H.263 VS client with 10 simultaneous voice calls experience a higher packet loss.
than other situations. This is because of poor coverage experienced by some of the users in that particular scenario. As some of the callers moved far away from the base station, the statistics showed more packet loss compared to other scenarios.

Table 4.7 presents the performance of a VS client while using H.263 and MPEG-4 codec. In terms of packet loss, H.263 codec shows a better performance under the UMTS RAN than MPEG-4 codec does. In the case of delay, the results indicate the same behaviour. The simulation results also demonstrate that the network experiences better performance with one H.263 codec-based VS client and twelve simultaneous voice calls in terms of packet loss.

![Number of calls vs packet loss](image)

**Figure 4.11** Packet loss for Voice Clients in the Presence of Streaming Client
However, in terms of delay, the platform shows an acceptable performance with one H.263-based VS client and twenty simultaneous voice calls. On the other hand, the network shows a different behaviour in the presence of the MPEG-4-based VS client. With the MPEG-4-based client, the optimal number is ten simultaneous voice calls and one VS client. Table 4.8 shows the percentage of packet loss in the presence of eight voice users, one VS client, and one VC user. The results indicate that the simultaneous presence of the VC and the VS client affects the performance of voice calls to a noticeable extent.

### Table 4.7 Packet loss for Streaming Client in the Presence of Different Number of Voice Clients

<table>
<thead>
<tr>
<th>Number of Voice calls</th>
<th>Number of streaming client</th>
<th>Packet loss for different codec</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>MPEG-4</td>
</tr>
<tr>
<td>8</td>
<td>1</td>
<td>5.58</td>
</tr>
<tr>
<td>10</td>
<td>1</td>
<td>5.56</td>
</tr>
<tr>
<td>12</td>
<td>1</td>
<td>5.75</td>
</tr>
<tr>
<td>14</td>
<td>1</td>
<td>5.93</td>
</tr>
<tr>
<td>16</td>
<td>1</td>
<td>5.79</td>
</tr>
<tr>
<td>18</td>
<td>1</td>
<td>5.62</td>
</tr>
<tr>
<td>20</td>
<td>1</td>
<td>5.75</td>
</tr>
</tbody>
</table>

### Table 4.8 Packet loss for Different Applications

<table>
<thead>
<tr>
<th>Type of application</th>
<th>Packet loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice application</td>
<td>13.55</td>
</tr>
<tr>
<td>Streaming application</td>
<td>1.0287</td>
</tr>
<tr>
<td>Video conferencing application</td>
<td>4.02</td>
</tr>
</tbody>
</table>

#### 4.4.2 LTE/LTE-UMTS-based Networks

Table 4.9 shows the data for the LTE network with different number of voice calls. The voice calls demonstrate a better performance compared to the UMTS network.
### Table 4.9 Data Analysis for LTE Network

<table>
<thead>
<tr>
<th>Number of voice calls</th>
<th>End-to-end delay (msec)</th>
<th>Packet loss (%)</th>
<th>Jitter (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>0.092</td>
<td>0.02</td>
<td>20</td>
</tr>
<tr>
<td>10</td>
<td>0.092</td>
<td>0.06</td>
<td>24</td>
</tr>
<tr>
<td>12</td>
<td>0.093</td>
<td>0.08</td>
<td>28</td>
</tr>
<tr>
<td>14</td>
<td>0.094</td>
<td>0.12</td>
<td>30</td>
</tr>
<tr>
<td>16</td>
<td>0.095</td>
<td>0.15</td>
<td>32</td>
</tr>
<tr>
<td>18</td>
<td>0.099</td>
<td>0.18</td>
<td>34</td>
</tr>
</tbody>
</table>

Figure 4.12 shows the comparison of packet loss for VS client when the LTE and the WiMAX-based servers are used for the VS session. The VS client is in the UMTS network. Figure 4.13 shows the comparison of end-to-end delay for the same. If the server side uses the LTE technology, the VS client experiences less packet loss and delay compared to the WiMAX-based network.

![Figure 4.12 Comparison of Packet loss for different Technologies](image)

**Figure 4.12** Comparison of Packet loss for different Technologies
Chapter 4: Fixed Weight-based QoS Analysis

4.4.3 Other Heterogeneous Networks

In the first phase of these simulations, two participants join the VC session from each cluster of RAS. For instance, there are two participants in the UMTS cluster, one of them is in a conference with an urban participant located in the Ethernet cluster, and another participant of the same cluster is in a conference with an urban participant from the WiMAX cluster. In the same manner, participants from the Ethernet and the WLAN cluster join the conference with an urban participant.

Several other phases of simulations are conducted with a different number of simultaneous participants in the VC session. In the second stage of simulations, the number of participants is increased to twenty in each RAS cluster. For example, in the UMTS cluster, altogether, there are twenty participants. Ten of them are in a conference session with an urban participant from the Ethernet cluster and ten other participants, are in a conference session with an urban participant from the WiMAX cluster. Both WLAN and Ethernet clusters support the same number of users and the same type of conferences. To elaborate, the urban participant located in the WiMAX cluster in the UAS initiates a VC session with ten participants from the Ethernet cluster, ten participants from the WLAN cluster, and ten participants from the UMTS cluster of the RAS.

In the third phase, the number of participants is reduced to ten resulting in an equal number of participants in each RAS cluster for each urban user. In the fourth and the fifth phase, the number of participants is reduced to eight and six respectively. Each VC session follows a specific naming convention, which is the name of the RAS cluster of the participant followed by the name of the UAS cluster of the participant.
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For instance, the conference between a participant from the UMTS cluster of the RAS and an urban participant from the WiMAX cluster is termed as UMTS-WiMAX pair/conference/transmission. Each scenario is run 13 times with different seed values and 95% confidence interval.

In the case of voice conferencing, the same pattern is followed for simulations. The simulation for voice conferencing is conducted in three phases involving three, five, and ten simultaneous participants. Performances for both video and voice transmissions of all VC sessions are analysed for each phase of simulations in terms of end-to-end delay, and packet loss. End-to-end delay is calculated based on the network delay, the encoding delay, the decoding delay, the compression delay, and the decompression delay. The acceptable performance values for these parameters are derived from Table 4.2.

### TABLE 4.10 SIMULATION RESULTS FOR RAS CLUSTERS – UAS CLUSTER (ETHERNET) CONFERENCE

<table>
<thead>
<tr>
<th>Conference types</th>
<th>Metrics</th>
<th>Resulting values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ethernet-Ethernet</td>
<td>End-to-end delay</td>
<td>1.3 msec</td>
</tr>
<tr>
<td></td>
<td>Packet loss</td>
<td>0.1 to 0.2%</td>
</tr>
<tr>
<td>WLAN-Ethernet</td>
<td>End-to-end delay</td>
<td>5.7 to 5.8 msec</td>
</tr>
<tr>
<td></td>
<td>Packet loss</td>
<td>0%</td>
</tr>
<tr>
<td>UMTS-Ethernet</td>
<td>End-to-end delay</td>
<td>110 to 120 msec</td>
</tr>
<tr>
<td></td>
<td>Packet loss</td>
<td>0%</td>
</tr>
</tbody>
</table>

In the first phase of simulations, with three participants in each conference cluster the sessions achieve an acceptable performance level. Table 4.10 presents these values. The conference in the WLAN-Ethernet shows better performance than other clusters in terms of packet loss. In the case of the conference between the RAS clusters and the WiMAX cluster in the UAS, the WLAN-WiMAX and the Ethernet-WiMAX conference experiences insignificant packet loss. Other performance parameters, such as end-to-end delay shows values within the acceptable level. However, the participants in the UMTS-WiMAX conference experience a higher packet loss than the former two. Similarly, they also exhibit a higher average end-to-end delay, which
is 170 msec.

Figure 4.14 Packet loss for UMTS Clients

Figure 4.15 Packet loss for WLAN Clients

In the next phase, the number of participants in each conference cluster is increased to ten. Therefore, each urban participant is in a conference with thirty other simultaneous participants using different technologies. In the case of the UMTS-Ethernet conference, only one out of ten participants is able to join the conference. Although, all participants under the WLAN and the Ethernet cluster are able to join the conference and receive video data, the quality of received transmission varies. In the case of conference with the urban user in the WiMAX cluster, seven out of ten participants in the Ethernet cluster are able to receive video transmission data. In the case of participants in the WLAN cluster, only six out of ten are able to receive data successfully and two out of ten transmissions are successful in the case of users in
the UMTS cluster. However, the quality of received data for all users degrades drastically.

To explore the capacity of the network further, in the third phase, the participant numbers are reduced to five in each conference cluster. Therefore, each urban user from the WiMAX and the Ethernet cluster are in conferences with fifteen simultaneous users respectively. This time the successful connection-ratio and packet loss improves significantly. Packet losses for all the Ethernet-Ethernet video transmissions is reduced to an average of 4.9%. The clients in the WLAN-WiMAX cluster experience an average of 1.65% packet loss. However, the Ethernet-WiMAX and the UMTS-WiMAX cluster still experience significant packet losses.

![Figure 4.16 Successful Received Packets for WLAN Clients in Voice Transmissions](image)

In the next phase, the simulation has been carried out with three simultaneous users in each conference cluster resulting in nine simultaneous RAS participants with one urban participant. As expected, the video transmission exhibits better results in terms of both capacity and quality. Participants in the UMTS-WiMAX and the Ethernet-WiMAX conferences also exhibit better performance. Figure 4.14 and 4.15 show packet loss for the video transmissions in the third phase of simulations. The figures clearly indicate that the WLAN-WiMAX and the UMTS-WiMAX conference show better performance than the WLAN-Ethernet and the UMTS-Ethernet conference. It
is noticeable that the combinations of wireless-based technologies outperform the wireless-Ethernet-based combinations. This is due to the suitable matching of QoS service classes between WLAN-WiMAX and UMTS-WiMAX technologies.

The voice traffics in the VC sessions also demonstrate interesting behaviour. In the first phase, the simultaneous users in the WLAN-Ethernet cluster experience 80 msec end-to-end delay and the participants in the UMTS-Ethernet conferences undergo 130 msec delay. There is no packet loss for all three types of transmissions. In the second phase, likewise video transmissions, not all participants are able to receive voice data successfully. In the case of the UMTS-Ethernet pair communication, only two out of ten users are able to receive data. All participants in the Ethernet-Ethernet and the WLAN-Ethernet conferences are able to receive data.

After the number of participants have been reduced, the existent participants experience a range of different performance in regards to packet loss and end-to-end delay. The UMTS-Ethernet conference experiences more packet loss in comparison with the former two types of conferences. The conferences with the urban user in the Ethernet cluster show a higher degree of packet loss than the conferences with the user in the WiMAX cluster. The values of end-to-end delay do not vary much in the presence of large and few numbers of simultaneous participants. However, in terms

![Figure 4.17 Successful Received Packets for UMTS Clients in Voice Transmissions](image)

Figure 4.17 Successful Received Packets for UMTS Clients in Voice Transmissions
of packet loss, voice transmissions from various conference types show different behaviours.

For voice transmissions, the conferences between the WLAN and the WiMAX cluster experience no packet loss regardless of the number of users. Figure 4.16 shows the number of successfully received packets for participants from the WLAN cluster. The figure clearly indicates that in case of voice transmissions, the WLAN-WiMAX conferences show a less packet loss. Figure 4.17 shows a comparison between the number of successful packets received for the UMTS-Ethernet and UMTS-WiMAX conferences. The UMTS-WiMAX conference shows less packet loss compared to the UMTS-Ethernet conference. This is due to the proper matching of QoS service classes between UMTS and WiMAX technology.

To summarise, the above conference scenarios show an acceptable performance level when there are ten simultaneous participants in each VC session. In terms of both video and voice transmission quality, the WLAN-WiMAX conference demonstrates better performance compared to other two conference clusters. This is due to the more suitable matching of QoS service classes between WLAN and WiMAX technologies. Participants in the UMTS cluster experience better performance in case of voice transmissions, compared to video transmissions. This is due to the underlying configuration of UMTS technology as it is designed mostly for voice transmission.

In the next phase, only voice conferencing-based simulations have been conducted. At first, the performance has been evaluated with one participant in each cluster. Then the performance has been assessed with ten and five simultaneous participants respectively. In contrast to VC, voice conference does not show much performance variation with the increasing number of participants.

In the first phase of the simulations, all the conferences in each cluster show an acceptable performance level. In the second phase of the simulations, with ten simultaneous participants, the UMTS-Ethernet outperforms the other clusters. In the third phase, with five simultaneous participants, except the Ethernet-WiMAX, the UMTS-WiMAX and the UMTS-Ethernet cluster, all other clusters achieve the highest performance. To summarise, the voice conferencing exhibits better performance in this heterogeneous architecture than video conferencing. With a
proper QoS matching, transmissions to the WiMAX server outperform transmissions to the Ethernet server. Table 4.11 shows the performance evaluation results of different clusters.

**4.5 QoS Analysis**

The simulation result analysis in the previous section demonstrates that the assessment of network performance by investigating the value of each QoS-related parameter individually is a challenging task. Different communication technologies and varying numbers of users in the picture makes it even more complicated. The analysis of various scenarios shows that in some cases of VC session, although the value of end-to-end delay is acceptable, the amount of packet loss is below the acceptable level. On the other hand, VC and voice conferencing show inconsistent performance under different communication technologies.

**TABLE 4.11 VOICE CONFERENCING DATA ANALYSIS**

<table>
<thead>
<tr>
<th>Connections</th>
<th>Number of Users</th>
<th>Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Delay (msec)</td>
</tr>
<tr>
<td>Ethernet-Ethernet</td>
<td>10</td>
<td>80</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>80</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>80</td>
</tr>
<tr>
<td>WLAN-Ethernet</td>
<td>10</td>
<td>80</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>79</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>78</td>
</tr>
<tr>
<td>WLAN-WiMAX</td>
<td>10</td>
<td>82.5</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>82</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>81.5</td>
</tr>
<tr>
<td>UMTS-WiMAX</td>
<td>10</td>
<td>130</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>130</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>130</td>
</tr>
</tbody>
</table>
The proposed fixed weight-based method can handle these complexities of QoS analysis by using QoS metric evaluation functions. The ranges of these QoS metrics are first interpreted. As stated in section 4.2, there are three QoS metrics to evaluate the network performance. The first one is the application QoS metric, the second one is the RAN QoS metric, and the last one is the network configuration QoS metric. A scale that ranges between 0 and 1 is used for each QoS metric; 1 being the highest and zero being the lowest level. Table 4.12 shows these scales.

**Table 4.12 Interpretation of QoS Metric Values**

<table>
<thead>
<tr>
<th>QoS Metric</th>
<th>Value Range</th>
<th>Interpretation</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Good</td>
<td>Average</td>
</tr>
<tr>
<td>Application QoS Metric</td>
<td>0-1</td>
<td>0.8-1</td>
</tr>
<tr>
<td>Access Network QoS Metric</td>
<td>0-1</td>
<td>0.8-1</td>
</tr>
<tr>
<td>Network Configuration QoS Metric</td>
<td>0-1</td>
<td>0.8-1</td>
</tr>
</tbody>
</table>

### 4.5.1 UMTS/UMTS-WiMAX-based Scenarios

In this section, the outcomes of the QoS analysis of the UMTS and UMTS-WiMAX technology-based scenarios are presented.

#### 4.5.1.1 Single Application and Single RAN-based Scenarios

Table 4.13 outlines the QoS analysis results of the UMTS-based scenarios in the presence of different number of voice conferencing participants. Some of the results presented in Table 4.13 are illustrated in the following calculations in detail. In all these calculations, the following denotations are used:

- **UMTS – U**
- **Voice application – V**
- **End-to-end delay – D**
- **Jitter – J**
- **Packet loss – PL**
The functions described in section 4.2 are applied in this section for the QoS analysis.

For eight simultaneous calls, the participants experience an average end-to-end delay of 202 msec, an average packet loss of 2.06% and an average jitter of 20 msec. The performance metric of these parameters for this scenario using equation (4.1) described in section 4.2.2 is calculated as:

\[
P_{QoS_{DV}} = \frac{400 - 202}{400 - 150} = 0.792
\]

\[
P_{QoS_{CV}} = 1
\]

\[
P_{QoS_{PLV}} = 1
\]

The application QoS metric using equation (4.2) described in section 4.2.3 is calculated as:

\[
QoSAM = 0.792 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9168.
\]

In this case, the RAN QoS metric and the network configuration metric have the same values as the application QoS metric. This is because the network has single application and single RAN.

For ten simultaneous calls, the participants experience an average end-to-end delay of 206 msec, an average packet loss of 2.90%, and an average jitter of 22 msec. The performance metric of these parameters for this scenario using equation (4.1) is calculated as:

\[
P_{QoS_{DV}} = \frac{400 - 206}{400 - 150} = 0.776
\]

\[
QoSAM = 0.776 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9104
\]

The application QoS metric using equation (4.2) described in section 4.2.3 is calculated as:

\[
QoSAM = 0.792 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9168.
\]

In this case, the RAN QoS metric and the network configuration metric have the same values as the application QoS metric. This is because the network has single application and single RAN.

For ten simultaneous calls, the participants experience an average end-to-end delay of 206 msec, an average packet loss of 2.90%, and an average jitter of 22 msec. The performance metric of these parameters for this scenario using equation (4.1) is calculated as:

\[
P_{QoS_{DV}} = \frac{400 - 206}{400 - 150} = 0.776
\]

\[
QoSAM = 0.776 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9104
\]

The application QoS metric using equation (4.2) described in section 4.2.3 is calculated as:

\[
QoSAM = 0.792 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9168.
\]

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\[
P_{QoS_{DV}} = \frac{400 - 206}{400 - 150} = 0.776
\]

\[
QoSAM = 0.776 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9104
\]

The application QoS metric using equation (4.2) described in section 4.2.3 is calculated as:

\[
QoSAM = 0.792 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9168.
\]

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\[
P_{QoS_{DV}} = \frac{400 - 206}{400 - 150} = 0.776
\]

\[
QoSAM = 0.776 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9104
\]

The application QoS metric using equation (4.2) described in section 4.2.3 is calculated as:

\[
QoSAM = 0.792 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9168.
\]

In this case, the RAN QoS metric and the network configuration metric have the same values as the application QoS metric. This is because the network has single application and single RAN.
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\[ P_{QP_{D,V}} = 1 \]

\[ P_{QP_{PL,V}} = 1 \]

The application QoS metric using equation (4.2) is calculated as:

\[ QoSAM^{U}_{D,V} = 0.776 \times 0.4 + 1 \times 0.4 + 1 \times 0.2 = 0.9104 \]

The results of other simulation scenarios have been calculated in a similar way.

4.5.1.2 Multiple Applications and RAN-based Scenarios

In this stage, the QoS is analysed for the next set of UMTS-WiMAX-based scenarios. As these scenarios have different radio access networks in the both ends of the connection, in order to distinguish between them, the equation (4.1) is expressed in the following form:

\[
P_{QP^{R,RS}_{k,A}} = \begin{cases} 
0, & RTP_{QP^{R,RS}_{k,A}} \geq UB_{QP_{k,A}} \\
1, & RTP_{QP^{R,RS}_{k,A}} \leq LB_{QP_{k,A}} \\
\frac{UB_{QP_{k,A}} - RTP_{QP^{R,RS}_{k,A}}}{UB_{QP_{k,A}} - LB_{QP_{k,A}}}, & UB_{QP_{k,A}} > RTP_{QP^{R,RS}_{k,A}} > LB_{QP_{k,A}} 
\end{cases} \quad (4.5)
\]

where RS is the technology of the other side of the connection.

The equation (4.2) is expressed in the following form for one end of the connection:

\[ QoSAM^{R,RS}_{A} = \sum_{k=1}^{p} P_{QP^{R,RS}_{k,A}} W_{QP_{k,A}} \quad (4.6) \]

The function for the RAN QoS that is the equation (4.3) is expressed as:

\[ QoSAM^{N}_{R,RS} = \sum_{i=1}^{m} QoSAM^{R,RS}_{A} W^{R}_{A} \quad (4.7) \]

The function to calculate the network configuration metric at the one end of the connection that is equation (4.4) is expressed as:

\[ QoSAM^{N}_{N} = \sum_{j=1}^{n} QoSAM^{N}_{R,j,RS} W^{N}_{R,j} \quad (4.8) \]
The equation (4.8) is used in the cases when there is one access technology on the other side of the connection. If there is a combination of multiple access technologies in the both sides of the connection, then the equation (4.8) is expressed as:

$$QoSCM_{N} = \sum_{i=1}^{g} \sum_{j=1}^{q} QoSRM_{R_{i},RS_{j}}^{N} W_{R_{j}}^{N}$$ (4.9)

To calculate the performance metric, application QoS metric, RAN QoS metric, and network configuration QoS metric for the other side of the connection the above equations can be expressed as:

$$P_{QoP_{k,A}}^{RS,R} = \begin{cases} 0 & , \frac{RTP_{QoS,i,a}}{RTP_{QoS,i,a}} \geq UB_{QoP_{k,A}} \\ 1 & , \frac{RTP_{QoS,i,a}}{RTP_{QoS,i,a}} \leq UB_{QoP_{k,A}} \\ UB_{QoP_{k,A}} - RTP_{QoS,i,a} & , \frac{RTP_{QoS,i,a}}{RTP_{QoS,i,a}} > UB_{QoP_{k,A}} \end{cases}$$ (4.10)

$$QoSAM_{A}^{RS,R} = \sum_{k=1}^{p} P_{QoP_{k,A}}^{RS,R} W_{QoP_{k,A}}$$ (4.11)

$$QoSRM_{RS,R}^{N} = \sum_{i=1}^{m} QoSAM_{A}^{RS,R} W_{RS}^{RS}$$ (4.12)

$$QoSCM_{N} = \sum_{j=1}^{n} \sum_{l=1}^{l} QoSRM_{R_{j},RS_{l}}^{N} W_{R_{l}}^{N}$$ (4.13)

If there are multiple access networks on the both sides of the connection, then the equation (4.13) is expressed as:

$$QoSCM_{N} = \sum_{j=1}^{n} \sum_{l=1}^{l} QoSRM_{R_{j},RS_{l}}^{N} W_{R_{l}}^{N}$$ (4.14)

For eight simultaneous voice calls and one VS session, in the UMTS network the voice calls experience an average end-to-end delay of 216 msec, an average packet loss of 2.92%, and an average jitter of 40 msec. In all the following calculations, the UMTS technology is expressed as U and the WiMAX technology is expressed as W. The performance metric of these parameters in voice calls for this scenario using equation (4.1) is calculated as:
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The application QoS metric using equation (4.2) is calculated as:

\[ P_{QoU}^{U,V} = \frac{400-216}{400-150} = 0.74 \]

\[ P_{QoV}^{U,V} = \frac{75-40}{75-30} = 0.77 \]

\[ P_{QoL}^{U,V} = 1 \]

The application QoS metric using equation (4.2) is calculated as:
### Table 4.14 QoS-related Parameter Values for Mixed Traffic

<table>
<thead>
<tr>
<th>Number of active calls/sessions</th>
<th>End-to-end delay (msec)</th>
<th>Packet loss (%)</th>
<th>Jitter (msec)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>V</td>
<td>VS</td>
<td>V</td>
</tr>
<tr>
<td>8</td>
<td>1</td>
<td>216</td>
<td>176</td>
</tr>
<tr>
<td>10</td>
<td>1</td>
<td>221</td>
<td>179</td>
</tr>
<tr>
<td>12</td>
<td>1</td>
<td>224</td>
<td>180</td>
</tr>
<tr>
<td>14</td>
<td>1</td>
<td>228</td>
<td>180</td>
</tr>
<tr>
<td>16</td>
<td>1</td>
<td>230</td>
<td>180</td>
</tr>
<tr>
<td>18</td>
<td>1</td>
<td>233</td>
<td>180</td>
</tr>
</tbody>
</table>

### Table 4.15 QoS Metric Values for Mixed Traffic
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\[ QoSAM^U_V = 0.74 \times 0.4 + 0.77 \times 0.4 + 1 \times 0.2 = 0.804 \]

The performance metrics for VS application are calculated using the equation (4.5).

\[ P_{QoSA_MVS}^U = 1 \]

\[ P_{QoSA_MVS}^V = 1 \]

The application QoS metric using equation (4.6) is calculated as:

\[ QoSAM^U_{VS} = 1 \times 0.5 + 1 \times 0.5 = 1 \]

The RAN QoS metric using equation (4.7) is calculated as:

\[ QoS_R^U = QoSAM^U_V \times W^U_V + QoSAM^U_{VS} \times W^U_{VS} = 0.8824 \]

The weight for the applications is set as 0.5. Table 4.14 and 4.15 show the QoS-related parameter values and the overall QoS measurement of this network.

For ten simultaneous voice calls and one VS session, the voice calls experience an average end-to-end delay of 221 msec, an average packet loss of 2.95%, and an average jitter of 58 msec.

The performance metric of these parameters in voice calls for this scenario using equation (4.1) is calculated as:

\[ P_{QoSA_DV}^U = \frac{400-221}{400-150} = 0.72 \]

\[ P_{QoSA_JV}^U = \frac{75-58}{75-30} = 0.38 \]

\[ P_{QoSA_PLV}^U = 1 \]

<table>
<thead>
<tr>
<th>Numbers of active calls/sessions</th>
<th>QoSAM(<em>A)</em>(^R)</th>
<th>QoSRM(<em>N)</em>(^R)</th>
<th>Overall QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td>V  VS</td>
<td>V  VS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>8  1</td>
<td>0.804</td>
<td>1</td>
<td>0.8824</td>
</tr>
<tr>
<td>10 1</td>
<td>0.64</td>
<td>1</td>
<td>0.784</td>
</tr>
<tr>
<td>12 1</td>
<td>0.57</td>
<td>1</td>
<td>0.74</td>
</tr>
<tr>
<td>14 1</td>
<td>0.52</td>
<td>1</td>
<td>0.71</td>
</tr>
<tr>
<td>16 1</td>
<td>0.488</td>
<td>1</td>
<td>0.69</td>
</tr>
<tr>
<td>18 1</td>
<td>0.43</td>
<td>1</td>
<td>0.66</td>
</tr>
</tbody>
</table>
Chapter 4: Fixed Weight-based QoS Analysis

The application QoS metric using equation (4.2) is calculated as:

\[ QoSAM_{V}^{U} = 0.72 \times 0.4 + 0.38 \times 0.4 + 1 \times 0.2 = 0.64 \]

The performance metric for VS application is calculated as:

\[ P_{QoS}^{U,W} = 1 \]
\[ P_{QoS}^{U,W} = 1 \]

The application QoS metric using equation (4.6) is calculated as:

\[ QoSAM_{VS}^{UW} = 1 \times 0.4 + 1 \times 0.6 = 1 \]

The RAN QoS metric is calculated using equation (4.7) is calculated as:

\[ QoSRM_{U,W}^{N} = QoSAM_{V}^{U} \times W_{V}^{U} + QoSAM_{VS}^{UW} \times W_{VS}^{U} = 0.784 \]

The performance metric for VS application is calculated as:

\[ P_{QoS}^{U,W} = 1 \]
\[ P_{QoS}^{U,W} = 1 \]

The application QoS metric using equation (4.6) is calculated as:

\[ QoSAM_{VS}^{UW} = 1 \times 0.4 + 1 \times 0.6 = 1 \]

The RAN QoS metric is calculated using equation (4.7) is calculated as:

\[ QoSRM_{U,W}^{N} = QoSAM_{V}^{U} \times W_{V}^{U} + QoSAM_{VS}^{UW} \times W_{VS}^{U} = 0.71 \]

The other results have been also calculated using the same equations.
Figure 4.18 shows the effects of VS application on the performance of voice application. Due to the presence of VS application in the network, the performance of voice application drops by 11.6% when there are eight simultaneous voice calls and one VS session on the network. When there are ten simultaneous voice calls, this performance decreases by 27.04%, and when there are twelve voice calls, it drops by 34%. With 18 simultaneous voice calls, the voice calls experience poor performance even without the presence of VS client. The Voice application QoS metric is rated as poor in this context.

Figure 4.19 shows the effects of different VS codec on the performance of the network. If the VS Session uses MPEG-4 codec, when there are nine active users in the network, the performance degrades by 35.24% compared to applying H.263 codec. When there are eleven active users, this performance degrades by 29%. When there are thirteen active users, this performance degrades by 26.8%. With the numbers of increased active users in the network, the performance difference between H.263 and MPEG-4 network becomes less. Because, in both cases, the number of increased active users affects the performance of the network.
4.5.2 LTE/LTE-UMTS-based Scenarios

Table 4.16 shows the QoS Metrics for the LTE network with different number of voice calls. Table 4.17 presents the comparison of application QoS Metrics for the UMTS and LTE access networks. The LTE access network demonstrates a better performance for voice calls than the UMTS access network. Figure 4.20 shows the comparisons of RAN QoS Metric for the UMTS and LTE access networks. For 20 voice calls, the LTE network experiences 30% better performance than the UMTS network. When there are few numbers of voice calls, the performance does not vary that much between these two networks.
### Table 4.16 QoS Metrics for LTE Traffic

<table>
<thead>
<tr>
<th>Number of voice calls</th>
<th>End-to-end delay (msec)</th>
<th>Packet loss (%)</th>
<th>Jitter (msec)</th>
<th>$P_{\alpha}^{0}$</th>
<th>$P_{\beta}^{0}$</th>
<th>$P_{\gamma}^{0}$</th>
<th>QoSAM*</th>
<th>Overall QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>0.092</td>
<td>0.02</td>
<td>20</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
<tr>
<td>10</td>
<td>0.092</td>
<td>0.06</td>
<td>24</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
<tr>
<td>12</td>
<td>0.093</td>
<td>0.08</td>
<td>28</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
<tr>
<td>14</td>
<td>0.094</td>
<td>0.12</td>
<td>30</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
<tr>
<td>16</td>
<td>0.095</td>
<td>0.15</td>
<td>32</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
<tr>
<td>18</td>
<td>0.099</td>
<td>0.18</td>
<td>34</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
</tbody>
</table>

### Table 4.17 Comparisons of QoS Metrics for LTE and UMTS Network

<table>
<thead>
<tr>
<th>Radio Access Network</th>
<th>Applications</th>
<th>Number of users</th>
<th>Application QoS Metric</th>
</tr>
</thead>
<tbody>
<tr>
<td>UMTS</td>
<td>Voice, VS</td>
<td>8</td>
<td>V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>10</td>
<td>VS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>14</td>
<td>VS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>20</td>
<td>VS</td>
</tr>
<tr>
<td>LTE</td>
<td>Voice, VS</td>
<td>8</td>
<td>V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>10</td>
<td>V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>14</td>
<td>V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>20</td>
<td>V</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Number of users</th>
<th>Application QoS Metric</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>0.804</td>
</tr>
<tr>
<td>10</td>
<td>0.64</td>
</tr>
<tr>
<td>14</td>
<td>0.52</td>
</tr>
<tr>
<td>20</td>
<td>0.40</td>
</tr>
<tr>
<td>8</td>
<td>1</td>
</tr>
<tr>
<td>10</td>
<td>1</td>
</tr>
<tr>
<td>14</td>
<td>1</td>
</tr>
<tr>
<td>20</td>
<td>1</td>
</tr>
<tr>
<td>8</td>
<td>0.85</td>
</tr>
</tbody>
</table>

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4.5.3 Other Heterogeneous Network-based Scenarios

In this stage, the QoS of heterogeneous scenarios is analysed. At first, the performance is examined for the VC sessions with different number of active users in the network. In these scenarios, several communication technologies are considered. The below acronyms are used for each communication technology in the following calculations:

- Ethernet – E
- WiMAX- W
- WLAN- WL
- UMTS – U
- LTE - L

With three users on the Ethernet-Ethernet RAN, the users experience a 5.8 msec delay, 0% packet loss, and 1.4 msec jitter. The performance metric of these parameters in VC session for this scenario using equation (4.5) is calculated as:

\[ E,E_D,VC \]

\[ E,E_J,VC \]

\[ E,E_P,VC \]

\[ \frac{12\text{ Ethernet, Ethernet}}{5 \cdot 4.98 \cdot 0.005 \cdot 51} = 0.005 \]

The application QoS metric using equation (4.6) is calculated as:
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\[ QoSAM_{VC}^{EE} = 1 \times 0.4 + 1 \times 0.4 + 0.005 \times 0.2 = 0.80 \]

Other calculations have been also conducted using a similar fashion.

Table 4.18 shows the values calculated for the VC sessions and the overall QoS recommendation.

<table>
<thead>
<tr>
<th>RANs</th>
<th>Number of users</th>
<th>Specifications</th>
<th></th>
<th>Overall QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>End-to-end delay (msec)</td>
<td>Packet loss (%)</td>
<td>Jitter (msec)</td>
</tr>
<tr>
<td>Ethernet-Ethernet</td>
<td>10</td>
<td>5.8</td>
<td>4.98</td>
<td>1.4</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>5.8</td>
<td>4.36</td>
<td>1.4</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>5.8</td>
<td>0</td>
<td>1.4</td>
</tr>
<tr>
<td>WLAN-Ethernet</td>
<td>10</td>
<td>5.8</td>
<td>19.5</td>
<td>0.2</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>5.8</td>
<td>3.3</td>
<td>0.2</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>5.8</td>
<td>3.3</td>
<td>0.2</td>
</tr>
<tr>
<td>WLAN-WiMAX</td>
<td>10</td>
<td>240</td>
<td>4</td>
<td>65</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>240</td>
<td>1.65</td>
<td>20</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>7.9</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>UMTS-WiMAX</td>
<td>10</td>
<td>260</td>
<td>26</td>
<td>85</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>220</td>
<td>13</td>
<td>70</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>140</td>
<td>3</td>
<td>14</td>
</tr>
</tbody>
</table>

The analysis using the RAN QoS evaluation function shows that the network performs well when there are three and five participants in each RAS cluster. The performance of WLAN-WiMAX RAN degrades by 32% when five more users join the conference. On the other hand, the performance improves by 43% when seven users leave the conference. The Ethernet-Ethernet and the WLAN-Ethernet network show a consistent performance. The increased or decreased numbers of participants
affect the performance of these clusters to a limited extent. The participants in the UMTS-WiMAX RAN experience comparatively a lower QoS than the participants of the other RANs. The performance of this cluster decreases by 77% when seven participants join the conference. This network performs well with the limited number of VC participants. Figure 4.21 shows the performance comparisons of different access networks in the rural area side.

![Figure 4.21 The Performance Comparisons of different Access Networks](image)

Figure 4.22 shows the performance of WLAN network in combination with the WiMAX and Ethernet networks. With an increased number of active users, the...
performance of WLAN-WiMAX network degrades by 23% than WLAN-Ethernet network. With a few number of users, three, in this case, the wireless network combinations show better performance.

The QoS metrics for Voice conferencing is also evaluated in the same manner. Table 4.19 shows the QoS analysis for voice conferencing. UMTS-WiMAX experiences higher packet delay as compared to Ethernet-Ethernet networks. This is due to the mobility of some of the users in the UMTS networks. During the video and voice conferencing sessions, some of the users moved away from the base stations and encountered comparatively higher end-to-end delay. The results presented in Table 4.19 are illustrated by following calculations in detail.

### Table 4.19 QoS Analysis of Voice Conferencing

<table>
<thead>
<tr>
<th>RANs</th>
<th>Number of users</th>
<th>Specifications</th>
<th>QoSAM&lt;sup&gt;R,RS&lt;/sup&gt;&lt;sub&gt;A&lt;/sub&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>End-to-end delay (msec)</td>
<td>Packet loss (%)</td>
</tr>
<tr>
<td>Ethernet-Ethernet</td>
<td>10</td>
<td>80</td>
<td>3.84</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>80</td>
<td>0.315</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>80</td>
<td>0</td>
</tr>
<tr>
<td>WLAN-Ethernet</td>
<td>10</td>
<td>80</td>
<td>4.165</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>79</td>
<td>0.34</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>78</td>
<td>0</td>
</tr>
<tr>
<td>WLAN-WiMAX</td>
<td>10</td>
<td>82.5</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>82</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>81.5</td>
<td>0</td>
</tr>
<tr>
<td>UMTS-WiMAX</td>
<td>10</td>
<td>130</td>
<td>3.125</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>130</td>
<td>2.25</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>130</td>
<td>1.865</td>
</tr>
</tbody>
</table>

With three users in the Ethernet-Ethernet network, the users experience 80 msec delay, 0% packet loss, and around zero msec jitter. The performance metrics of these parameters in the voice conferencing session for this scenario using equation (4.5) are calculated as:

\[
P_{\text{QoS}_{\text{AV}}} = 1
\]

\[
P_{\text{QoS}_{\text{JV}}} = 1
\]
Chapter 4: Fixed Weight-based QoS Analysis

The application QoS metric using equation (4.6) is calculated as:

\[ QoSAM_{V}^{E,E} = 1 \]

With five users on the Ethernet-Ethernet RAN, the application QoS metric is also calculated as 1.

With ten users on the Ethernet-Ethernet RAN, the users experience 80 msec delay, 3.84% packet loss, and around 0.25 msec jitter. The performance metrics of these parameters in the voice conferencing session for this scenario using equation (4.5) are calculated as:

\[ P_{D,V}^{E,V} = 1 \]

\[ P_{D,V}^{E,V} = \frac{20 - 3.84}{20 - 3} = 0.95 \]

\[ P_{PL,V}^{E,V} = 1 \]

The application QoS metric using equation (4.6) is calculated as:

\[ QoSAM_{V}^{E,E} = 1 \times 0.4 + 1 \times 0.4 + 0.95 \times 0.2 = 0.99 \]

Figure 4.23 The Performance Analysis of UMTS-WiMAX RAN for different Applications

Other results are calculated using the same equations.
In this case, the participants in the WLAN-WiMAX RAN experience the best performance indicating a consistent QoS value. The voice conferencing participants on the UMTS-WiMAX RAN experience better performance than the VC participants. Figure 4.23 compares the VC and voice conferencing performance on the UMTS-WiMAX RAN. The performance of this RAN is improved by 77% with 10 participants, and 69% with 5 participants in the case of voice conferencing.

Figure 4.24 shows the comparison of voice and VC application performance on the Ethernet-Ethernet network. The voice conferencing on the Ethernet-Ethernet network shows a 19% improvement for 20 active users and 17% for 10 active users in the network compared to VC.

Figure 4.25 demonstrates the performance comparison of WLAN-Ethernet network in the case of voice and VC applications. With 20 active users on the network, voice conferencing shows 18% better performance and with 10 active users in the network, this figure is 11%.
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Figure 4.26 compares the performance of voice and VC applications on the WLAN-WiMAX network. With 20 active users in the network, the voice conferencing shows 43% better performance than VC. With 10 active users in the network, this figure is 11%.

Now, the RAN QoS Metric is calculated for these networks. As both applications have the same number of users, therefore, a weight of 0.5 is assigned to each of this application in order to normalise the RAN QoS Metric values. Table 4.20 shows the QoS analysis of different RANs.

Some of the calculation results presented in Table 4.20 are illustrated in detail here. When there are 10 VC and ten voice users on the WLAN-WiMAX RAN, the RAN QoS metric is calculated using equation (4.7) as:

\[ QoS_{RM}^{WLAN,WiMAX} = 0.57 \times 0.5 + 1 \times 0.5 = 0.785 \]

When there are five VC and five voice users on the WLAN-WiMAX RAN, the RAN QoS metric is calculated as:

\[ QoS_{RM}^{WLAN,WiMAX} = 0.89 \times 0.5 + 1 \times 0.5 = 0.945 \]

For three VC and three voice users on the WLAN-WiMAX RAN, the RAN QoS metric is calculated as:

\[ QoS_{RM}^{WLAN,WiMAX} = 1 \]
When there are 10 VC and 10 voice users on the WLAN-Ethernet network, the RAN QoS metric is calculated as:

\[ QoSM_{\text{WL,E}} = 0.8 \times 0.5 + 0.99 \times 0.5 = 0.895 \]

For five VC and five voice users on the WLAN-Ethernet network, the RAN QoS metric is calculated as:

\[ QoSM_{\text{WL,E}} = 0.886 \times 0.5 + 1 \times 0.5 = 0.943 \]

For three VC and three Voice users on the WLAN-Ethernet network, the RAN QoS metric is:

<table>
<thead>
<tr>
<th>TABLE 4.20 QoS ANALYSIS OF DIFFERENT RANs</th>
</tr>
</thead>
<tbody>
<tr>
<td>RANs</td>
</tr>
<tr>
<td>-----------------------</td>
</tr>
<tr>
<td>Ethernet-Ethernet</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>WLAN-Ethernet</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>WLAN-WiMAX</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>UMTS-WiMAX</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>
The analysis demonstrates that the participants on the WLAN-WiMAX RAN experience a 17.5% better performance compared to the participants on the UMTS-WiMAX RAN when twenty people join the conference. In the case of ten participants, the WLAN-WiMAX RAN users experience 29% better performance. The WLAN-Ethernet RAN experiences 11% better performance than the WLAN-WiMAX RAN when there are 20 active users in the network and the WLAN-WiMAX RAN experiences a slightly better performance than the WLAN-Ethernet RAN when there are 10 active users in the network.

### 4.6 Summary

In this chapter, a fixed weight-based QoS evaluation method has been proposed and evaluated. Through different simulation scenarios and result analysis, the efficiency of this method is presented. In the network planning and designing stage, it is possible to evaluate the QoS of heterogeneous network-based service model applying this method. For instance, the values of RAN QoS metrics, which are calculated using the simulation results, show that when the network is congested, the LTE-based networks perform better compared to the UMTS-based networks. On the other hand, with a few of number users, the UMTS-based networks almost have the same performance as the LTE-based network. Such observations, which are derived using the values of RAN QoS metrics, could be used to recommend the LTE-based networks in a densely populated rural area and the UMTS-based networks for a less densely populated rural area. It is also possible to use this method for QoS management and monitoring. For example, when the codec of the VS session has been changed from H.263 to MPEG-4, the performance of the network degrades by 35.24%. The unified application, RAN and network QoS metrics can remove the overhead of the service providers to analyse each dynamic of the network individually for QoS evaluation. For example, using the QoS analysis, it was easy to identify that the WLAN in combination with the WiMAX technology is more
suitable for VC services than the UMTS RAN or the UMTS-WiMAX network when there are an increased number of users. Whereas, using the result analysis, the values of delay, jitter, and packet loss are compared separately for the performance evaluation of VC sessions in each RAN. The end-users are also left with the options to choose the most suitable network, according to their requirements by simply analyse these metric values. They can also investigate if they are getting the services they are paying for.

However, this method has some limitations in relation to weight assignment. A fixed weight is used throughout this method for all the QoS-related parameters, applications, and RANs for QoS evaluation. These weights are subject to change following the network transition and additional service requirements. One such example is varying user demands of applications from an industry to a home environment. Additionally, not all the applications are actively used in a network. Some applications may be absent from a network. To deal with such case, a dynamic weight calculation method would be more appropriate than the fixed weight-based method to handle these situations. Therefore, the next chapter presents a more complex QoS analysis method that applies dynamic weights.
Chapter 5

Dynamic Weight-based QoS Evaluation

The presence of different types of communication technologies and a varying number of users make the QoS evaluation in a heterogeneous network a very challenging task. In the previous chapter, the simulation data analysis has demonstrated that the same network configuration can suggest varying QoS results based on the measurement of packet loss and end-to-end delay. For example, with twenty voice clients and one streaming client on the network, the end-to-end delay of voice traffic shows an acceptable value and in terms of packet loss, the voice calls do not achieve an acceptable performance value for the same number of users. On the other hand, the same network shows an acceptable performance for voice calls with twelve voice clients and one VS client in terms of packet loss. Therefore, in some networks, while the end-to-end delay may be at an acceptable level for some applications, packet loss may simply be too high. In addition, the effects of different communication technologies on the performance of various applications, say, voice and video, must be accounted for in an efficient and methodical manner.

In the previous chapter, a fixed weight-based QoS evaluation method is proposed and investigated. The method has some limitations in relation to weight assignment as it has used fixed values for QoS-related parameter and application weights. However, in reality, these weights cannot be always a fixed value. They can have dynamic value depending on the particular context, such as a health service-related video conference, casual conversational call or a video lecture related to an online course. Additionally, not each application is active in a network all the time. The absence or the presence of the application in a network will also have an impact on these weights. Therefore, a weight assignment method that incorporates these changing circumstances of the network is proposed and evaluated in this chapter. Using this method, the QoS level of the network can reflect the active changes in the network. This method is able to evaluate the QoS of more complex networks with relatively
Chapter 5: Dynamic Weight-based QoS Evaluation

ever-changing applications and technologies. This method also considers application importance while QoS evaluation. The method is more costly to implement, however, more efficient and accurate.

The rest of the chapter is arranged as follows: Section 5.1 outlines the introduction, and Section 5.2 discusses the proposed dynamic weight-based QoS evaluation method. Section 5.3 presents the simulation scenarios. Section 5.4 analyses the impacts of application significance weights on the network QoS level. In Section 5.5, some of the simulation scenarios are analysed using the proposed weight assignment method. Section 5.6 presents the performance comparisons of this method with the other methods. Section 5.7 summarises this chapter and gives a glimpse view of the next chapter.

5.1 Introduction
The primary issue with the conventional QoS analysis method is that the parameter weights are assigned to QoS-related parameters according to separate objective functions, and the significances of the present applications in the network are ignored. These weights have both subjective and objective characteristics. For example, the importance of network-related parameters such as Received Signal Strength (RSS) and bandwidth are objective in nature. Application-related parameters such as end-to-end delay, packet loss, and jitter seem to be objective in nature. However, some studies have demonstrated that in some cases, they could be subjective in nature too. For example, a study conducted in Tanzania, which is discussed in Chapter 3, shows that the users give moderate importance to end-to-end delay over packet loss. The study by ETSI reveals that the users give strong importance to end-to-end delay over packet loss. These studies have already been highlighted in Chapter 3. Therefore, the importance of application-related performance parameters can vary based on changing contexts, for example, between home and industrial environments or urban and rural areas. The significance of applications can vary depending on the context as well. For example, an application under the education service would have more importance than an application under the entertainment service. Moreover, the absence or presence of an application in the network will affect the weights of other applications.
During the weight assignment, to bring the above-discussed context-based information into the picture, FAHP with extent analysis method is applied. This method is capable of handling ambiguity in any particular subject. It is also possible to assign the weights dynamically to the relevant parameters by using this method.

The significance weights of applications are considered during the QoS evaluation of radio access networks. To determine the weights of the available applications in the network, the purpose and the number of users using those applications are considered. The radio access network metric can label a network as any particular service-oriented network such as education or health by integrating the application weights. For instance, if the radio access network QoS metric value of a network, which is heavily used for health service relevant applications, is good, then that network can be taken as a suitable health service-oriented network for any future use. These concepts are discussed in detail in the following example.

**Table 5.1 Example of Application Weight Calculation**

<table>
<thead>
<tr>
<th>Networks</th>
<th>Considered Parameters</th>
<th>Weights</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Application</td>
<td>Service</td>
</tr>
<tr>
<td>A</td>
<td>x</td>
<td>Education</td>
</tr>
<tr>
<td></td>
<td>y</td>
<td>Entertainment</td>
</tr>
<tr>
<td>B</td>
<td>x</td>
<td>Health</td>
</tr>
<tr>
<td></td>
<td>y</td>
<td>Education</td>
</tr>
</tbody>
</table>

Table 5.1 shows the example of two networks, $A$ and $B$, which have applications $x$ and $y$ with different number of users. The application weights are expressed as $w^x_A$, $w^y_A$, $w^x_B$, and $w^y_B$. In this thesis, the weights of these applications are defined based on two criteria, the importance of the service, which it is under, and the number of users using that application. Other evaluation rules can be integrated based on individual needs. In the network $A$, it is considered that $w^x_A > w^y_A$. This is determined depending on the fact that in the network $A$, the application $x$ is used by more users than the application $y$ and it is used for educational services, whereas, the application $y$ is used for entertainment services. Therefore, when the QoS metric in the network $A$ is calculated considering these application weights, the QoS metric value reflects the significance of the service the application is under. As a result, if
the QoS value is good for the application $y$ and poor for the application $x$, the outcome of the QoS level of the network will be poor as the majority of the users in that network experience poor performance for an important service. If, in any case, the entertainment service application has more users, the result will also be same as the education service is set to have higher significance than the entertainment service. These findings can change based on specific network requirements.

On the other hand, Network $B$ supports both education and health services. As more users are using the health services compared to the education services, the application weight of $x$ is greater than $y$, $w^x_B > w^y_B$. If the QoS value of network $B$ is good, then it can be categorised as a health service-oriented network. Therefore, the configurations of $B$ can be recommended for any network that aims to deploy network-based health services in the future. Service operators can input these criteria to change the weights dynamically for any network.

### 5.2 Dynamic Weight-based QoS Evaluation

In this section, the steps of the proposed Dynamic weight-based QoS Evaluation Method (DWQEM) are discussed in detail. The proposed DWQEM uses a hierarchical calculation strategy. It involves a four-step calculation process. In the first step, a performance metric for each of the QoS-related parameter is calculated. Using this metric value and the related weight for each of the QoS-related parameters, an integrated QoS metric for the ongoing applications in each RAN is quantified. Then in the third step, the integrated metric for the present RANs in the network are calculated based on the application weights and the application QoS metric. Finally, the values of all RAN QoS metrics are combined to derive the integrated QoS metric for the whole network configuration.

The proposed method involves two major phases. In the first phase, similar to the fixed weight-based method, a set of QoS-related parameters are selected for each application, and then context-based ranges are established for them. The application weights are also established in this phase. The second step involves the actual calculations of the set of integrated metrics. A set of RANs present in any network is defined as $R_j$, $j = \{1, 2, ..., n\}$. For example, $R_1 = UMTS, R_2 = WLAN, R_3 = WiMAX, R_4 = LTE$ etc. A set of available
applications in each RAN is defined as $A_i$, $i=\{1,2,...,m\}$. For example, $A_1 = \text{Voice}, A_2 = \text{VC}, A_3 = \text{VS}$ etc. A set of QoS-related parameters for each application is defined as $Q_{PR_{k,A}}$, where $A$ denotes the application, $R$ denotes the radio access network, and $k$ denotes the QoS-related parameter for application $A$. The steps of the proposed DWQEM are illustrated in Figure 5.1. These steps are explained in more detail as follows.

**Figure 5.1 Workflow for the Proposed QoS Evaluation Method**

**Phase 1:**

- At first, a set of QoS-related parameters referred to as $Q_{PR_{k,A}}$ for each application is selected. This selection is based on their significance to the performance evaluation of that particular application. Then context-based benchmark ranges for each of this parameter is established.

- Then, the weights for the considered QoS-related parameter in each application are established. This weight is referred to as $W_{Q_{PR_{k,A}}}$. This weight is calculated...
using the context-based importance information, which is gathered from different recommendations available in the literature.

- A weight for each application referred to as $W_A^R$ is defined, where $A$ is the active application in any RAN $R$. This weight is determined according to the context-based importance of that application.

- A weight for each RAN referred to as $W_R^N$ is defined, where $R$ is the radio access network in any network $N$. This weight is determined based on the number of users present in that RAN. The more users it has a larger weight it is assigned.

**Phase 2:**

- A Performance Metric refers to as $P_{QP,kA}^R$ is calculated for each QoS-related parameter. The benchmark range and a Real-Time Performance Metric refer to as $RTP_{QP,kA}^R$ of that parameter is used for the calculation. This $RTP_{QP,kA}^R$ is collected from the ongoing session of the application considered.

- The QoS-related parameter weight ($W_{QP,kA}^R$) and performance metric ($P_{QP,kA}^R$) are used to calculate an application-based QoS metric for the application $A$ under the RAN $R$, which is referred to as $QoSAM_A^R$.

- Then the application weight ($W_A^R$) and the application QoS metric $QoSAM_A^R$ are used to calculate the RAN QoS metric $QoSRM_N^R$ for any RAN $R$, where $N$ is considered as the whole network configuration.

- Finally, all the $QoSRM_N^R$ metrics are amalgamated to derive the network configuration QoS metric $QoSCM_N$ for the given network configuration.

The first step of phase 1, which involves identifying the key QoS-related parameters and their context-based benchmark ranges have been discussed in Chapter 3 in detail. The steps of phase 2 have been discussed in Chapter 4. Therefore, in this chapter the dynamic weight assignment methods for QoS-related parameters and applications are studied in detail.
5.2.1 Dynamic Weight Calculation Details

The dynamic weight calculation method uses a set of criteria to determine the weights of QoS-related parameters and applications. In this work, the weights $W_{Q_1^{'},A}$ for the QoS-related parameters are calculated based on their impacts on the application performance. These effects are characterised according to the recommendations from different institutions and empirical results available in the literature from QoE measurements. The weights for the relevant applications $W_A^R$ are also determined in a similar way. Figure 5.2 depicts this calculation method regarding the application weight assignment. For example, in a particular network $N$, two service operators $A$ and $B$ have set different significance levels for voice, VC, and VS applications. To determine a weight for these applications, their importance level set up by the service operators are considered. In order to measure the QoS of the network $N$, the weights of the applications are applied. The FAHP with extent analysis method is used to determine the weight value of the applications, and finally QoS of the network $N$ is calculated. The FAHP method has been explained in Chapter 2. The weights are calculated dynamically by considering weight assignment.
criteria of fuzziness and the uncertainties. For example, users like to use terms such as “important,” “very important,” “less important” to express their views in evaluating the importance of the QoS-related parameters or applications rather than using any accurate value. People’s backgrounds also influence the judgements up to a certain limit the way they attach importance to these parameters. By using FAHP method, it is possible to take these kinds of expressions into account and define the weights dynamically for QoS-related parameters and applications.

The weight calculation involves two steps. At first, the alternatives, criteria, and the fuzzy judgement matrix are defined. Then in the second step, the actual weight is calculated based on those criteria. FAHP-based calculations include: establishing a set of alternatives \( X = \{x_1, x_2, \ldots, x_m\} \), a set of goal or evaluation criteria \( G = \{g_1, g_2, \ldots, g_n\} \), a fuzzy judgement matrix (FJM), with elements \( \tilde{r}_{ij} \) that represents the relative importance of each pair of criteria \( i \) and \( j \), and a weighting vector \( \mathbf{w} = (w_1, w_2, \ldots, w_n) \). Both steps involve the concept of Triangular Fuzzy Number (TFN) and fuzzy addition and multiplication operations. The details on TFNs and the operations of TFNs have been discussed in Chapter 2.

The first step involves the following process:

1) **Define hierarchy:** The hierarchy is defined based on the alternatives and criteria. In this case, when the weight is calculated for the QoS-related parameters, the applications are considered as the alternatives and the QoS-related parameters for each application are considered as the criteria. The reason for this, as discussed earlier, each application has a unique view of the importance of different parameters. Therefore, the set of the alternatives can be expressed as \( X = \{x_i \mid x_i \in A\} \) where \( x_1 = \text{Voice}, x_2 = \text{VC} \) etc., and \( A \) denotes application. The set of criteria can be expressed as: \( G = \{g_i \mid g_i \in QP\} \) where \( g_1 = \text{Delay}, g_2 = \text{Jitter} \) etc., and \( QP \) denotes QoS-related parameters.

When the weights are calculated for the applications, the RANs are treated as the alternatives and the applications are used as the criteria. Because a single application can have different significance values in each RAN, depending on separate criteria defined by the service operators. Therefore, the alternatives,
in this case, are expressed as $X = \{x_1 | x_1 \in R\}$, where $x_1 = UMTS$, $x_2 = WiMAX$ etc., and $R$ denotes Application. The set of criteria is expressed as $G = \{g_i | g_i \in A\}$ where $g_1 = Voice$, $g_2 = VC$ etc., and $A$ denotes application.

2) **Fuzzy pair-wise evaluation:** In this step, the pairwise comparison matrix is defined for all the criteria. TFNs $K_1, K_3, K_5, K_7, K_9$, are used to represent the views from equally important to extremely important. The TFNs $K_2, K_4, K_6, K_8$ represent the middle values. Table 5.2 shows the TFN $K_t = (l_t, m_t, u_t)$, where $t=1, 2, \ldots, 9$ and where $l_t$, $u_t$ and $m_t$ are the lower, upper and the middle value of the fuzzy number $K_t$ respectively. $\sigma$ represents the fuzzy degree of judgement where $\sigma = m_t - l_t = u_t - m_t$. When $\sigma = 0$, it is a non-fuzzy number. This value should be greater than or equal to one-half according to the study in [166]. Based on that study, the $\sigma$ is defined for this study.

### Table 5.2 A FAHP-based Pair-wise Comparison Importance Scale

<table>
<thead>
<tr>
<th>Fuzzy Numbers $k_t(l_t, m_t, u_t)$</th>
<th>Definition</th>
<th>Triangular Fuzzy Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>$k_1(l_1, m_1, u_1)$</td>
<td>Equal importance</td>
<td>$(1,1,1)$</td>
</tr>
<tr>
<td>$k_2(l_2, m_2, u_2)$</td>
<td>Intermediate values</td>
<td>$(1/2,3/4,1)$</td>
</tr>
<tr>
<td>$k_3(l_3, m_3, u_3)$</td>
<td>Moderate importance</td>
<td>$(2/3,1,3/2)$</td>
</tr>
<tr>
<td>$k_4(l_4, m_4, u_4)$</td>
<td>Intermediate values</td>
<td>$(1,3/2,2)$</td>
</tr>
<tr>
<td>$k_5(l_5, m_5, u_5)$</td>
<td>Strong importance</td>
<td>$(3/2,2,5/2)$</td>
</tr>
<tr>
<td>$k_6(l_6, m_6, u_6)$</td>
<td>Intermediate values</td>
<td>$(2,5/2,2)$</td>
</tr>
<tr>
<td>$k_7(l_7, m_7, u_7)$</td>
<td>Very strong importance</td>
<td>$(5/2,3,7/2)$</td>
</tr>
<tr>
<td>$k_8(l_8, m_8, u_8)$</td>
<td>Intermediate values</td>
<td>$(3,7/2,4)$</td>
</tr>
<tr>
<td>$k_9(l_9, m_9, u_9)$</td>
<td>Extreme importance</td>
<td>$(7/2,4,9/2)$</td>
</tr>
</tbody>
</table>
Table 5.3 represents the pair-wise comparison of the importance of the QoS-related parameters of voice applications. The criteria, which are considered, are delay, jitter, and packet loss. To derive the pair-wise comparisons of delay and packet loss, two recommendations are considered, which have been discussed in Chapter 3. According to the study conducted in Tanzania, the users give strong importance to packet loss over delay up to a certain extent. On the other hand, according to ETSI and Cisco, delay has a strong importance over packet loss in case of voice applications. To derive the comparisons of delay and jitter, the recommendations from Cisco, ETSI and the study from Olusegun Obafemi are used which have been discussed in Chapter 3. A fuzzy judgement matrix for the QoS-related parameters of voice application is formulated by applying the equation (2.26) and the data from Table 5.3. This is as follows:

\[
FJM_{\text{voice}} = \left[ \begin{array}{ccc}
Q_{P_1} & Q_{P_2} & Q_{P_3} \\
(1,1,1) & (0.84,1.25) & (0.915,1.2,1.5) \\
(0.84,1.25) & (1,1,1) & (1.5,2.5) \\
(1.2,1.5,1.84) & (0.4,0.5,0.67) & (1,1,1)
\end{array} \right]
\]

(5.1)
where \( Q_{P1} = {\text{Delay}}, Q_{P2} = {\text{Jitter}}, Q_{P3} = {\text{Packet loss}} \). The fuzzy number for the pair-wise comparison between \( Q_{P1} \) and \( Q_{P3} \), which is the third column of the first row of \( FJM_{\text{Voice}} \) is calculated as:

\[
\frac{3/2+1/3}{2}=0.915 \\
\frac{2+2/5}{2}=1.2 \\
\frac{5/2+1/2}{2}=1.5
\]

Table 5.4 shows the pair-wise comparison for the QoS-related parameters of VC applications. Three parameters, namely, delay, jitter, and packet loss are considered as the criteria for the performance evaluation. The comparisons are derived using the recommendations from Cisco, ETSI, Pinger reports and study of Verscheure, et al. These recommendations have been discussed in detail in Chapter 3.

A similar fuzzy judgement matrix like voice QoS-related parameters is formed for VC QoS-related parameters by applying the equation (2.26) and the data from Table 5.4:

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Source</th>
<th>Delay</th>
<th>Jitter</th>
<th>Packet loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay</td>
<td>Cisco/ETSI</td>
<td>(1, 1, 1)</td>
<td>(1,1,1)</td>
<td>(1,1,1)</td>
</tr>
<tr>
<td></td>
<td>Pinger Report</td>
<td>(1, 1, 1)</td>
<td>N/A</td>
<td>(2/3,1,3/2)</td>
</tr>
<tr>
<td></td>
<td>Verscheure et al., 1998</td>
<td>(1, 1, 1)</td>
<td>(1/2,2/3,1)</td>
<td>(1/2,2/3,1)</td>
</tr>
<tr>
<td>Jitter</td>
<td>Cisco/ETSI</td>
<td>(1, 1, 1)</td>
<td>(1,1,1)</td>
<td>(1, 1, 1)</td>
</tr>
<tr>
<td></td>
<td>Verscheure et al., 1998</td>
<td>(1,3/2,2)</td>
<td>(1,1,1)</td>
<td>(1,3/2,2)</td>
</tr>
<tr>
<td>Packet loss</td>
<td>Cisco/ETSI</td>
<td>(1, 1, 1)</td>
<td>(1, 1, 1)</td>
<td>(1, 1, 1)</td>
</tr>
<tr>
<td></td>
<td>Pinger Report</td>
<td>(2/3,1,3/2)</td>
<td>N/A</td>
<td>(1, 1, 1)</td>
</tr>
<tr>
<td></td>
<td>Verscheure et al., 1998</td>
<td>(1,3/2,2)</td>
<td>(1,3/2,2)</td>
<td>(1, 1, 1)</td>
</tr>
</tbody>
</table>
Chapter 5: Dynamic Weight-based QoS Evaluation

\[ FJM_{VC} = \begin{bmatrix} QP_1 & QP_2 & QP_3 \\ QP_1 & (1,1) & (0.75,0.84,0.5) & (0.72,0.89,1.17) \\ QP_2 & (0.84,1,1.25) & (1,1) & (1.5,2,2.5) \\ QP_3 & (1.2,1.5,1.84) & (0.4,0.5,0.67) & (1,1) \end{bmatrix} \] (5.2)

where \( QP_1 = \text{Delay}, QP_2 = \text{jitter}, QP_3 = \text{Packet loss}. \)

Table 5.5 shows the pair-wise significance comparison for the QoS-related parameters of VS applications. The pair-wise comparisons, in this case, are derived using the recommendations from Cisco, and ITU, which have been also discussed in Chapter 3. The fuzzy judgement matrix for VS application is as follows:

\[ FJM_{VS} = \begin{bmatrix} QP_1 & QP_2 \\ QP_1 & (1,1) & (0.34,0.42,0.59) \\ QP_2 & (2,2.5,3) & (1,1) \end{bmatrix} \] (5.3)

**Table 5.5 Pair-wise Comparisons for QoS-related Parameters of VS Application**

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Source</th>
<th>Delay</th>
<th>Packet Loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay</td>
<td>Cisco</td>
<td>(1, 1, 1)</td>
<td>(2/7,1/3,2/5)</td>
</tr>
<tr>
<td></td>
<td>ITU</td>
<td>(1, 1, 1)</td>
<td>(2/5,1/2,2/3)</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>Cisco</td>
<td>(5/2,3,7/2)</td>
<td>(1,1,1)</td>
</tr>
<tr>
<td></td>
<td>ITU</td>
<td>(3/2,2,5/2)</td>
<td>(1,1,1)</td>
</tr>
</tbody>
</table>

Using a similar approach, the comparison matrixes for the considered applications are also formulated. As previously mentioned, the significance of these applications is subject to change depending on the requirements of particular networks. The service operators can update these criteria according their specific circumstances. In this case, as the simulation studies are designed as distance education-based service models, the criteria have been formulated using some relevant studies [167]. One such study reveals that for distance education service, VC is a better choice compared to voice-based applications. On the other hand, in a more general sense, VC may be less significant than voice-based applications as the latter usually have more users. Therefore, the criteria considered in this regard are the number of users using the application and the purpose and the context of application usage. Table 5.6 shows the pair-wise comparison matrix for VC, voice, and VS applications. If one of the applications is absent from the network, these weights are subject to change.
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Now, the second step of the weight calculation is conducted which involves the actual weight calculation of the QoS-related parameters and applications. Different FAHP-based methods are proposed in the literature for weight calculation. The most prominent one is Chang’s extent analysis method [144]. This method is chosen as it provides easy and flexible options for calculation. The steps of the extent analysis method are as follows:

At first, the sums of the each row of the defined fuzzy comparison matrix are calculated. Then the normalization of the row sums is conducted using fuzzy multiplication to obtain fuzzy synthetic analysis. Therefore, in the fuzzy comparison matrix, the fuzzy synthetic analysis of criteria $G_i$ of alternative $X_m$ is calculated as:

$$D_{G_i}^{X_m} = \left[ \sum_{j=1}^{n} \tilde{r}_{ij} \right]^{-1} \left[ \sum_{i=1}^{n} \sum_{j=1}^{n} \tilde{r}_{ij} \right]$$

where $i, j = \{1, 2, 3, \ldots, n\}$ and $n$ is the number of criteria.

In step 2, in order to rank the criteria against each alternative, the degree of possibility of two fuzzy numbers is applied. Therefore,

$$D_{G_i}^{X_m}(l_2,m_2,u_2) \geq D_{G_i}^{X_m}(l_1,m_1,u_1)$$

is computed by the following equation:

---

### Table 5.6 Pair-wise Comparison Matrix for Different Applications

<table>
<thead>
<tr>
<th>Applications</th>
<th>Criteria</th>
<th>VC</th>
<th>Voice</th>
<th>VS</th>
</tr>
</thead>
<tbody>
<tr>
<td>VC</td>
<td>Purpose of Usage</td>
<td>(1, 1, 1)</td>
<td>(3/2,2,5/2)</td>
<td>(2/3,1,3/2)</td>
</tr>
<tr>
<td></td>
<td>Number of Users</td>
<td>(1, 1, 1)</td>
<td>(2/3,1,3/2)</td>
<td>(1, 3/2, 2)</td>
</tr>
<tr>
<td>Voice</td>
<td>Purpose of Usage</td>
<td>(2/5,1/2,2/3)</td>
<td>(0.54, 0.75, 1.09)</td>
<td>(1, 1, 1)</td>
</tr>
<tr>
<td></td>
<td>Number of Users</td>
<td>(2/3,1,3/2)</td>
<td>(1, 1, 1)</td>
<td>(5/2,3,7/2)</td>
</tr>
<tr>
<td>VS</td>
<td>Purpose of Usage</td>
<td>(2/3, 1,3/2)</td>
<td>(0.59, 0.84, 1.25)</td>
<td>(2/3,1,3/2)</td>
</tr>
<tr>
<td></td>
<td>Number of Users</td>
<td>(1/2,2/3,1)</td>
<td>(2/7,1/3,2/5)</td>
<td>(1, 1, 1)</td>
</tr>
</tbody>
</table>
Chapter 5: Dynamic Weight-based QoS Evaluation

\[ V(D_{G_2}^{X_m} \geq D_{G_1}^{X_m}) = \sup \left[ \min \left( \mu_{G_1}(x), \mu_{G_2}(y) \right) \right] \]  \hspace{1cm} (5.5)

And can be also expressed as:

\[ V(D_{G_2}^{X_m} \geq D_{G_1}^{X_m}) = hgt \left( D_{G_2}^{X_m} \cap D_{G_1}^{X_m} \right) = \mu_{G_2}(d) \]

\[
\begin{align*}
1 & \quad \text{if } m_2 \geq m_1 \\
0 & \quad \text{if } l_1 \geq u_2 \\
\frac{l_1 - u_1}{(m_2 - u_2) - (m_1 - l_1)} & \quad \text{otherwise}
\end{align*}
\]

(5.6)

And \( V(D_{G_1}^{X_m} \geq D_{G_2}^{X_m}) \)

\[
\begin{align*}
1 & \quad \text{if } m_2 \geq m_1 \\
0 & \quad \text{if } l_1 \geq u_2 \\
\frac{l_1 - u_1}{(m_1 - u_1) - (m_2 - l_2)} & \quad \text{otherwise}
\end{align*}
\]

(5.7)

where \( d \) is the ordinate to validate if the highest intersection point \( D \) is between \( \mu_{G_2}^{X_m} \) and \( \mu_{G_1}^{X_m} \). Both the values of \( V(D_{G_1}^{X_m} \geq D_{G_1}^{X_m}) \) and \( V(D_{G_2}^{X_m} \geq D_{G_2}^{X_m}) \) are required to compare \( \mu_{G_2}^{X_m} \) and \( \mu_{G_1}^{X_m} \). For large numbers of criteria, the degree of possibility is applied as:

\[ V(D_{G_1}^{X_m} \geq D_{G_1}^{X_m}, D_{G_1}^{X_m}, \ldots, D_{G_1}^{X_m}) = V\left( (D_{G_1}^{X_m} \geq D_{G_1}^{X_m}) \text{ and } (D_{G_1}^{X_m} \geq D_{G_1}^{X_m}) \text{ and } \ldots \text{ and } (D_{G_1}^{X_m} \geq D_{G_1}^{X_m}) \right) \]

\[ = \min V\left( d_{G_1}^{X_m} \geq d_{G_1}^{X_m} \right) \]  \hspace{1cm} (5.8)

Assume that \( d'(C_{G_1}^{X_m}) = \min V\left( d_{G_1}^{X_m} \geq d_{G_1}^{X_m} \right) \)

In step 3, the weight vector \( \mathbf{w} \) for each alternative is calculated. This is obtained as:

\[ \mathbf{w}_m = \left( d'(C_{G_1}^{X_m}), d'(C_{G_2}^{X_m}), \ldots, d'(C_{G_1}^{X_m}) \right)^T \]  \hspace{1cm} (5.9)

In step 4, the normalised weight vector is calculated for each alternative as:
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\[ w_m = \left( d\left(C_{G_i}^{X_{n}}\right), d\left(C_{G_j}^{X_{n}}\right), \ldots, d\left(C_{G_k}^{X_{n}}\right) \right)^T \]

\[ = \left( \frac{d(C_{G_i}^{X_{n}})}{\sum_{j=1}^{n} C_{G_i}^{X_{n}}} \frac{d(C_{G_j}^{X_{n}})}{\sum_{j=1}^{n} C_{G_j}^{X_{n}}} \ldots \frac{d(C_{G_k}^{X_{n}})}{\sum_{j=1}^{n} C_{G_k}^{X_{n}}} \right) \]  

(5.10)

The following are the examples of weight calculation for the QoS-related parameters and applications.

1. **Weight Calculation for the QoS-related parameters of Voice Application:**

   Equation (5.4) is applied to calculate the fuzzy synthetic degrees of delay, jitter, and packet loss for the voice application. For simplicity of expressions, delay, jitter and packet loss are expressed as D, J and PL subsequently. With index 1, 2, and 3 representing delay, jitter and packet loss, respectively, the calculation is as follows:

   \[ \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} = ((1,1,1) \oplus (0.84,1,1.25) \oplus (0.915,1.2,1.5) \ldots \oplus (1,1,1)) \]

   \[ = (8.7,10.2,12.01) \]

   \[ \left( \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} \right)^{-1} = (0.083,0.09,0.11) \]

   \[ \sum_{j=1}^{3} r_{ij} = (1,1,1) \oplus (0.84,1,1.25) \oplus (0.915,1.2,1.5) = (2.755,3.2,3.75) \]

   \[ \sum_{j=1}^{3} r_{ij} = (1,1,1) \oplus (1,1,1) \oplus (1.5,2,2.5) = (3.34,4,4.75) \]

   \[ \sum_{j=1}^{3} r_{ij} = (1.2,1.5,1.84) \oplus (0.4,0.5,0.67) \oplus (1,1,1) = (2.6,3,3.51) \]

   Therefore, the fuzzy synthetic degrees of delay, jitter, and packet loss are:

   \[ D_{D}^{Voice} = \sum_{j=1}^{3} r_{ij} \ominus \left[ \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} \right]^{-1} = (0.23,0.29,0.412) \]

   \[ D_{J}^{Voice} = \sum_{j=1}^{3} r_{ij} \ominus \left[ \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} \right]^{-1} = (0.28,0.36,0.52) \]
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\[ D_{PL}^{\text{Voice}} = \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} \] 
\[ = \begin{bmatrix} 0.21 & 0.27 & 0.39 \\ 0.21 & 0.27 & 0.39 \\ 0.21 & 0.27 & 0.39 \end{bmatrix}^{-1} = (0.21, 0.27, 0.39) \]

Then the degrees of possibility for these parameters are calculated using (5.6), (5.7), and (5.8):

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = \frac{l_{2} - u_{1}}{(m_{1} - u_{1}) - (m_{2} - l_{2})} = 0.65 \]

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = 1 \]

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = 1 \]

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = 1 \]

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = 0.88 \]

\[ V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}} \right) = 0.5 \]

\[ d'(C_{D}^{\text{Voice}}) = \min V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}}, D_{PL}^{\text{Voice}} \right) = \min \{0.65, 1\} = 0.65 \]

\[ d'(C_{D}^{\text{Voice}}) = \min V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}}, D_{PL}^{\text{Voice}} \right) = \min \{1, 1\} = 1 \]

\[ d'(C_{D}^{\text{Voice}}) = \min V\left(D_{D}^{\text{Voice}} \geq D_{j}^{\text{Voice}}, D_{PL}^{\text{Voice}} \right) = \min \{0.88, 0.5\} = 0.5 \]

The weight vector of the voice service QoS-related parameter is calculated using (5.9) and (5.10) as:

\[ w_{\text{Voice}} (D,J,PL) = (0.30, 0.47, 0.23) \]

2. Weight Calculation for the QoS-related parameters of VC Application:
   Equation (5.4) is applied to calculate the fuzzy synthetic degrees of delay, jitter, and packet loss for the VC application as well. With index 1, 2, and 3 representing delay, jitter and packet loss, respectively, the calculation is as follows:

\[ \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} = \left( \{(1,1) \oplus (0.75,0.84,0.5) \oplus (0.72,0.89,1.17) \ldots \oplus (1,1) \} \right) = (8.41, 9.73, 10.68) \]

\[ \left( \sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} \right)^{-1} = (0.09, 0.10, 0.12) \]
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\[ \sum_{j=1}^{3} r_{ij} = (1,1,1) \oplus (0.75,0.84,0.5) \oplus (0.72,0.89,1.17) = (2.47,2.73,2.67) \]

\[ \sum_{j=1}^{3} r_{ij} = (0.84,1,1.25) \oplus (1,1,1) \oplus (1.5,2,2.5) = (3.34,4,4.5) \]

\[ \sum_{j=1}^{3} r_{ij} = (1.2,1.5,1.84) \oplus (0.4,0.5,0.67) \oplus (1,1,1) = (2.6,3,3.51) \]

Therefore, the fuzzy synthetic degrees of delay, jitter, and packet loss are:

\[ D_{ij}^{VC} = \sum_{i=1}^{3} r_{ij} \otimes \left[ \sum_{j=1}^{3} \sum_{i=1}^{3} r_{ij} \right]^{-1} = (0.22,0.27,0.32) \]

\[ D_{ij}^{VC} = \sum_{i=1}^{3} r_{ij} \otimes \left[ \sum_{j=1}^{3} \sum_{i=1}^{3} r_{ij} \right]^{-1} = (0.30,0.4,0.54) \]

\[ D_{ij}^{VC} = \sum_{i=1}^{3} r_{ij} \otimes \left[ \sum_{j=1}^{3} \sum_{i=1}^{3} r_{ij} \right]^{-1} = (0.23,0.3,0.42) \]

Then the degrees of possibility for these parameters are calculated using (5.6), (5.7), and (5.8):

\[ V(D_{ij}^{VC} \geq D_{ij}^{VC}) = \frac{\mu_j - u_j}{(m_1 - u_j) - (m_2 - \mu_j)} = 0.13 \]

\[ V(D_{ij}^{VC} \geq D_{ij}^{VC}) = 0.75 \]

\[ V(D_{ij}^{VC} \geq D_{ij}^{VC}) = 1 \]

\[ V(D_{ij}^{VC} \geq D_{ij}^{VC}) = 1 \]

\[ V(D_{ij}^{VC} \geq D_{ij}^{VC}) = 0.54 \]

\[ d'(C_{ij}^{VC}) = \min \{ V(D_{ij}^{VC} \geq D_{ij}^{VC}, D_{ij}^{VC}) \} = 0.13 \]

\[ d'(C_{ij}^{VC}) = \min \{ V(D_{ij}^{VC} \geq D_{ij}^{VC}, D_{ij}^{VC}) \} = 1 \]

\[ d'(C_{ij}^{VC}) = \min \{ V(D_{ij}^{VC} \geq D_{ij}^{VC}, D_{ij}^{VC}) \} = 0.54 \]

The weight vector of the VC related QoS-related parameters is calculated using (5.9) and (5.10) as:

\[ w_{VC}(D,J,PL) = (0.07, 0.6, 0.32) \]
3. Weight Calculation for the QoS-related parameters of VS Application:

Equation (5.4) is applied to calculate the fuzzy synthetic degrees of delay and packet loss for the VS application. With index 1 and 2 representing delay and packet loss, respectively, the calculation is as follows:

\[
\sum_{i=1}^{2} \sum_{j=1}^{2} r_{ij} = \left( (1,1,1) \oplus (0.35,0.42,0.54) \ldots \oplus (1,1,1) \right) = (4.35,4.92,5.54)
\]

\[
\left( \sum_{i=1}^{2} \sum_{j=1}^{2} r_{ij} \right)^{-1} = (0.18,0.20,0.23)
\]

\[
\sum_{j=1}^{2} r_{ij} = (1,1,1) \oplus (0.75,0.84,0.5) \oplus (0.72,0.89,1.17) = (1.35,1.42,1.54)
\]

\[
\sum_{j=1}^{2} n_{ij} = (0.84,1.1,2.5) \oplus (1,1,1) \oplus (1.5,2,2.5) = (3.3,3.5,4)
\]

Therefore, the fuzzy synthetic degrees for delay and packet loss are as follows:

\[
D_{VS}^{PL} = \sum_{j=1}^{2} n_{ij} \oplus \left[ \sum_{i=1}^{2} \sum_{j=1}^{2} r_{ij} \right]^{-1} = (0.243,0.284,0.35)
\]

\[
D_{VS}^{PL} = \sum_{j=1}^{2} n_{ij} \oplus \left[ \sum_{i=1}^{2} \sum_{j=1}^{2} r_{ij} \right]^{-1} = (0.54,0.7,0.92)
\]

Then, the degrees of possibilities for these parameters are calculated using (5.6), (5.7), and (5.8):

\[
V \left( D_{VS}^{PL} \geq D_{PL}^{VS} \right) = 0.72
\]

\[
V \left( D_{PL}^{VS} \geq D_{VS}^{PL} \right) = 1
\]

\[
d' \left( C_{VS}^{PL} \right) = \min V \left( D_{VS}^{PL} \geq D_{PL}^{VS} \right) = 0.72
\]

\[
d' \left( C_{PL}^{VS} \right) = \min V \left( D_{PL}^{VS} \geq D_{VS}^{PL} \right) = 1
\]

The weight vector of the VS application QoS-related parameters is calculated using (5.9) and (5.10) as:

\[
w_{VS} \left( D,PL \right) = (0.41,0.59)
\]
Table 5.7 shows the weights for the QoS-related parameters of each considered application.

4) Example of Weight Calculation for Applications: In this example, the weights for the considered applications are calculated. At first, the fuzzy synthetic analysis is obtained using the comparison matrix illustrated in Table 5.6. With index 1, 2, and 3 representing VC, voice, and VS respectively, the calculation is as follows:

\[
\sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij} = [(1,1) \oplus (1.09,1.5,2) \oplus (0.84,1.25,1.75) \ldots \oplus (1,1,1)] = (8.13,10.01,11.54)
\]

\[
\left(\sum_{i=1}^{3} \sum_{j=1}^{3} r_{ij}\right)^{-1} = (0.087,0.09,0.123)
\]

\[
\sum_{j=1}^{3} n_{j} = (1,1) \oplus (1.09,1.5,2) \oplus (0.84,1.25,1.75) = (2.93,3.75,4.75)
\]

\[
\sum_{j=1}^{3} n_{j} = (0.54,0.75,1.09) \oplus (1,1,1) \oplus (1.59,2,2.5) = (3.13,3.75,4.59)
\]

\[
\sum_{j=1}^{3} r_{ij} = (0.59,0.84,1.25) \oplus (0.48,0.67,0.95) \oplus (1,1,1) = (2.07,2.57,3.2)
\]

The fuzzy synthetic degree values of the considered services using (5.6), (5.7), and (5.8) are:
Then the degree of possibilities for these applications using (5.9) and (5.10) are calculated as:

\[ V \left( D_{VC}^{ULMTS} \geq D_{Voice}^{ULMTS} \right) = \frac{I_{2} - U_{j}}{(m_{j} - U_{j}) - (m_{2} - I_{2})} = 1 \]

\[ V \left( D_{VC}^{ULMTS} \geq D_{VS}^{ULMTS} \right) = 1 \]

\[ V \left( D_{Voice}^{ULMTS} \geq D_{VC}^{ULMTS} \right) = 0.94 \]

\[ V \left( D_{Voice}^{ULMTS} \geq D_{VS}^{ULMTS} \right) = 1 \]

\[ V \left( D_{VS}^{ULMTS} \geq D_{VC}^{ULMTS} \right) = 0.56 \]

\[ V \left( D_{VS}^{ULMTS} \geq D_{Voice}^{ULMTS} \right) = 0.61 \]

\[ d' \left( e_{VC}^{ULMTS} \right) = \min V \left( D_{VC}^{ULMTS} \geq D_{Voice}^{ULMTS}, D_{VS}^{ULMTS} \right) = \min[1,1] = 1 \]

\[ d' \left( e_{Voice}^{ULMTS} \right) = \min V \left( D_{Voice}^{ULMTS} \geq D_{VC}^{ULMTS}, D_{VS}^{ULMTS} \right) = \min[0.94,1] = 0.94 \]

\[ d' \left( e_{VS}^{ULMTS} \right) = \min V \left( D_{VS}^{ULMTS} \geq D_{VC}^{ULMTS}, D_{Voice}^{ULMTS} \right) = \min[0.56,0.61] = 0.56 \]

The weight vector of the considered services is calculated as:

\[ \mathbf{w}_{\mathbf{A}}^{'}(VC, Voice, VS) = (1, 0.94, 0.56) \]

The normalisation weight vector is calculated as:

\[ \mathbf{w}_{\mathbf{A}} = (w_{VC}, w_{Voice}, w_{VS}) = (0.4, 0.38, 0.224) \]

Figure 5.3 shows the weights for the considered applications that are derived from these calculations.
5.3 Simulation Scenarios

Most of the simulation scenarios considered in this chapter have been discussed in Chapter 4. A few more new simulation scenarios are outlined in this chapter. These scenarios can be categorised as WLAN-UMTS-WiMAX Integration and LTE-UMTS-Ethernet Integration.

5.3.1 WLAN-UMTS-WiMAX Integration

In this category, three phases of simulation studies are conducted. The main focuses of these studies are as follows:

1. Investigate the effects of VC transmission on the quality of voice communications.
2. Examine the impact of the presence of WLAN on the performance of the UMTS network.
3. Study the network capacity.

In the first phase of the simulations, a WLAN-UMTS-WiMAX integrated network in a rural-urban environment is simulated. The UMTS network has a three-cell architecture with each cell having a base station in the centre of the cell. Each cell has one VC participant and different number voice calls. There WLANs are placed on the UMTS network using a loose-coupling architecture. Each WLAN also has a VC participant. WiMAX network is on the urban side, which has one VC participant. Figure 5.4 shows the simulation design. The simulation configuration for UMTS, WLAN and WiMAX networks are presented in Section 3.4 of Chapter 3. The configurations for VC and voice applications are shown in Table 5.8 and Table 5.9. In the first phase, the number of voice calls varies; however, the number of VC users remains the same. In the first scenario, the voice calls are limited to six, in the second scenario, this is changed to twelve and in the third scenario the number of calls is increased to 20.

Figure 5.4 UMTS-WLAN-WiMAX Integration
In the second phase, a UMTS network with three hexagonal cells, each covering one kilometre of the area is simulated. The rural area is chosen as the background for the UMTS network, and the urban area is selected for the WiMAX network. A three-sector base station is placed in the joint of three cells to cover this three-kilometre area. There are eight ongoing voice calls in each UMTS cell. In the first scenario, three VC participants are placed in the UMTS network, one in each cell. These participants are in a VC session with another participant who is in the WiMAX network.

In the second scenario, the load of the UMTS network traffic is reduced by placing three WLAN hotspots, one in each UMTS cell. Each of this WLAN has one VC client. In the third scenario, the number of WLAN hotspots is reduced to one, and all the VC participants are placed in one WLAN hotspot. The UMTS-WLAN clients are...
in a tight-coupling architecture. In the fifth scenario, the network architecture for the fourth scenario is simulated with a loose-coupling architecture. Tight-coupling and loose-coupling architectures have been discussed in Chapter 2. For voice clients, the best effort type of service is used for local voice (within UMTS cells) transmissions, and interactive voice has been set up for voice transmissions in the VC sessions. The simulation settings are the same as the simulation settings of Chapter 3.

5.3.2 LTE-UMTS-Ethernet Integration

In this category, an integration of LTE, UMTS and Ethernet network is simulated. The focuses of these simulation studies are:

1. Compare the performance of UMTS and LTE network in the presence of different type of application traffic.
2. Investigate the capacity of UMTS and LTE access networks.

In the first scenario of this category, different number of voice calls is simulated in both UMTS and LTE network. The VC sessions are simulated between the UMTS and LTE network. One VS session is simulated among the Ethernet, UMTS, and LTE network. The VS server is placed on the Ethernet server, and the VS clients are placed on the LTE and UMTS network.

In the second scenario of this category, one VS server is set on the LTE network, and another VS server is placed on the Ethernet network. The VS clients are put on the LTE and the UMTS network. Figure 5.5 depicts the simulation scenario. The configuration of the LTE network is same as the configuration described in Chapter 3. The UMTS network uses the same configuration described in Chapter 3 as well.
5.4 Impact of Application Significance

In Chapter 4, the fixed application weights are used to analyse the performance of any access network. In contrast to that in this section, a dynamic weight is calculated for each application according to the changing circumstances of the network. These weights are entered as inputs to the QoS evaluation method to derive the network QoS metric.

Recall that in Chapter 4, the QoS for the UMTS-WiMAX integrated network scenarios have been analysed using fixed weight-based method. The voice application is set to have a higher importance than the VS application in those simulation scenarios and the weights have been fixed as 0.6 and 0.4. Here, a detailed analysis is conducted to present the effects of dynamic weights on the performance.
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of the same network. Figure 5.6 shows the QoS analysis of the scenario with twelve voice calls and one VS session on the network. The dynamic weights are calculated using the FAHP method discussed in section 5.2.1. The figure clearly indicates that when the voice and VS applications have equal importance, the access network has a good QoS level (e.g. 0.81) with an average QoS level (e.g. 0.62) for voice application and a good QoS level (e.g. 1) for VS application. When the importance level of voice application has been changed from having equal to extreme importance over VS application, the access network QoS comes down to an average value of 0.63. Although, the performance of the VS application is good, because of having a lower importance, it has minimal effect on the network QoS level. On the other hand, the voice application, because of being extremely important, has a greater impact on the network QoS level.

![Figure 5.6 The Effect of Application Significance on Network QoS](image)

Voice Application QoS: 0.62
VS Application QoS :1
Voice Weight: 0.97
VS Weight :0.03
Voice Application is extremely important than VS
Voice Weight: 0.6
VS Weight :0.4
Voice Application is moderately Important than VS
Voice Weight: 0.5
VS Weight :0.5
Voice and VS application Equally Important

487x514
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Figure 5.7 The Effect of different Application Significance on the Network Performance

Figure 5.8 The Performance Analysis with different Application Significance
Figure 5.7 shows a similar type of analysis with the altered significances of voice and VS applications. When the VS application has extreme importance, the network QoS improves due to the impact of application significance on the network performance. Figure 5.8 shows the comparisons of network performance for few other scenarios with changing application significances. The network shows a better QoS level when the VS application has extreme importance because the VS application shows better performance than the voice application.

![Network Performance Analysis](image)

**Figure 5.9 UMTS-WiMAX-based Network Performance Analysis with different Application Significance**

Figure 5.9 shows the QoS in the UMTS-WiMAX integrated network for different number of calls when the significance of the application changes. When the VC application has extreme importance and moderate importance over voice application respectively, the network shows a poor QoS level. The reason is that the VC application with ten and twenty voice calls on the network experience a poor quality. On the other hand, the network takes an average QoS level with ten and twenty voice calls.
calls when the voice application has moderate and extreme importance over VC application respectively.

The above-discussed analyses clearly indicate that using application significance the QoS level can be adjusted to reflect the service-centric performance of any network. If a significant application in a network experiences poor QoS, the overall network is recognised to have a poor performance regardless of the performance of other applications. In this way, the user experience or QoE can be integrated into the network QoS level as well. Because, if the users experience poor performance for a significant application, the overall network performance reflects that performance with a lower QoS level. Therefore, it is easy to figure out the application for which the network is experiencing a poor performance.

5.5 QoS Analysis using Dynamic Weight-based Method

In Chapter 4, the challenges related to the performance evaluation of different networks and applications using conventional methods have been outlined. These challenges are mainly faced due to the assessment of each QoS-related parameter individually. In these methods, the values of the QoS-related parameters are usually compared with their benchmark values, and the application performance is evaluated based on the comparison results. It has been demonstrated in the QoS analysis section of Chapter 4 that a unified QoS metric makes such analysis more efficient and easy. Therefore, in this chapter, the various simulation scenarios are analysed using the dynamic weight-based QoS method directly. Recall that the scale used for the QoS analysis has been illustrated in Section 4.5.
At first, the WLAN-UMTS-WiMAX integrated network performance is evaluated. Table 5.10 shows the values for different QoS-related parameters in voice calls in the first phase of the simulations. Using this data, the performance metric of each of such parameter is calculated using equation (4.2) and the application QoS metric is derived using equation (4.3). The results are shown in Table 5.11. When six new calls are added to the network, the overall performance of the voice calls has dropped by 21%. With the addition of fourteen new calls, the performance has decreased by 68%.

**Table 5.10 Voice Application Data for the First Phase of Simulations**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>End-to-end delay (msec)</th>
<th>Jitter (msec)</th>
<th>Packet loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Six voice calls</td>
<td>150</td>
<td>20</td>
<td>1</td>
</tr>
<tr>
<td>Twelve voice calls</td>
<td>250</td>
<td>25</td>
<td>1.5</td>
</tr>
<tr>
<td>Twenty voice calls</td>
<td>400</td>
<td>60</td>
<td>3.5</td>
</tr>
</tbody>
</table>

Table 5.12 shows the values of QoS-related parameters in VC sessions. Using these data, the performance metric for each of this parameter and the application QoS metric is calculated. Table 5.13 shows these results. In the UMTS network, with the addition of six new voice calls, the performance of VC sessions drops by only 3%. In the WLAN network, the performance remains the same. However, with the
admission of fourteen new calls, the performance of VC sessions drop by 50% on the UMTS network and about 52% on the WLAN network.

### Table 5.12 Data Analysis of VC Application in the First Phase of Simulations

<table>
<thead>
<tr>
<th>Scenario</th>
<th>End-to-end delay (msec)</th>
<th>Jitter (msec)</th>
<th>Packet loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>WLAN</td>
<td>UMTS</td>
<td>WLAN</td>
</tr>
<tr>
<td>Six Voice Calls + 6 VC users</td>
<td>80</td>
<td>130</td>
<td>15</td>
</tr>
<tr>
<td>Twelve Voice calls + 6 VC users</td>
<td>100</td>
<td>250</td>
<td>20</td>
</tr>
<tr>
<td>Twenty voice calls + 6 VC users</td>
<td>300</td>
<td>320</td>
<td>35</td>
</tr>
</tbody>
</table>

### Table 5.13 QoS Metrics for VC Application in the First Phase of Simulations

<table>
<thead>
<tr>
<th>Scenarios</th>
<th>$P_{QoP_{k,a}}^R$</th>
<th>$P_{QoS_{k,a}}^R$</th>
<th>$P_{QoS_{k,a}}^R$</th>
<th>$QoSAM_{k,a}^R$</th>
<th>Overall QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>WL</td>
<td>U</td>
<td>WL</td>
<td>U</td>
<td>WL</td>
</tr>
<tr>
<td>Six voice calls + 6 VC users</td>
<td>1</td>
<td>0.9</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Twelve voice calls + 6 VC users</td>
<td>1</td>
<td>0.5</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Twenty voice calls + 6 VC users</td>
<td>0.33</td>
<td>0.27</td>
<td>0.33</td>
<td>0.56</td>
<td>0.8</td>
</tr>
</tbody>
</table>

In the next stage, the RAN QoS metrics are calculated for all these scenarios using equation (4.3). Figure 5.10 shows the comparisons. With the addition of six new calls, the performance of the whole network drops by 7% in the rural side and with the addition of fourteen new calls, the performance drops by 56%.
The performance evaluation is also conducted for the simulation scenarios of the second phase. In the second stage of the first simulation scenario, there are eight ongoing voice calls, and three VC participants on the UMTS networks. In the second scenario, the VC participants are moved to three WLAN hotspots. In the third scenario, all the VC participants are moved to one WLAN hotspot. Table 5.14 shows the result analysis of the voice application and Table 5.15 shows the QoS metrics.
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The voice calls have the best performance when there is one WLAN hotspot on the network with three VC participants, and the other three VC participants are on the UMTS network. When all the VC participants are on the UMTS network, the voice calls experience the worst performance. The voice application performance downgrades by 48%.

Table 5.16 shows the values of QoS metrics for the VC sessions of the second phase of the simulations. The calculations of the performance metrics are presented in Table 5.17. When there is one WLAN hotspot on the UMTS network, the VC sessions experience the best performance. When all the traffics are moved on the UMTS network from the WLAN, the performance degrades by 32%. Figure 5.11 shows the comparisons of the performance of the network for all these three scenarios. The performance of UMTS networks improves by 34% by moving the VC traffic to three different WLAN hotspots, and this performance is improved by 42% by moving them on one WLAN hotspot. Although adding three different WLANs on the UMTS network has improved the performance, the access points and WLAN getaways in each UMTS cell were generating some extra traffic. Therefore,

**Table 5.16 Data Analysis of VC Application in the Second Phase of Simulations**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>End-to-end delay (msec)</th>
<th>Jitter (msec)</th>
<th>Packet loss (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario 1</td>
<td>300</td>
<td>50</td>
<td>0.05</td>
</tr>
<tr>
<td>Scenario 2</td>
<td>200</td>
<td>35</td>
<td>0.04</td>
</tr>
<tr>
<td>Scenario 3</td>
<td>100</td>
<td>10</td>
<td>0.01</td>
</tr>
</tbody>
</table>

**Table 5.17 QoS Metrics for VC Application in the Second Phase of Simulations**

<table>
<thead>
<tr>
<th>Scenarios</th>
<th>$P_{QoSR_kA}$</th>
<th>$P_{QoSM_kA}$</th>
<th>$P_{QoSS_kA}$</th>
<th>$QoSAM_kA$</th>
<th>Overall QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario 1</td>
<td>0.33</td>
<td>0.56</td>
<td>1</td>
<td>0.68</td>
<td>Average</td>
</tr>
<tr>
<td>Scenario 2</td>
<td>0.67</td>
<td>0.88</td>
<td>1</td>
<td>0.906</td>
<td>Good</td>
</tr>
<tr>
<td>Scenario 3</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>Good</td>
</tr>
</tbody>
</table>
removing two WLANs and moving the VC participants to one WLAN has improved the performance of the UMTS network.

Figure 5.11 Comparisons of Network Performance for different Configurations

Figure 5.12 Performance Comparisons of Loose-coupling vs Tight-coupling Architecture
In the fifth scenario, the comparison is conducted to evaluate the performance of tight-coupling and loose-coupling architecture-based networks. In this scenario, the VC clients experience a delay of 120 msec, a jitter of 40 msec, and a packet loss of 0.02%. For voice calls, the average delay is 260 msec, jitter is 30 msec, and packet loss is 2%. Figure 5.12 shows the performance comparison between the tight-coupling and loose-coupling UMTS-WLAN interworking architecture. The tight-coupling architecture shows a slightly better performance (e.g. 9%) than the loose-coupling architecture.

![Figure 5.13  Performance Comparisons of LTE and UMTS Network](image)
Chapter 5: Dynamic Weight-based QoS Evaluation

Then, the simulation data is analysed for the LTE-UMTS-Ethernet Integration scenarios. Figure 5.13 presents the analysis. The LTE-based network shows a better performance than the UMTS-based network. The voice calls experience a 15% better performance and the VC sessions experience a 7% better performance on the LTE network compared to the UMTS network. For VS session, the performance shows around 16% improved performance. The LTE network shows an overall 13% improved performance than the UMTS network. Figure 5.14 shows this analysis.

The above performance studies for different network scenarios clearly demonstrate the efficiency of the dynamic weight-based QoS evaluation method. This method can be utilised to prepare a detail QoS management and monitoring report of heterogeneous-based networks. By comparing the values of application, access network, and network configuration QoS metrics, it is possible to represent the network and application performance improvement or degradation information in an exact percentage. For instance, when six new calls are admitted in a simulation scenario on the UMTS network, the performance of the whole network drops by 7%. On the other hand, when all the VC traffics are shifted from the UMTS network to the WLAN hotspots, the performance of the overall network is improved by 42%. Hence, this QoS evaluation method is very efficient in studying the network performance supporting different traffic strategies.
5.6 Comparison with Other Methods

In this section, the efficiency and effectiveness of the dynamic weight-based method are compared with the other methods. The comparison results with the fixed weight-based method show that the QoS in the application level does not vary much. Because in both of these methods, the significance of the QoS-related parameters is almost the same. On the other hand, the QoS in the access network level shows significant variations as the dynamic weight-based QoS evaluation considers context-based application weights. Table 5.18 outlines the application QoS metric calculated using the dynamic weight-based method. These data are used for the calculation of RAN QoS metric, which are explained in the later paragraphs.

**Table 5.18 QoS Analysis of Different RANs**

<table>
<thead>
<tr>
<th>RANs</th>
<th>Numbers of users</th>
<th>Application QoS Metric</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Voice</td>
</tr>
<tr>
<td>WLAN-WiMAX</td>
<td>16</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>40</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>80</td>
<td>1</td>
</tr>
<tr>
<td>UMTS-WiMAX</td>
<td>16</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>40</td>
<td>0.77</td>
</tr>
<tr>
<td></td>
<td>80</td>
<td>0.6</td>
</tr>
</tbody>
</table>
Chapter 5: Dynamic Weight-based QoS Evaluation

Figure 5.15 shows the comparison of RAN QoS metrics, which are calculated using the fixed and dynamic weight-based method. The fixed weight-based method assigns an equal weight to all the applications regardless of their importance in the network. On the other hand, the dynamic method uses a range of criteria to assign weights to the applications. In this case, the number of users in each application and the purpose of usage are used as criteria to determine the weights of the applications. Therefore, the RAN QoS metric values reflect the application significance and assess the QoS level of the network more precisely. For instance, when there are 40 users in the network, the QoS metric for the WLAN-WiMAX access network is 0.71 and 0.82 by applying dynamic weight-based, and fixed weight-based method respectively. The reason behind this is the VC application gets an importance weight of 0.8 in dynamic weight-based as it carries more significance for the users in that scenario. Therefore, while using this method, the network performance drops by 11% compared to the fixed weight-based method.

Figure 5.16 shows the detail QoS analysis of the same simulation scenario. The study
Chapter 5: Dynamic Weight-based QoS Evaluation

demonstrates that when there are 80 users in the network, the overall network performance degrades by 13.5%, while using the dynamic weight-based method compared to the fixed weight-based method. Because with the fixed weight-based method, each RAN adopts an equal weight, depending on the number of RAN on a particular network, while using the dynamic weight-based method, a weight is assigned to each RAN, based on the number of users in that specific RAN. As a result, in this scenario, the UMTS network is assigned a higher weight than the WLAN network due to having more users. Therefore, due to the poor performance of the UMTS network, the QoS of the whole network drops significantly.

![Figure 5.16: Detail Comparison of Fixed and Dynamic Weight-based Method](image)

The performance of this method is also compared to the E-model of ITU. The E-model of ITU is usually used for assessing the transmission quality of voice applications. The model considers various factors in its assessment process. The delay and packet loss are the two primary parameters, which E-model uses to assess the transmission quality factor \( R \) and user satisfaction factor MOS. The E-model does not consider jitter as a parameter for quality evaluation. To have an overall comparison of the E-model vs. the dynamic weight-based method, some of the simulation data from voice calls are assessed using both of these methods. The
advantage of adding jitter as an assessment criterion for voice quality evaluation is
analysed.

Table 5.19 shows the result analysis from voice calls on the UMTS network. It is
apparent that as E-model ignores jitter to evaluate the quality of transmission, the
assessment result can demonstrate a higher level than the actual quality. For
example, the evaluation result of 10 ongoing voice calls in the UMTS network
shows the transmission quality as “High”. On the other hand, with the considera-
tion of jitter in the DWQEM method, the voice quality is assessed as “Average”. The
reason behind this is for ten voice calls, even though, packet loss and delay is low,
jitter is comparatively high. The assessment results of the other scenarios also
demonstrate similar types of results.

**Table 5.19 QoS Metrics for Mixed Traffic**

<table>
<thead>
<tr>
<th>Number of active calls</th>
<th>E2ED (msec)</th>
<th>PL (%)</th>
<th>J (msec)</th>
<th>E-model</th>
<th>DWQEM</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>R Factor</td>
<td>Speech Transmission Quality</td>
</tr>
<tr>
<td>8</td>
<td>216</td>
<td>2.92</td>
<td>40</td>
<td>83.8</td>
<td>High</td>
</tr>
<tr>
<td>10</td>
<td>221</td>
<td>2.95</td>
<td>58</td>
<td>83.1</td>
<td>High</td>
</tr>
<tr>
<td>12</td>
<td>224</td>
<td>3.28</td>
<td>65</td>
<td>82.7</td>
<td>High</td>
</tr>
<tr>
<td>14</td>
<td>228</td>
<td>4.41</td>
<td>68</td>
<td>82.2</td>
<td>High</td>
</tr>
<tr>
<td>16</td>
<td>230</td>
<td>5.34</td>
<td>70</td>
<td>81.9</td>
<td>High</td>
</tr>
<tr>
<td>18</td>
<td>233</td>
<td>6.46</td>
<td>99</td>
<td>81.5</td>
<td>High</td>
</tr>
</tbody>
</table>

**5.7 Summary**

In this chapter, a dynamic weight-based QoS analysis method has been proposed and
evaluated. The consideration of application significance in assessing the QoS level of
heterogeneous network is the core idea of this method. The analysis of the simulation
studies demonstrates that if application significance is considered for QoS
evaluation, the unified QoS metrics for access network and network configuration
can reflect the network QoS level in a more realistic fashion. For instance, the QoS
analysis of the UMTS-WiMAX integrated network with ten and twenty voice calls
and one VC session demonstrates some interesting findings. When the VC
application is simulated having an extreme and a moderate importance over voice
application, the network shows a “Poor” QoS level. On the other hand, for the same scenario when the significance is altered with the voice application, the network takes an “Average” QoS level. Because, in those scenarios the VC application experiences a poor quality, however, voice calls experience an average quality. Therefore, the performance of the whole network degrades due to an important application having poor QoS. When the importance is altered with the voice application, the network performance improves as, a less significant application experience poor performance. Therefore, it has less effect on the overall network performance. The simulation results also demonstrate that using the dynamic weight-based method, it is possible to calculate the exact performance improvement of a network with the changing dynamics of heterogeneous networks. For instance, the comparison results of tight-coupling vs. loose-coupling-based networks indicate that the tight-coupling architecture experiences a 9% better performance than the latter one under given configurations. The comparison with the E-model shows that as the dynamic method uses jitter as one of its QoS assessment parameters, the quality assessment of voice application is more precise.

However, this method has some limitations in relation to QoS level assessment. For example, the QoS level of an application or a RAN can overlap between two consecutive measurement metrics. For example, if any, application or RAN has a QoS level between 0.6-0.8, that application or RAN is assessed as having “Medium” level QoS. If the application or RAN has a QoS level between 0.8-1, then, the QoS level is regarded as “Good”. Now, there are situations where an application or RAN has a QoS metric value of 0.79 or 0.8. Using the fixed and dynamic weight-based method, for a fraction of the difference in the QoS level, the application, or RAN QoS level can suddenly fall into a much lower category. To overcome these limitations, in the next chapter a QoS evaluation method based on fuzzy logic concepts is proposed, which can handle these uncertain situations.
The analysis in Chapter 3 has manifested the dynamic nature of the QoS-related parameters. The significance of these parameters can change depending on the user and network circumstances. Similarly, the importance of applications in each network can also vary according to the service requirements. For instance, the acceptable values of QoS-related parameters may differ from a developing country to an industrialised one. These values can also vary between a multimedia and non-multimedia network. The significance of a voice-based application can change according to its usage purposes, such as for education or health service. Any QoS evaluation method should be able to accommodate such changing conditions of the heterogeneous network environments. In this relation, the solutions developed using fuzzy logic can be easily modified by simply adjusting the inference rules and the membership functions.

The previous chapter has outlined dynamic weight-based QoS analysis that uses context-based criteria to evaluate the network performance. In some situations, the QoS analysis using this method is hard which is demonstrated by some of the simulation studies. Because there is a certain degree of uncertainty involves in this matter. To deal with such changing dynamics, in this chapter, the QoS evaluation is improved through utilising fuzzy logic concepts. A fuzzy logic-based QoS evaluation model is developed using MATLAB tool. The application of fuzzy logic concepts makes the entire methodical approach more sophisticated. The rest of the chapter is arranged as follows. Section 6.1 outlines the introductory concepts of this chapter and Section 6.2 describes the main idea behind the proposed method. Section 6.3, 6.4, 6.5, and 6.6 illustrate the subsystems of the designed model. Then, Section 6.7 presents the defuzzification method for this model and Section 6.8 discusses some case studies for performance evaluation of this method. Section 6.8 presents a comprehensive comparison with other methods. Finally, Section 6.9 summarises the outcomes of this chapter.
6.1 Introduction

The previous two chapters have proposed fixed and dynamic weight-based methods. However, those two methods are not comprehensive in a sense that they are not able to handle all the network dynamics of a complex network. In order to implement a more comprehensive QoS evaluation method in this chapter a fuzzy logic-based approach is adopted. The QoS analysis applying the dynamic and fixed weight-based method show that there could be situations where the QoS level of an application or access network overlaps between two consecutive measurement metrics. For example, if any, application or access network has a QoS level value between 0.6-0.8, the application or the RAN is regarded as having medium level QoS. If the application or RAN has a QoS level value between 0.8-1, then, the QoS level is considered as good. Now, there are cases where an application or RAN could have a QoS level value of 0.79 or 0.8. In such overlapping cases, using the fixed and dynamic weight-based methods, it is hard to determine the QoS of that application or RAN precisely. For instance, it is hard to classify the QoS level of that application or RAN as “Good” or “Medium”. Therefore, the evaluation process of any network can involve a certain degree of uncertainty. The QoS level, sometimes is not just a crisp value due to the multiple engaging factors. Fuzzy logic is suitable for dealing with this sort of uncertain and imprecise parameters.

As fuzzy logic-based solutions use linguistic variables and inference rules, the solutions are closer to the way people think. As a result, these solutions can lead to more rational and dynamic outcomes than the conventional algorithms. Using fuzzy logic, the accumulated operator, and user knowledge about the QoS evaluation can be easily accommodated. Another advantage of using fuzzy logic is its simplicity. QoS evaluation of heterogeneous network itself is complex in nature. Therefore, the proposed solutions for this type of network should be simple enough and should avoid the complex mathematical analytic. The fuzzy logic-based mathematical reasoning in this regard is very simple and easy to understand. This simplicity makes fuzzy logic a perfect candidate for providing simple QoS evaluation solutions.

6.2 The Proposed Evaluation Method

In Chapter 5, FAHP weight-based method is applied to derive network configuration, RAN, and application QoS metrics. The fixed and dynamic weight-based evaluation
methods do not use any strict principle to evaluate the QoS of applications running on a network and the QoS of the network itself. These two methods mainly depend on the numerical analysis of the gathered data. To analyse the QoS in a more efficient manner, the central idea behind this method is to use a set of unified rules. The rules will help to evaluate the performance of a network or applications running on it in any situation.

In Chapter 2, the concepts related to fuzzy logic have been discussed in detail. It is viewed as a multi-valued logic, which deals with the approximate mode of reasoning rather than the precise mode. It can be described as computing with words as well. It maps the imprecise values or the expression of words to a crisp number and provides the options to use natural language for rule classification. As a result, a sophisticated algorithm of a mathematical model could be easily explained and understandable by non-experts. A fuzzy inference system or fuzzy-rules-based system has five main functional blocks; these are: a rule base, which contains the if-then rules, the input membership functions, a fuzzification interface to transform the crisps input parameters to match linguistic variables, and a defuzzification interface, which converts the fuzzy inputs into a crisp number. The process is as follows:

- The first step involves the formulation of the problem and definition of the input and the output variables. The universe of discourse and the linguistic terms for each variable are also defined. For example, a fuzzy set \( A \) in the universe of discourse \( U \).
- Design the structure of the system, for instance, when there are multiple subsystems present, the input, and the output variables should be defined clearly. The input variable of one subsystem could be the output variable of another subsystem.
- Define fuzzy membership functions for each variable. For example, a linguistic variable \( x \) in the universe of discourse \( U \) is defined by \( T(x) = \{T_x^1, T_x^2, ..., T_x^l\} \) and \( \mu(x) = \{\mu_x^1, \mu_x^2, ..., \mu_x^l\} \), where \( T(x) \) is a term set of \( x \).
- Define the rules for the system. If there are multiple subsystems, then rules are set for each of this subsystem blocks.
- Perform defuzzification to derive a crisp value from the fuzzy values.
A hierarchical fuzzy logic system is used as there are several subsystems involve in this method. The application QoS evaluation subsystems derive the application QoS metrics; the RAN QoS evaluation subsystems derive the RAN QoS metrics, and the network configuration subsystem derives the configuration QoS metric. Figure 6.1 shows this model. The first level has the application QoS evaluation subsystem and under this, there are three modules. They are the voice, VC, and VS application-based modules. The input parameters for the first level of subsystems are delay, jitter, and packet loss. The second tier profiles the applications depending on their significance and the evaluated QoS outcome. The third level has several RAN QoS evaluation subsystems. The input parameters for these subsystems are the output parameters from the previous subsystems. The fourth level is the subsystem for network configuration QoS evaluation. The input for this subsystem is the output from the previous subsystem, which is the RAN QoS metric. The output of the network configuration subsystem is the network configuration QoS metric. MATLAB tool is used to design this conceptual model.
6.3 Fuzzy Modelling of Application QoS Evaluation Subsystem

There are three application-based modules in the first level of the fuzzy system. These are the voice, VC and VS modules. The specifications of these modules are described as follows.
6.3.1 Variables

The following tasks are related to variables:

- Define the input and the output variables
- Define the universe of discourse for each variable
- Choose the number of linguistic terms that states each variable.

Most of the fuzzy logic-based systems use three, five or seven terms for each linguistic variable. Fewer than three terms are rarely used as most concepts in human language consider at least two extremes and the middle ground. On the other hand, one rarely uses more than seven terms because humans interpret technical figures using their short-term memory. In general, human short-term memory can only compute up to seven symbols at a time [168]. Therefore, the odd cardinality of the linguistic set is used. According to the discussions in Chapter 3, the key parameters to evaluate QoS of any network-based real-time application are delay, jitter, and packet loss. Therefore, these QoS-related parameters are used as input variables of the application QoS evaluation subsystems. The justifications for choosing these variables are as follows:

- **Delay**: Delay is one of the important parameters for measuring the performance of real-time applications such as voice and video conferencing. It impacts the timeliness of the delivery of information and maintains the quality of service.
- **Jitter**: Value of jitter is one of the best indicators for observing service performance, especially real-time or delay-sensitive applications. Jitter can be used to model the packet loss of a network.
- **Packet Loss**: Packet loss is another important parameter for evaluation of network QoS. Packet-based metric is quite promising for evaluation of quality. It also contributes to the calculation of Peak Signal to Noise Ratio (PSNR).

The variables that are used in the application-based subsystems are as follows:

- $QP_{k,A}$: Input QoS-related parameters in an application $A$, where $k$ is the index for QoS-related parameters and $k=\{1, 2, \ldots, p\}$.
- $QoSAM_{A}$: QoS Metric for any application $A$ where $A= \text{Voice, VC, etc.}$
- $U$: Universe of Discourse for the QoS-related input variables
- $V_{A}$: Universe of Discourse for the application QoS metric
In the previous chapter, the dynamic weight-based QoS evaluation method uses an acceptable range for QoS-related parameters based on different contexts. In the proposed fuzzy logic-based approach, these ranges are used as the universe of discourse for each of the QoS-related input parameters. Table 6.1 to 6.3 show the universe of discourse for the QoS-related parameters of each application. The universe of discourse for each input variable is expressed as:

\[ U = [U_{\text{min}}, U_{\text{max}}] \]  \hspace{1cm} (6.1)

The universe of discourse for the output variable application QoS metric \( QoSAM_A \) is set as \([0, 1]\). It is expressed as:

\[ V^A = [V^A_{\text{min}}, V^A_{\text{max}}] \]  \hspace{1cm} (6.2)

**Table 6.1 Universe of Discourse for VC Application**

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Universe of Discourse</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-to-end delay (sec)</td>
<td>0-400</td>
</tr>
<tr>
<td>Jitter (sec)</td>
<td>0-75</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>0-5</td>
</tr>
</tbody>
</table>

**Table 6.2 Universe of Discourse for Voice Application**

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Universe of Discourse</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-to-end delay (sec)</td>
<td>0-400</td>
</tr>
<tr>
<td>Jitter (sec)</td>
<td>0-75</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>0-12</td>
</tr>
</tbody>
</table>

**Table 6.3 Universe of Discourse for VS Application**

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Universe of Discourse</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-to-end delay (sec)</td>
<td>0-10</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>0-5</td>
</tr>
</tbody>
</table>
6.3.2 Membership Functions

The ranges of the universe of discourse, which are defined in Section 6.3.1, are used to determine the fuzzy membership functions for the application QoS evaluation subsystems. Each input variable has three different membership functions: Low (L), Medium (M), and High (H). The membership functions are designed based on the empirical and simulation analysis results illustrated in Chapter 3. The design follows the conditions that each membership function only overlaps with the closest neighbouring membership function and for any possible input data, its membership values in all relevant fuzzy sets should sum to one or close to one [169]. Each membership function is expressed through a set of linguistic terms. The input variables associated with the membership functions are also termed as the linguistic variable.

Recall from the equation (2.1) that if $X$ is a collection of objects denoted graphically by $x$, then a fuzzy set $A$ in $X$ is a set of ordered pairs:

$$A = \{(x, \mu_A(x) | x \in X)\}$$  \hspace{1cm} (6.3)

Using this concept, the fuzzy set of the QoS-related parameters in the universe of discourse $U$ is defined as:

$$Q = \{QP_{k,A}, \mu_Q(QP_{k,A}) | QP_{k,A} \in U\}$$  \hspace{1cm} (6.4)

The term set for the QoS-related parameter is defined as:

$$T(QP_{k,A}) = \{T^1_{QP_{k,A}}, T^2_{QP_{k,A}}, T^3_{QP_{k,A}}\}$$  \hspace{1cm} (6.5)

where $T^1_{QP_{k,A}} = Low$, $T^2_{QP_{k,A}} = Medium$ and $T^3_{QP_{k,A}} = High$.

For example, the fuzzy set for end-to-end delay in a voice application is expressed as:

$$Q = \{QP_{d,V}, \mu_Q(QP_{d,V}) | QP_{d,V} \in U\}$$  \hspace{1cm} where $d$ denotes end-to-end delay, and $V$ denotes voice application.

The term set for the end-to-end delay for voice application is expressed as:

$$T(QP_{d,V}) = \{Low, Medium, High\}$$
The fuzzy set for the application QoS metric \( QoSAM_A \) in the universe of discourse \( V^A \) is defined as:

\[
A = \{QoSAM_A, \mu_A(QoSAM_A)|QoSAM_A \in V^A\}
\]  

(6.6)

The term set for the application QoS metric \( QoSAM_A \) is defined as:

\[
T(QoSAM_A) = \{T^1_{QoSAM_A}, T^2_{QoSAM_A}, T^3_{QoSAM_A}\}
\]  

(6.7)

where \( T^1_{QoSAM_A} = \text{Poor} \), \( T^2_{QoSAM_A} = \text{Average} \) and \( T^3_{QoSAM_A} = \text{Good} \).

For example, the term set for the voice application QoS metric is as follows:

\[
T(QoSAM_v) = \{\text{Poor}, \text{Average}, \text{Good}\}
\]

Table 6.4 shows the ranges of these membership functions.

**Table 6.4 Membership Functions for Application QoS Metric**

<table>
<thead>
<tr>
<th>Membership Function</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>0.0-0.6</td>
</tr>
<tr>
<td>Average</td>
<td>0.6-0.8</td>
</tr>
<tr>
<td>Good</td>
<td>0.8-1.0</td>
</tr>
</tbody>
</table>

The justifications for choosing the ranges for the membership functions of the QoS-related parameters are described as follows:

*Voice Application (V)*: For voice applications, three input variables are considered; these are delay, jitter, and packet loss. Table 6.5 shows the ranges of the membership functions for these variables. For each of these variables, three membership functions are defined. The justifications for choosing these ranges are as follows:

*End-to-end delay (E2ED)*: For this metric, the membership functions which are designed for the rural area have the ranges as L: 0-320, M: 300-400 and H: 380-600. The low for a rural area is chosen based on the simulation results in Chapter 3. The simulation results show that with an average delay of around 300 msec, most of the calls achieve an expected QoS level. The recommendations from the ITU and 3GPP also confirm the same ranges. The recommendations state that an end-to-end delay of
around 400 msec is considered to be acceptable considering the constraints in the current technology. The medium and high fuzzy sets are derived based on [169] using an overlap of 20.

### Table 6.5 Membership Functions for Voice Application

<table>
<thead>
<tr>
<th>Metrics</th>
<th></th>
<th>Membership Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Low</td>
<td>Medium</td>
</tr>
<tr>
<td>End-to-end delay (msec)</td>
<td>Rural</td>
<td>0-320</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-150</td>
</tr>
<tr>
<td>Jitter (msec)</td>
<td>Rural</td>
<td>0-50</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-30</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>Rural</td>
<td>0-5</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-3</td>
</tr>
</tbody>
</table>

The membership functions of the urban area are designed based on the recommendations from industrial environments such as Cisco. Therefore, the low range is set to 0-150 msec. Cisco recommends that for enterprise networks, delay should be no more than 150 msec to ensure quality voice calls. For high-quality speech, this should be no more than 100 msec.

**Jitter (J):** The values of membership functions of jitter are also defined using the same method. In the rural area, L: 0-50, M: 45-75 and H: 70-100. For instance, Cisco requires the jitter for an audio call to be less than 30 msec and the requirement for jitter stated in ITU-T REC. Y.541 is less than 50 msec. The limit for low is defined according to these recommendations, the simulation results, and other relevant studies stated in Chapter 3. The medium and high fuzzy sets are derived using an overlap of five. In the urban area, low is set to 0-30 msec. This is based on the recommendation from Cisco that jitter should be less than 30 msec for quality voice calls.

**Packet Loss (PL):** The fuzzy sets of packet loss of the rural area are defined according to the simulation results studied in Chapter 3, the recommendations of ITU, Cisco, 3GPP, and a study that has gathered data from the African Continent. All these sources have been outlined in Chapter 3. For rural areas, L: 0-5, M: 4-9, H: 8-20 and for the urban area this is L: 0-3, M: 2-5 and H: 4-20.

**Video conferencing (VC):** For VC, three input variables are considered, which are delay, jitter, and packet loss. For each of these variables, three membership functions
are defined as well. The ranges are outlined in Table 6.6. The justifications for choosing these ranges are as follows:

*End-to-end delay (E2ED):* The ranges of membership functions for VC application in a rural area network are defined based on the simulation results in Chapter 3 and the recommendations from the ITU and 3GPP. The ranges of the urban area are defined mainly based on the recommendations for enterprise networks such as Cisco recommendations.

*Jitter (J):* The ranges of jitter for VC also use the same recommendations as voice applications. The ranges of membership functions for the rural area are determined according to the experimental simulation results in Chapter 3, recommendations from the ITU, and 3GPP. The ranges of the urban area are defined based on the recommendations from Cisco.

*Packet Loss (PL):* The fuzzy sets for packet loss is defined for the rural area based on the experimental results in Chapter 3, the recommendations from the ITU, Cisco, 3GPP, and PingER research that has gathered data from the African Continent. For rural areas, L: 0-2.5, M: 1-5, H: 4-12 and for the urban area this is L: 0-2, M: 1-2.5 and H: 2-12. The experimental results from PingER show that for a VC session, packet loss below 1% is regarded as the most satisfactory and 1%-2.5% of loss is regarded as acceptable. However, if the loss is above 2.5% and the sessions experience a loss of 2.5%-5%, then, the VC quality is poor. If this loss is between 5%-12%, then the quality is very poor, and the loss above 12% is unacceptable. The

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Membership Functions</th>
<th>Low</th>
<th>Medium</th>
<th>High</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-to-end delay (msec)</td>
<td>Rural</td>
<td>0-200</td>
<td>180-400</td>
<td>380-600</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-100</td>
<td>50-200</td>
<td>180-600</td>
</tr>
<tr>
<td>Jitter (msec)</td>
<td>Rural</td>
<td>0-50</td>
<td>45-75</td>
<td>70-100</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-30</td>
<td>25-50</td>
<td>45-75</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>Rural</td>
<td>0-2.5</td>
<td>1.5-5</td>
<td>4-12</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
<td>0-2</td>
<td>1-3</td>
<td>2-12</td>
</tr>
</tbody>
</table>
results also indicate that if the loss is between 4%-6%, then it becomes impossible for non-native speakers to communicate.

**Video streaming (VS):** Delay and packet loss are considered as the input variables for the VS application subsystem. Jitter is not seen as an essential parameter for VS application performance evaluation. For each of these two variables, three membership functions are defined. The justifications for the ranges of these membership functions are as follows:

**End-to-end delay (E2ED):** For the VS application, ITU and Cisco recommend different end-to-end delay. The experimental results in Chapter 3 also show the effect of different environments on packet loss of VS sessions. For example, when the VS server is in an urban outdoor environment and the client is in the rural outdoor environment, the VS client experiences a 314.5 msec delay and 5.72% packet loss. On the other hand, while the server is placed in a suburban environment, the same users experience 6.21% packet loss and the delay almost remains the same. Based on these observations and the available recommendations, the ranges for the end-to-end delay are defined.

**Packet Loss (PL):** The membership functions for packet loss in the context of rural and urban area are also defined according to the same recommendations and simulation results in Chapter 3. Table 6.7 shows the values of these membership

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Membership Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Low</td>
</tr>
<tr>
<td>End-to-end delay (sec)</td>
<td>Rural</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
</tr>
<tr>
<td>Packet Loss (%)</td>
<td>Rural</td>
</tr>
<tr>
<td></td>
<td>Urban</td>
</tr>
</tbody>
</table>
functions.

The proposed QoS evaluation method applies Gaussian membership functions for several reasons. Primarily, these membership functions are flexible in nature. They can be easily modified by simply adjusting the mean and the variance of the membership functions. Secondly, the boundless increase of the number of
statistically independent samples takes the probability distribution of the sample
mean to a Gaussian shape. This is stated in the central limit theorem. Since many real
dlife random phenomena are a sum of a large number of independent fluctuations; it
can be expected that a Gaussian process will perform well. Thirdly, the single
sigmoid function does not represent a closed class interval. Finally, the triangular
function is unable to ensure that all inputs are fuzzified in some classes [170]. Figure
6.2 to 6.4 show the Gaussian membership functions for voice application, which are
designed based on the limits of rural areas. Figure 6.5 shows the membership
function for application QoS metric.

6.3.3 Rules

The definition of fuzzy rules is followed by the definition of the input and output
membership functions of each subsystem. The fuzzy rules are in the form of if-then
statements. These statements look at the system inputs and determine the desired
output. If there are \( R \) rules, each with \( l \) premises in the system, the \( r \)th rule has the
following form:

\[
R_r: \text{If } QP_{k, A} \text{ is } Q^i_k \text{ then } QoSAM_{r, A} \text{ is } A^i_r
\]

Mamdani fuzzy implication method is used in this model for its simplicity. Recall
from the equation (2.13) that for two fuzzy sets \( A \) and \( B \) the Mamdani implication
can be expressed as:

\[
\mu_{R_{out}} (x, y) = \min \left[ \mu_A (x), \mu_B (y) \right]
\]

Or it can be expressed as:

\[
\Phi \left[ \mu_A (x), \mu_B (y) \right] = \mu_A (x) \cap
\]

The antecedent of the previously discussed general rule is interpreted as a fuzzy
relationship, which is obtained from the intersection of the fuzzy sets \( Q^i_k \) and \( A^i_r \):

\[
FR_r = Q^i_k \cap \]

The membership function for this relationship is defined as:

\[
FR_r (QP_{k, A}, QoSAM_A) = Q^i_k (QP_{k, A}) \cap A^i_r (QoSAM_A)
\]

Using Mamdani method the generalized rule can be denoted as:
Using the \( \max \) operator, the aggregation is denoted as:

\[
Ag(z) = \bigcup_{q \in \mathcal{P}} \left( \bigcap_{i=1}^{n} q_i \right)
\]  

(6.13)

The VC and voice application-based subsystem has \( 3^3 \) rules depending on the number of input variables and membership functions. For VS application-based subsystem, the numbers of rules are \( 3^2 \). The rules are designed according to the behaviour analysis of the considered parameters. These studies are conducted using various experimental data available in the literature.

**Voice Application (V):** For voice applications, the rules are designed based on the performance analysis of voice application in different conditions. The experimental results from E-model and other studies are considered to define the rules. For example, the first rule is:

**If Voice Delay is Low, Voice Jitter is Low and Voice Packet Loss is Low then Voice Application QoS is Good.**

Now how it is defined that if all these three parameters are low, then the application QoS for voice is usually good? The behavioural analyses of delay, jitter, and packet loss in the voice application show that the voice application has a higher level of QoS if these parameters have lower value. The ranges that regarded as low for these parameters have been discussed in Chapter 3. These ranges are determined based on the various experimental results, which are rendered with numerical values.

The E-model has been discussed in detail in Chapter 2. It is a well-known model established by ITU, which estimates the communication quality and the user satisfaction. To evaluate the user satisfaction and communication quality the model uses several parameters; these are basic signal to noise ratio, delay impairment factor, equipment impairment factor and an advantage factor. The equipment impairment includes the influence of packet loss. The advantage factor represents an advantage of access, which particular systems may provide, compared to conventional systems. The primary output of the E-model is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other
quality measures such as Mean Opinion Score (MOS), Percentage Good or Better (GoB) or Percentage Poor or Worse (PoW). The relationship between R factor and other parameters have been discussed in Chapter 2.

**TABLE 6.8 R FACTOR, MEAN OPINION SCORE AND SATISFACTION**

<table>
<thead>
<tr>
<th>R</th>
<th>MOS</th>
<th>User Satisfaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>90 &lt; R &lt; 100</td>
<td>4.34 &lt;MOS &lt;4.5</td>
<td>Very satisfied</td>
</tr>
<tr>
<td>80 &lt; R &lt; 90</td>
<td>4.03 &lt;MOS &lt;4.34</td>
<td>Satisfied</td>
</tr>
<tr>
<td>70 &lt; R &lt; 80</td>
<td>3.60 &lt;MOS &lt;4.03</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>60 &lt; R &lt; 70</td>
<td>3.10 &lt;MOS &lt;3.60</td>
<td>Many users dissatisfied</td>
</tr>
<tr>
<td>50 &lt; R &lt; 60</td>
<td>2.58 &lt;MOS &lt;3.10</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>

A value between 90 and 100 for R factor leads to a higher user satisfaction. Table 6.8 shows the interpretations of R factor for different values, the interpretations of Mean Opinion Score (MOS) and the satisfaction measurement based on MOS. Some of the experimental results show that with a packet loss of 0% and delay of 185.8 msec, the R factor is 90. The E-model does not consider jitter as one of its parameters to evaluate the voice transmission quality. However, other experiments show that for larger jitter values, the user satisfaction in terms of MOS decreases. Most of the test results in the literature outline that regardless of codec and other factor, a low delay, jitter, and packet loss lead to a good QoS level.

The experimental results also show that if the packet loss is high (12%), even with a smaller delay (28.8ms) the MOS value of the network decreases dramatically. According to this analysis, the second rule is:

**If Voice Delay is Low, Voice Jitter is Low and Voice Packet Loss is High then Voice Application QoS is Poor.**

Table 6.9 outlines the conditions for the output QoS level of the applications running on any network. As shown in the table, a rule with poor QoS level, evaluates if any of the considered parameters within the fuzzy rule have a high value. The application that undergoes this performance level triggers prompt action from the service
providers for root cause analysis of the poor performance. This application performance can affect the whole network QoS level.

**Table 6.9 Conditions for Voice Application QoS Level**

<table>
<thead>
<tr>
<th>Fuzzy Consequent</th>
<th>Fuzzy Conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>• Any of the evaluated parameters has a high value.</td>
</tr>
</tbody>
</table>
| Average          | • All the evaluated parameters have a medium value.  
                     • At least two parameters have a medium value, and the other parameter has a value rather than high. |
| Good             | • At least one has a medium value, and the other two have low values.  
                     • All the evaluated parameters have a low value. |

A rule with average QoS level evaluates if at least two of the parameters within the fuzzy rule have a medium value, and the other parameter has a value rather than high. The rule can also have an average value if all the parameters are evaluated as having medium values. The application that undergoes this performance level can have a long-term investigation from service providers for root cause analysis of the performance.

A rule with good QoS level evaluates if all the parameters within the fuzzy rule have a lower value, or at least one parameter has a medium value while other two have low value. The application that undergoes this performance level can be a model case for service operators to assess other networks. Figure 6.6 to 6.8 shows the relationship of voice application QoS metric to E2EDelay, jitter, and packet loss. It is apparent from the graphs that the voice application QoS metric decreases when the values of these parameters increases. In diagram 6.6, if the jitter value goes around 100 and E2Edelay value is around 600, the voice application QoS metric is zero.
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Video conferencing (VC): The rules for VC applications are derived by analysing the impact of various ranges of delay, jitter, and packet loss on their performance. Real-time applications are in general delay-sensitive, but loss-tolerant and non-realtime applications are delay-tolerant but sensitive to packet loss. In Chapter 3, some data have been presented to back up these theories. For instance, based on PingER
report, for a VC session, packet loss below 1% is regarded as satisfactory, 1%-2.5% of loss are acceptable, 2.5%-5% is poor, 5%-12% are very poor, and loss above 12% are unacceptable. The observations indicate that when any VC session experiences 4%-6% of packet loss, it is hard for non-native speakers to communicate properly.

For conversational and videophone applications, ITU, Cisco, and 3GPP these organizations agree that it is preferred to have an end-to-end delay of no more than 150 msec. However, according to ITU-T, this refers to a long-term achievable value. Given the current technology, an end-to-end delay of around 400 msec is considered to be acceptable by both ITU-T and 3GPP. The ranges also vary between developed and developing countries or an industry and home environment. All these analysis have been presented in Chapter 3. The rules use the same pattern as the voice application that is outlined in Table 6.9. The first rule of the VC application subsystem is:

**If VC Delay is Low, VC Jitter is Low and VC Packet Loss is Low then VC Application QoS is Good.**

*Video Streaming (VS):* In case of VS applications two parameters are considered to form the rules, these are delay and packet loss. The reasons for considering only these two parameters have been discussed in Chapter 3. In this case, the rules take the pattern as outlined in Table 6.10.

<table>
<thead>
<tr>
<th>Fuzzy Consequent</th>
<th>Fuzzy Conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>• Any of the evaluated parameters has high value.</td>
</tr>
<tr>
<td>Average</td>
<td>• All the evaluated parameters have medium value.</td>
</tr>
<tr>
<td>Good</td>
<td>• All the evaluated parameters have low value.</td>
</tr>
</tbody>
</table>

**6.4 Fuzzy Modelling of Profiled Application QoS Subsystem**

After analysing the QoS level of the applications running on a network, the application significances are applied to assess the overall QoS of the application.
6.4.1 Variables

This subsystem takes two input parameters. One of them is the output from the previous application-based subsystems, which is the application QoS metric. The second input parameter is the significance of those applications depending on the particular context. Therefore, the input variables are:

\[ QoSAM_A : \text{Application QoS metric.} \quad S_A : \text{Application Significance.} \]

The output variable is: \[ QoSWAM_A \] (Profiled Application QoS Metric)

The other notations that are used in this subsystem are:

\[ V^A : \text{Universe of Discourse for the input variable application QoS metric} \]

\[ V^S : \text{Universe of Discourse for the input variable application significance} \]

\[ WV^A : \text{Universe of Discourse for the profiled application QoS metric} \]

The universe of discourse for the significance of the application \( S_A \) is set to \([0, 1]\).

The fuzzy set for the significance of the application \( S_A \) in the universe of discourse \( V^S \) is defined as:

\[
SG = \left\{ S_A, \mu_{SG} (S_A) \mid S_A \in V^S \right\} \quad (6.14)
\]

The term set for this variable is defined as:

\[
T(S_A) = \left\{ T_{S_A}^1, T_{S_A}^2, T_{S_A}^3 \right\} \quad (6.15)
\]

where \( T_{S_A}^1 = \text{Low}, \ T_{S_A}^2 = \text{Medium} \) and \( T_{S_A}^3 = \text{High} \).

For example, the fuzzy set for the input variable voice application significance is expressed as:

\[
SG = \left\{ S_V, \mu_{SG} (S_V) \mid S_V \in V^S \right\} \quad (6.16)
\]

The universe of discourse for the profiled application QoS metric output variable \( WV^A \) is \([0, 1]\). It is expressed as:

\[
WV^A = \left\{ WV^A_1, WV^A_2, \ldots, WV^A_l \right\} \quad (6.17)
\]
The fuzzy set for the profiled application QoS metric output variable $QoSWAM_A$ in the universe of discourse $WV^A$ is defined as:

$$WA = \left\{QoSWAM_A, \mu_{WA}(QoSWAM_A) | QoSWAM_A \in WV^A \right\}$$  \hspace{1cm} (6.18)

$$T(QoSWAM_A) = \left\{T_{QoSWAM_A}^1, T_{QoSWAM_A}^2, T_{QoSWAM_A}^3 \right\}$$  \hspace{1cm} (6.19)

where $T_{QoSWAM_A}^1 = Low$, $T_{QoSWAM_A}^2 = Medium$ and $T_{QoSWAM_A}^3 = High$.

### 6.4.2 Membership Functions

In this case, also the membership functions are formed as Gaussian functions. The reason for using Gaussian membership functions are illustrated in Section 6.3.2. The ranges of the input and output variables of the membership functions are a scale between zero and one. The details of the application QoS metric are outlined in Section 6.3.2. The input variable application significance takes a scale between zero and one. The output variable profiled application QoS metric also uses a scale between zero and one having three membership functions as good, average and poor. Table 6.11 shows the ranges for application significance metric.

<table>
<thead>
<tr>
<th>Membership Function</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low</td>
<td>0-0.6</td>
</tr>
<tr>
<td>Medium</td>
<td>0.6-0.75</td>
</tr>
<tr>
<td>High</td>
<td>0.75-1.0</td>
</tr>
</tbody>
</table>

The membership function for voice application QoS metric is presented in Figure 6.5. The membership functions for VC and VS application are described in the following figures. The membership functions for the other input variable application significance and the output variable profiled application QoS metric are also outlined in the following figures.
6.4.3 Rules

Each of the profiled application QoS evaluation subsystem has $3^2$ rules depending on the number of membership functions and input variables. The rules are designed based on the simulation result analysis in Chapter 5. In those studies, the impact of application significance on the overall network performance has been outlined. In one of the simulation result analysis, the network QoS demonstrates a good performance level with a voice application QoS level being average and VS application QoS level being good when the voice and VS application both have equal importance. When the significance of voice applications has been changed to extremely important compared to VS application, the network QoS comes down to
an average QoS level. Although, the performance of the VS application was good, because of having a lower importance, it had minimal effect on the network QoS level. On the other hand, the voice application having greater importance affected the network QoS level to a great extent. Based on such analysis results, the rules are defined for the profiled application subsystem.

**Table 6.12 Conditions for Profiled Application QoS Level**

<table>
<thead>
<tr>
<th>Fuzzy Consequent</th>
<th>Fuzzy Conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>• The application with high/medium importance has poor QoS level.</td>
</tr>
<tr>
<td>Average</td>
<td>• The application with high/medium importance has average QoS level.</td>
</tr>
<tr>
<td></td>
<td>• The application with low importance has poor QoS level.</td>
</tr>
<tr>
<td>Good</td>
<td>• The application with high/medium/low importance has good QoS level.</td>
</tr>
<tr>
<td></td>
<td>• The application with low importance has average QoS level</td>
</tr>
</tbody>
</table>

Table 6.12 illustrates the conditions for the output QoS level of the profiled application QoS evaluation subsystems. As shown in the table, a rule with poor QoS level, evaluates if any of the considered applications with a higher importance has a poor QoS level. The profiled application QoS adopts a value as “Good” if one the high important application has good QoS. The medium or low important application, in this case, gets lower priority compared to high important application. Therefore, even if these applications have medium QoS level, they affect the overall application QoS level to a limited extent. The average QoS level is also determined based on similar criteria. The example of one of the rules from the profiled application QoS evaluation subsystem is:

If Voice Application Significance is Low and the Voice Application QoS is Poor then the Profiled Application QoS is Average.

If there are $R$ rules, each with $l$ premises in the system, the $r$th rule has the following form:
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\( Ru_x : \text{If } S_A \text{ is } S_G^i \text{ and } QoSAM_A \text{ is } A'_i \text{ then } QoSWAM_A \text{ is } W_A^i \)

The antecedent of this general rule can be interpreted as a fuzzy relationship which is obtained through the intersection of the fuzzy sets \( S_G^i, A'_i \) and \( W_A^i \):

\[
FR_i = S_G^i \cap A'_i \cap W_A^i
\]  

(6.20)

The membership function for this relationship is defined as:

\[
FR_i(S_A, QoSAM_A, QoSWAM_A) = S_G^i(S_A) \cap A'_i(QoSAM_A) \cap W_A^i(QoSWAM_A)
\]  

(6.21)

Using the max operator, the aggregation is denoted as:

\[
Ag(z) = \bigcup_{q=1}^{P} \cap \cap \cap 1_A)
\]  

(6.22)

6.5 Fuzzy Modelling of Radio Access Network QoS Subsystem

In this Section, the fuzzy modelling of radio access network QoS evaluation subsystem is outlined. One network can have multiple radio access networks. As a result, the QoS level of each RAN is assessed separately. The number of RAN subsystem is defined based on the number of RANs present in the network. The specifications of the RAN subsystem are described as follows.

6.5.1 Variables

The input parameter for this subsystem is the output parameter from the previous subsystems which is the profiled application QoS metric.

\( QoSWAM_A^R \): Profiled application QoS metric

\( QoSAM_R \): RAN QoS metric

\( W^R \): Universe of Discourse for the RAN QoS Metric

The universe of discourse for RAN QoS Metric output variable \( W^R \) is \([0, 1]\). It is expressed as:

\[
W^R = \{W_1^R, W_2^R, ..., W_i^R\}
\]  

(6.23)

The fuzzy set for RAN QoS metric output variable \( QoSAM_R \) in the universe of discourse \( W^R \) is defined as:
\[ RQ = \{ QoS_{RM}^R, \mu_{RQ}(QoS_{RM}^R) | QoS_{RM}^R \in W^R \} \]  

(6.24)

The term set for this \( QoS_{RM}^R \) is defined as:

\[ T(QoS_{RM}^R) = \{ T_{QoS_{RM}^R}^1, T_{QoS_{RM}^R}^2, T_{QoS_{RM}^R}^3 \} \]  

(6.25)

where \( T_{QoS_{RM}^R}^1 = \text{Poor} \), \( T_{QoS_{RM}^R}^2 = \text{Average} \) and \( T_{QoS_{RM}^R}^3 = \text{Good} \).

### 6.5.2 Membership Functions

In this case, the membership functions are also defined as Gaussian functions. The reason for using Gaussian membership functions are illustrated in Section 6.3.2. The ranges of the input and output variables of the membership functions use a scale between zero and one. The details for the profiled application QoS metric is outlined in Section 6.4.2. The output variable RAN QoS metric adapts a scale between zero and one, having three membership functions as good, average and poor. Table 6.13 illustrates the ranges for each membership function in the RAN QoS evaluation subsystem.

**Table 6.13 Membership Functions for Network RAN QoS Metric**

<table>
<thead>
<tr>
<th>Membership Function</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>0.0-0.50</td>
</tr>
<tr>
<td>Average</td>
<td>0.50-0.75</td>
</tr>
<tr>
<td>Good</td>
<td>0.75-1.0</td>
</tr>
</tbody>
</table>

The output variable RAN QoS metric is presented in the following figures.
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6.5.3 Rules
The number of rules for each RAN QoS evaluation subsystem depends on the number of membership functions and input variables. For example, if there are three applications in an access network, the number of rules for that access network subsystem is $3^3$. In Chapter 5, the simulation result analysis demonstrates that the applications with lower importance have minimal effect on the network QoS level. On the other hand, the applications, which have greater importance, affected the network QoS level to a great extent. As the application significant parameters are already embedded into the profiled QoS level, the rules defined in the RAN subsystems are straightforward.

Figure 6.13 Membership Functions for RAN QoS Metric
Table 6.14 outlines the conditions for the output QoS level of the radio access networks. As shown in the table, a rule with a poor QoS level evaluates if any of the profiled application QoS levels is poor. This rule is in synchronization with the rules of the profiled application QoS evaluation subsystem. According to the rule patterns of those subsystems, if any of the application with high or medium importance has poor QoS then the profiled application QoS for that application is evaluated as poor.

The RAN QoS level is set to good if any of the profiled application is labelled as having good QoS. According to the rule in the previous subsystem, the profiled QoS of any application is labelled as good if any of the applications, regardless of importance has good QoS level or any application with low importance has average QoS level. The low important application, in this case, gets lower priority as they do not affect the overall network QoS level that much. The example of one of the rules from the RAN QoS evaluation subsystem is:

**If in network R the profiled voice Application QoS is Poor and if the profiled VC Application QoS is Good and the profiled VC Application QoS is Poor Then the RAN QoS Metric is Poor.**

If there are $R$ rules, each with $l$ premises in the system, the $r$th rule has the following form:

---

<table>
<thead>
<tr>
<th>Fuzzy Consequent</th>
<th>Fuzzy Conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>• The profiled QoS level of any application is poor.</td>
</tr>
</tbody>
</table>
| Average          | • All or the maximum numbers or the equal numbers of profiled application QoS are average.  
                   | • No profiled application QoS is poor. |
| Good             | • All or the maximum numbers of profiled application QoS levels are good.  
                   | • No profiled application QoS is poor. |
Chapter 6: QoS Evaluation through Fuzzy Logic

\[ Ru_q : \text{If } QoSWAM_i \text{ is } W_A^i \text{ then } QoSRM_R \text{ is } RQ^l_i \]

where \( i \) denotes the number of applications and \( i=\{1,2,\ldots,m\} \). The antecedent of this general rule can be interpreted as a fuzzy relationship, which is obtained from the intersection of the fuzzy sets, \( W_A^i \) and \( RQ^l_i : \)

\[ FR_i = W_A^i \cap RQ^l_i \]  \hspace{1cm} (6.26)

The membership function for this relationship is defined as:

\[ FR_i (QoSWAM_A, QoSRM_R) = W_A^i (QoSWAM_A) \wedge RQ^l_i (QoSRM_R) \]  \hspace{1cm} (6.27)

Using the \( \text{max} \) operator, the aggregation is denoted as:

\[ Ag(z) = \bigcup_{q=1}^{n} \bigcap_{q=1}^{n} \]  \hspace{1cm} (6.28)

6.6 Fuzzy Modelling of Network Configuration QoS Subsystem

This subsystem produces the final network configuration QoS metric by utilising the RAN QoS metrics.

6.6.1 Variables

The output variables from RAN subsystems are the input variable of this subsystem.

\( QoSRM^N \): Input variable RAN QoS metric

\( QoSCM^N \): Output variable network configuration QoS metric

\( X^N \): Universe of Discourse for network configuration QoS metric

The universe of discourse for configuration QoS metric \( QoSCM^N \) is \([0, 1]\). It is expressed as:

\[ X^N = \{X_1^N, X_2^N, \ldots, X_i^N\} \]  \hspace{1cm} (6.29)

The fuzzy set for \( QoSCM^N \) in the universe of discourse \( X^N \) is defined as:

\[ NC = \{QoSCM^N, \mu_{NC}(QoSCM^N) | QoSCM^N \in X^N\} \]  \hspace{1cm} (6.30)

The term set for this \( QoSCM^N \) is defined as:

\[ T(QoSCM^N) = \{T_1^{QoSCM^N}, T_2^{QoSCM^N}, T_3^{QoSCM^N}\} \]  \hspace{1cm} (6.31)
where $T^1_{QoSCM_x} = \text{Poor}$, $T^2_{QoSCM_x} = \text{Average}$ and $T^3_{QoSCM_x} = \text{Good}$.

### 6.6.2 Membership Functions

The following figure shows the membership function for the network configuration QoS metric. Gaussian membership functions are used for the same reasons as stated in section 6.3.2. The output variable network configuration QoS metric takes a scale between 0 and 1 having three membership functions as good, average and poor. Table 6.15 shows the ranges for application significance metric. Figure 6.14 illustrates the Gaussian membership functions.

<table>
<thead>
<tr>
<th>Membership Function</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>0-0.6</td>
</tr>
<tr>
<td>Average</td>
<td>0.6-0.75</td>
</tr>
<tr>
<td>Good</td>
<td>0.75-1.0</td>
</tr>
</tbody>
</table>

**Figure 6.14 Membership Function for the Network Configuration QoS Metric**
6.6.3 Rules
The number of access network in a network determines the number of rules for this subsystem. If the network has two RAN, then the number of rules is \(3^2\). The rules are designed according to the simulation result analysis from Chapter 4 and Chapter 5. In those Chapters, the network configuration QoS metric has been calculated by combining the performance metrics of different RANs in a network. The analyses from those calculations demonstrate that if all the RAN has average QoS usually the overall network QoS is average. If one of the RAN has poor QoS, then the overall network QoS is usually poor. In order to achieve a good network QoS, all the RAN QoS should be good. Based on such analysis results, the rules are defined for the RAN subsystem. Table 6.16 shows the rule patterns for this QoS evaluation subsystem.

<table>
<thead>
<tr>
<th>Fuzzy Consequent</th>
<th>Fuzzy Conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poor</td>
<td>At least one or All the RAN has poor QoS level.</td>
</tr>
<tr>
<td>Average</td>
<td>All or the maximum numbers of RAN have average QoS level. No RAN has poor QoS.</td>
</tr>
<tr>
<td>Good</td>
<td>All or the maximum numbers of RAN have good QoS level. No RAN has poor QoS level.</td>
</tr>
</tbody>
</table>

If there are \(R\) rules, each with \(l\) premises in the system, the \(r\)th rule has the following form:

**If** \(QoSRM_{R_i}^j\) **is** \(RQ_i^j\) **then** \(QoSCM_i^j\) **is** \(NC_i^j\)

where \(j\) is the number of RANs present in the network and \(j=\{1,2,\ldots, n\}\). The antecedent of this general rule can be interpreted as a fuzzy relationship, which is obtained by the intersection of the fuzzy sets \(RQ_i^j\) and \(NC_i^j\) :

\[
FR_i = RQ_i^j \cap NC_i^j
\]  

(6.32)
The membership function for this relationship is defined as:

$$ FR_i(QoS_{RM_{N}}^{R_i}, QoS_{CM_{N}}) = RQ_i(QoS_{RM_{N}}^{R_i}) \land NC_i(QoS_{CM_{N}}) $$  \hspace{1cm} (6.33)

If the subsystem has \( r \) number of rules, then using the \( \text{max} \) operator the aggregation is denoted as:

$$ Ag(z) = \bigcup \bigcap $$  \hspace{1cm} (6.34)

### 6.7 Defuzzification

The next step is the defuzzification process. Each rule produces a fuzzy output set. These sets are then mapped to a crisp value through the defuzzification method. There are many ways of defuzzification, such as the centre of gravity (CoG) or centroid method, weighted mean method and min-max method. The most commonly used strategy centroid method is used for this thesis. The centroid calculation is denoted as:

$$ QoS_{N}^{*} = \frac{\sum_{i=1}^{n} \mu_{NC_i}(QoS_{CM_{N}}) \cdot QoS_{CM_{N}} \cdot dQoS_{CM_{N}}}{\sum_{i=1}^{n} \mu_{NC_i}(QoS_{CM_{N}}) \cdot dQoS_{CM_{N}}} $$  \hspace{1cm} (6.35)

where \( n \) is the number of outputs.

### 6.8 Case Study

In this section, a case study is outlined to evaluate the efficiency of the proposed QoS evaluation method. A service operator \( Z \) maintains a network \( N \), which is used for mainly voice applications. In the near future, this network will be heavily used for education service. As a result, video conferencing and video streaming applications will be two major applications in this network along with voice applications. The operator wants to investigate the performance of the network with the new configuration settings before deploying the final network. They also want to examine the required QoS for each application and each RAN in the network. The network has UMTS and LTE radio access networks along with WLANs. The proposed method discussed previously is used to carry out the analyses.
In the first phase, the current network for Z is simulated with the actual number of voice users. The network QoS is calculated for this stage. Then, in the second phase, the network with the upcoming changes is simulated with the newly added VC and VS users. The performance is analysed for each application, access network, and the whole network configuration. The results of the first and the second phase are compared to provide recommendations for the upcoming upgrades. Table 6.17 shows the network settings in both phases. At first stage, the whole network has total 46 users and in the second phase, this number is increased to 113. The UMTS RAN has 52 users, and the LTE RAN has 51 users.

**Table 6.17 Network Settings**

<table>
<thead>
<tr>
<th>Network</th>
<th>Number of Users in Voice Application</th>
<th>Number of users in VC Application</th>
<th>Number of users in VS Application</th>
</tr>
</thead>
<tbody>
<tr>
<td>UMTS +WLAN (1st phase)</td>
<td>20</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>UMTS (2nd phase)</td>
<td>40</td>
<td>8</td>
<td>4</td>
</tr>
<tr>
<td>LTE (1st phase)</td>
<td>20</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>LTE (2nd phase)</td>
<td>40</td>
<td>6</td>
<td>5</td>
</tr>
</tbody>
</table>

**Table 6.18 QoS Analysis of Voice Application**

<table>
<thead>
<tr>
<th>RANs</th>
<th>Number of users</th>
<th>Specifications</th>
<th></th>
<th></th>
<th>QoSAM$_A$</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>End-to-end Delay (msec)</td>
<td>Jitter (msec)</td>
<td>Packet loss (%)</td>
<td></td>
</tr>
<tr>
<td>UMTS</td>
<td>20</td>
<td>240</td>
<td>45</td>
<td>6.25</td>
<td>0.836</td>
</tr>
<tr>
<td></td>
<td>40</td>
<td>330</td>
<td>55</td>
<td>8.10</td>
<td>0.677</td>
</tr>
<tr>
<td>LTE</td>
<td>20</td>
<td>120</td>
<td>25</td>
<td>5.15</td>
<td>0.877</td>
</tr>
<tr>
<td></td>
<td>40</td>
<td>200</td>
<td>45</td>
<td>6.23</td>
<td>0.837</td>
</tr>
</tbody>
</table>
Table 6.19 QoS Analysis of VC Application

<table>
<thead>
<tr>
<th>RANs</th>
<th>Number of users</th>
<th>Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>End-to-end Delay (msec)</td>
</tr>
<tr>
<td>UMTS</td>
<td>1</td>
<td>65</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>320</td>
</tr>
<tr>
<td>LTE</td>
<td>2</td>
<td>30</td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>135</td>
</tr>
</tbody>
</table>

Table 6.20 QoS Analysis of VS Application

<table>
<thead>
<tr>
<th>RANs</th>
<th>Number of users</th>
<th>Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>End-to-end Delay (msec)</td>
</tr>
<tr>
<td>UMTS</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>LTE</td>
<td>2</td>
<td>0.30</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>2</td>
</tr>
</tbody>
</table>

Table 6.18 shows the QoS analysis of voice application with the current and the forthcoming changes in the network. The performance of voice application on the UMTS network decreases by 23.3% due to the increased numbers of voice, VC and VS users. For the LTE network, this performance degradation is only 3.6%. Table 6.19 depicts the QoS analysis for VC application. With the new settings on the UMTS network, the performance of VC application degrades by around 20%. For the LTE network, this number is 12%. Table 6.20 illustrates the performance analysis results for the VS application. With three more users, the performance of VS application on the UMTS network decreases by 29.1%. For the LTE network, the performance deterioration is only 5.6%.
After analysing the data for application level, the performance is of each radio access network are evaluated. For the first phase, the importance for the VC application is set to medium, the importance for voice application is set as high, and the importance for VS application is set as low. For the second phase, the significance for VC, voice and VS applications are set as high, medium, and high respectively. The analysis shows that for the newly added applications and users, the performance of the overall network drops by 54%. This is mainly due to the performance drop of the UMTS network. The analysis is outlined in Figure 6.15.

![Figure 6.15 Performance Evaluation of Network Configuration](image)

<table>
<thead>
<tr>
<th></th>
<th>1st Phase</th>
<th>2nd Phase</th>
<th>Network QoS Metric</th>
</tr>
</thead>
<tbody>
<tr>
<td>UMTS QoS Metric</td>
<td>0.899</td>
<td>0.284</td>
<td>0.899</td>
</tr>
<tr>
<td>LTE QoS Metric</td>
<td>0.899</td>
<td>0.899</td>
<td>0.214</td>
</tr>
</tbody>
</table>

Using such analysis, it is easier for the operator Z to figure out which part of the network can face potential performance issues when the new configurations are deployed. After detecting the performance issue in the UMTS network, the operator Z makes some changes to their planned configurations. As a part of these changes, some users are moved from the UMTS to the LTE network. Among them, there are six VC users and two VS users. The performance is analysed again after these changes. Table 6.21 shows the performance evaluation with the new user arrangements.
Figure 6.16 shows the comparison with the old upgrade configurations. In the UMTS network, the voice performance improves by 9%. For VC and VS applications, the improvements are 34.9% and 28.5% respectively. In the case of LTE networks, although the performance decreases for all the applications, it is still within the acceptable level. Figure 6.17 outlines the comparisons of the overall performance of the network. With the newly planned settings, the performance of the whole network is improved by 50.9%. With a 54.7% improvement in the UMTS network and 7.7% improvement in the LTE network. Although the performance of the LTE network decreases, it is still within the acceptable QoS level.
Chapter 6: QoS Evaluation through Fuzzy Logic

The analysis of this case study demonstrates that using the proposed method it is possible to evaluate the performance of heterogeneous network efficiently. If the
operators plan to upgrade any network configurations, the potential issues with the new settings in the post-deployment stage can be figured out beforehand. It is easier to detect the specific settings that could affect the whole network performance. Based on such analysis, the configurations can be adjusted for better performance.

Using this evaluation method, the QoE level of a network can be analysed. The QoS metrics for the voice application of this case study are used to interpret the QoE, and the results are compared to the E-model of ITU. For effective comparison, the voice QoS metric values are converted to percentile to match with the QoE scale of E-model. Figure 6.18 outlines the comparisons. The QoS metric values annotated as QoE; Setting-1 refers to the previously-planned configurations, and Setting-2 refers to the newly planned configurations. The measured values from the E-model do not show much difference with the voice QoS metric values. Therefore, it is possible to measure the QoE of the voice application using the QoS metrics derived using this method.

![Comparison of QoS and QoE Value](image)

**Figure 6.18 Comparison of QoS and QoE Value**

### 6.9 Comparison with Other Methods

In this section, the comparison results of the fuzzy logic-based QoS evaluation method (FLQEM) with the fixed and the dynamic weight-based methods are presented. Table 6.22 shows the comparison results with the dynamic weight-based method. Most of the time, the networks which are evaluated using both of these methods have demonstrated the same ranking. However, it differs in the case when
the voice calls experience a delay of 224 msec, jitter of 65 msec, and packet loss of 3.28%. The dynamic weight-based method shows the QoS level as “Poor” in this situation. However, using fuzzy logic approach this is measured as “Average”. Because the fuzzy logic approach can handle an overlapping situation efficiently compared to dynamic weight-based method. With a delay of 224 msec, jitter of 65 msec and packet loss of 9.24%, the voice QoS takes a value of 0.6. Using the dynamic weight-based method, it is hard to define the QoS level of this value as the “Average”, and the “Poor” QoS values overlaps. The fuzzy logic method uses rules, therefore, firing the proper rule it calculates the QoS level as “Average”.

**Table 6.22 QoS Metrics for Mixed Traffic**

<table>
<thead>
<tr>
<th>Number of active calls</th>
<th>E2ED (msec)</th>
<th>J (msec)</th>
<th>PL (%)</th>
<th>DWQEM</th>
<th>FLQEM</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>QoS Level</td>
<td>QoS Level</td>
</tr>
<tr>
<td></td>
<td>Application QoS Metric</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>216</td>
<td>40</td>
<td>2.92</td>
<td>0.82</td>
<td>Good</td>
</tr>
<tr>
<td>10</td>
<td>221</td>
<td>58</td>
<td>2.95</td>
<td>0.62</td>
<td>Average</td>
</tr>
<tr>
<td>12</td>
<td>224</td>
<td>65</td>
<td>3.28</td>
<td>0.54</td>
<td>Poor</td>
</tr>
<tr>
<td>14</td>
<td>228</td>
<td>68</td>
<td>4.41</td>
<td>0.49</td>
<td>Poor</td>
</tr>
<tr>
<td>16</td>
<td>230</td>
<td>70</td>
<td>5.34</td>
<td>0.45</td>
<td>Poor</td>
</tr>
<tr>
<td>18</td>
<td>233</td>
<td>99</td>
<td>6.46</td>
<td>0.38</td>
<td>Poor</td>
</tr>
</tbody>
</table>

**Table 6.23 Mapping of QoE scale of E-model**

<table>
<thead>
<tr>
<th>R Factor Scale</th>
<th>QoS Metric Scale</th>
<th>QoS Level</th>
<th>QoE Level</th>
</tr>
</thead>
<tbody>
<tr>
<td>90&lt; R &lt;100</td>
<td>0.8-1</td>
<td>Good</td>
<td>Very satisfied</td>
</tr>
<tr>
<td>80&lt; R &lt;90</td>
<td></td>
<td></td>
<td>Satisfied</td>
</tr>
<tr>
<td>70&lt; R &lt;80</td>
<td>0.6-0.8</td>
<td>Average</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>60&lt; R &lt;70</td>
<td></td>
<td></td>
<td>Many users dissatisfied</td>
</tr>
<tr>
<td>50&lt; R &lt;60</td>
<td>0-0.6</td>
<td>Poor</td>
<td>Nearly all users dissatisfied</td>
</tr>
</tbody>
</table>
Table 6.23 shows the mapping of the QoE scale of E-model to the proposed QoS metric scale. It should be noted that this is only for voice application as E-model can only evaluate the voice transmission quality. It uses a five-term scale to analyse the QoE of any voice transmission quality. The proposed fuzzy logic approach uses a three-term scale to measure the QoS of voice application. This three-scale measurement is mapped to the five-scale QoE measurement scale of E-model to evaluate the QoS level in terms of QoE. Figure 6.19 shows the comparisons of QoS values of voice application on some networks using the dynamic weight-based and fuzzy logic-based method. It also shows their closeness to the measured QoE value using the E-model. The fuzzy logic approach shows closer measurement result to the measured QoE value of E-model than the DWQEM method.

6.10 Summary

In this chapter, a novel QoS evaluation method based on fuzzy logic concepts has been proposed and evaluated. The fundamental idea behind this method is the consideration of uncertain situations in performance analysis of heterogeneous
networks. There could be situations where the QoS level of an application or access network overlaps between two consecutive measurement metrics. In those cases, it is hard to assess the QoS level accurately. Through the simulation analysis, it has been demonstrated that without considering the overlapping values, the network QoS could be evaluated with a higher or lower QoS level than the expected level. The proposed method can effectively handle these situations. The method uses a set of rules for assessing the QoS of heterogeneous networks. Hence, the results can be categorised in a more efficient manner. The case study of a network planning stage shows that using the fuzzy logic-based method, the effects of any new upgrade on application and network performance can be examined in a very efficient manner.

For instance, the result analysis of this case study has outlined that the performance of voice application on the UMTS network has decreased by 23.3% and 3.6% on the LTE network due to the newly added users. After finding out such performance issues, the operator has updated the previously planned configuration. The comparison results of the newly planned settings with the old one demonstrate that, on the UMTS network, the voice performance improves by 9%. For VC and VS applications, the improvements are 34.9% and 28.5% respectively. In the case of LTE networks, although the performance decreases for all the applications, it is still within the acceptable level. The comparisons of the overall performance of the network show that with the changed settings, the performance of the whole network is improved by 50.9%. With a 54.7% improvement in the UMTS network and 7.7% improvement in the LTE network. After having such numerical analysis results, it is very easy for the service operator to go with the planned network upgrades.

The QoE level of a network has been also analysed using this method. The values of QoS-related parameters have been utilised to measure the voice QoS and QoE metrics. The voice QoS metric has been measured using the fuzzy logic-based method, and the QoE metric has been measured using the E-model. The voice QoS metric values have been converted to percentile to match with the QoE scale of E-model. The measured values from the E-model do not show much difference when compared to the voice QoS metric values. Therefore, it is possible to measure the QoE of the voice application using the QoS metrics which are derived using this method. This method is also compared with the dynamic weight-based evaluation. For instance, with a delay of 224 msec, jitter of 65 msec, and packet loss of 9.24%,
the voice QoS on the UMTS network takes a value of 0.6. Using the dynamic weight-based method, it is hard to determine the QoS level of this value as the “Average”, and the “Poor” QoS values overlaps. The fuzzy logic method uses rules, therefore, firing the proper rule it calculates the QoS level as “Average”. Therefore, the fuzzy logic-based method is more efficient in the cases when the QoS level overlaps.
CHAPTER 7

Conclusions

The transmission of multimedia-based applications over cellular and wireless technologies is on a very steep rise. Because of this growing demand, the communication networks are experiencing a massive increase in the traffic they carry. To cope up with this traffic, the service operators are deploying heterogeneous network architectures. However, due to the diverse characteristics of communication technologies and the different QoS requirements of applications utilising them, the effective QoS evaluation is still a challenging task. Various methods have been proposed in the literature to deal with these challenges. However, most of them take into account the performance of single radio access network or application. They only consider the performance of the access network in the one end of the connection and ignore the access networks on the other side of the connection. This thesis proposes a methodical approach to enhance the QoS evaluation in heterogeneous networks, which considers these different aspects.

This chapter provides a summary and conclusions from the research. The chapter also sheds light on the potential future research directions of this thesis.

The objective of this research is to come up with a methodical approach to enhance the QoS analysis, management and monitoring of heterogeneous networks. To accomplish this goal, an application-based QoS evaluation approach has been introduced that adopted the concept of unified QoS metrics. Three QoS assessment methods have been proposed under this approach, which can characterise the access networks and applications on them, with a unified QoS metric.

At first, a set of key QoS-related parameters and their benchmark ranges have been defined. In order to do that, detailed simulation studies have been carried out. These studies have helped to investigate the impact of different factors on the application performance in a network. The results from these analyses have outlined that the application performance is affected by both technological and environmental factors.
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to a certain extent. As a result, the ranges for the key performance parameters of various applications are hard to define depending on only technological factors. Therefore, the benchmark ranges for these parameters have been established according to the data from different user experience evaluation experiments, the recommendations of various organisations that work in the rural and industrial environment and the simulation results. These analyses have been described in Chapter 3.

After establishing the benchmark ranges of the key QoS-related parameters of each application, at first, a fixed weight-based QoS evaluation method has been proposed and investigated. This method assigns a fixed weight to the QoS-related parameters of each application depending on their impact level on the performance evaluation of that particular application. A fixed weight is also assigned to the applications and the active radio access networks in a heterogeneous network configuration applying an equal importance-based policy. Then, three functions have been defined to calculate a set of unified QoS metrics; these are the application QoS metric, the radio access network QoS metric or the RAN QoS metric and the network configuration QoS metric. These functions have used a hierarchical-calculation strategy to derive these metrics.

The efficiency of this method has been illustrated through a diverse range of simulation scenarios. The reported results have demonstrated that using this method, it is possible to evaluate the QoS of heterogeneous network-based service models in any given context prior to deployment. For example, the values of RAN QoS metrics calculated using the simulation results have outlined that the LTE-based networks show a 30% better performance than the UMTS-based networks when the network is comparatively congested. On the other hand, with fewer numbers of users the overall performance differs by only 12%. Such observations, which have been derived using the values of RAN QoS metric, could be used in various ways to improve the performance of heterogeneous networks. For example, the LTE networks could be recommended for a densely populated rural area and the UMTS networks for a less densely populated rural area. Or, the LTE can be chosen among a range of access networks when congestions occur. The integrated QoS metrics referred to as application, RAN, and network QoS metrics can also remove the overhead of service providers of comparing each dynamics of the network separately for QoS evaluation.
For example, the simulation results have clearly indicated that WLAN in combination with WiMAX technology is more suitable for VC services if the network needs to accommodate more users compared to UMTS or UMTS-WiMAX combinations. Other analysis shows that MPEG-4 codec can degrade the performance of a voice dominant network by 35.24% compared to H.263 codec. The end-users are also left with the options to choose the most suitable network, according to their requirements by simply looking at the access network rankings, which can be formed using these metric values. The detailed result analysis and recommendations can be found in Section 4.5 of Chapter 4.

However, this method has demonstrated some limitations in relation to weight assignment option. A fixed weight is used for all the QoS-related parameters and applications for QoS evaluation. However, these weights are subject to change according to the situational demands of the network or service requirements. One such example is varying user needs of applications from an industrial to a home environment. To handle such challenges, a dynamic weight-based calculation method would be more appropriate than the fixed weight-based method.

Therefore, as the next step of the methodical evaluation approach, a dynamic weight-based QoS evaluation method has been designed and assessed. The inclusion of application significance in the QoS assessment is the fundamental feature of this method. Through simulation analysis, it has been illustrated that without considering the application significance, in some cases, the evaluated network QoS level could be misleading. For instance, the QoS analysis of the UMTS-WiMAX integrated network with ten and twenty voice calls and one VC session demonstrates some interesting findings. When the VC application is simulated having an extreme and a moderate importance over voice application, the network shows a “Poor” QoS level. On the other hand, for the same scenario when the significance is altered with the voice application, the network takes an “Average” QoS level. Because, in those scenarios the VC sessions experience a poor quality, and the voice calls experience an average quality. Therefore, the performance of the whole network declines due to an important application having poor QoS. When the importance is altered with the voice application, the network performance improves due to a less significant application having poor performance. Therefore, it has a comparatively less effect on the overall network performance. The simulation results also demonstrate that using
the dynamic weight-based method, it is possible to calculate the exact performance improvement of a network with the changing dynamics of heterogeneous networks. For instance, the comparison results of the tight-coupling vs. loose-coupling based networks indicate that under given configurations, the tight-coupling architecture experiences a 9% better performance than the latter. The detailed analysis has been outlined in Section 5.5 of Chapter 5.

The method is also compared with the fixed weight-based method proposed in the Chapter 4 of this thesis and with the E-model implemented by ITU. The comparisons of RAN QoS metric for some networks using fixed and dynamic weight-based methods show some interesting findings. For instance, due to usage of application significance for VC application, for a certain network configuration, the performance has been dropped by 11% compared to fixed weight-based method. A detailed analysis of such findings is reported in Section 5.6 of Chapter 5. The comparison results with the E-model shows that as this model does not include jitter for QoS or transmission quality evaluation, the outcomes could be imprecise.

However, this method has some limitations. The simulation result analysis in Chapter 5 illustrates that in some cases, the application or RAN has a QoS level as 0.79. In such situations, it is hard to determine if the application or RAN would be assessed as having “Good” or the “Medium” level QoS. As a result, in Chapter 6, a novel QoS evaluation method is proposed, which integrates the concepts of fuzzy logic to handle these situations. The key feature of this method is the ability to deal with uncertainties related to network QoS evaluation. Through simulation studies, it has been demonstrated that this novel method improves the network QoS evaluation up to a certain extent. The analysis of a case study shows that it is possible to assess the performance of any network in the pre-deployment stage efficiently. For example, the result analysis of this case study has outlined that the performance of voice application on the UMTS network has decreased by 23.3% and 3.6% on the LTE network due to the newly added users. After finding out such performance issues, the operator has updated the previously planned configuration. The comparison results of the newly planned settings with the old one demonstrate that, on the UMTS network, the voice performance improves by 9%. For VC and VS applications, the improvements are 34.9% and 28.5% respectively. In the case of LTE networks, although the performance decreases for all the applications, it is still within the
acceptable level. The comparisons of the overall performance of the network show that with the changed settings, the performance of the whole network is improved by 50.9%. With a 54.7% improvement in the UMTS network and 7.7% improvement in the LTE network. After having such numerical analysis results, it is very easy for the service operator to go with the planned network upgrades. A detailed description of such analysis is presented in Section 6.8 of Chapter 6.

Using this evaluation method, it is also possible to compare the unified QoS metric values against the QoE measurement. The QoS metrics for the voice application of this case study are compared with the QoE measurement of E-model. In order to do that, the voice QoS metrics are converted to percentile to match with the QoE scale. These analyses are illustrated in Section 6.9 of Chapter 6 in detail.

The work reported in this thesis can be easily extended in a number of ways. For instance, it can be modified as an energy-efficient application-based handover management framework. The result of the extensive growth of mobile services is motivating the communication industry to shift the focus from the cost-effective to the energy-efficient solutions. Therefore, the QoS evaluation methods, the VHO and admission control algorithms of heterogeneous networks should emphasise on energy-efficiency. The most of the existing handover algorithms tend to ignore the relatively high power consumption of the Mobile Terminal (MT) resulting from the continuous scanning process. Generally, all the interfaces of an MT are always on, which causes extra power consumption. Using the application-based framework, the energy-efficiency of the classical VHO process can be improved by discarding such unnecessary scanning. Before starting a new application, the MT can ask the Handover and Resource Manager (HRM) to provide information about the available access networks. If the active network is appropriate for that application, the MT keeps attached to the active network. Otherwise, it handovers to another network with a more suitable setting. This decision can be taken based on the value of RAN QoS metric defined in the application-based evaluation approach. HRM stores the information regarding the QoS evaluations of the networks and combines them with the application level parameters. It will then store this overarching QoS metric, relating application-based performance to each network and its communication technology. HRM sends this information to BS periodically. The MT can ask for information update each time it starts a new application or initiates a handover
request for any other reason. Thus, by optimising the scanning process and transferring the responsibility of network profiling to the resource manager, this framework will be able to reduce the VHO-related additional power consumption of the MT.

The proposed approach in this thesis can also further developed by including energy-efficiency as a parameter for the performance evaluation of heterogeneous networks. To define whether a network is energy-efficient or not, the power consumption of all the entities in a network needs to be considered. A benchmark range for this parameter is required to be defined in a similar way it has been established for the QoS-related parameters. The inclusion of energy-efficiency as a network performance evaluation parameter can result in extensive improvements in the overall evaluation approach. Consequently, the potential energy-efficient VHO algorithm and QoS evaluation approach expanded from this work are worthy of future research.
References


[61] Technical specification group radio access network, 3GPP.


[63] 3GPP system to Wireless Local Area Network (WLAN) interworking; System description, Rel-12, 3GPP.

[64] Generic Access Network (GAN); Stage 2, Rel-12, 3GPP.

[65] Stage 2, 3GPP WiMAX Interworking, Release 1.

[66] Architecture Enhancements for Non-3GPP Accesses (Release 8), 3GPP.

[67] Improved Network Controlled Mobility between E-UTRAN and 3GPP2/Mobile WiMAX Radio Technologies, 3GPP.


[146] The E-model, a computational model for use in transmission planning, ITU-T.


[148] Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl, ITU-T.


[154] 3GPP TR 26.937 version 1.2.0 RTP usage model, 3GPP.


Publications arising from the work reported in the Thesis

Appendix A

This section shows some of the scripts for the calculations of the chapter 4 and 5.

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Weight Calculation with FAHP

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Appendix B

This appendix contains all the rules of the Fuzzy logic-based QoS evaluation method, referred to in chapter 6.

Rules for Application-based Subsystems (Voice)

1. If $E2E_{Delay}$ is Low and $Jitter$ is Low and Packet loss is Low then Voice Application QoS is Good
2. If $E2E_{Delay}$ is Low and $Jitter$ is Low and Packet loss is Medium then Voice Application QoS is Good
3. If $E2E_{Delay}$ is Medium and $Jitter$ is Low and Packet loss is Low then Voice Application QoS is Good
4. If $E2E_{Delay}$ is Medium and $Jitter$ is Low and Packet loss is Medium then Voice Application QoS is Average
5. If $E2E_{Delay}$ is Low and $Jitter$ is Medium and Packet loss is Low then Voice Application QoS is Average
6. If $E2E_{Delay}$ is Low and $Jitter$ is Medium and Packet loss is Medium then Voice Application QoS is Average
7. If $E2E_{Delay}$ is Low and $Jitter$ is Medium and Packet loss is High then Voice Application QoS is Poor
8. If $E2E_{Delay}$ is Medium and $Jitter$ is Low and Packet loss is Low then Voice Application QoS is Average
9. If $E2E_{Delay}$ is Medium and $Jitter$ is Low and Packet loss is Medium then Voice Application QoS is Poor
10. If $E2E_{Delay}$ is Medium and $Jitter$ is Low and Packet loss is High then Voice Application QoS is Poor
11. If $E2E_{Delay}$ is Low and $Jitter$ is High and Packet loss is Low then Voice Application QoS is Poor
12. If $E2E_{Delay}$ is Low and $Jitter$ is High and Packet loss is Medium then Voice Application QoS is Poor
13. If $E2E_{Delay}$ is Low and $Jitter$ is High and Packet loss is High then Voice Application QoS is Poor
14. If $E2E_{Delay}$ is Medium and $Jitter$ is High and Packet loss is Low then Voice Application QoS is Poor
15. If $E2E_{Delay}$ is Medium and $Jitter$ is High and Packet loss is Medium then Voice Application QoS is Poor
16. If $E2E_{Delay}$ is Medium and $Jitter$ is High and Packet loss is High then Voice Application QoS is Poor
17. If $E2E_{Delay}$ is High and $Jitter$ is Low and Packet loss is Low then Voice Application QoS is Poor
18. If $E2E_{Delay}$ is High and $Jitter$ is Low and Packet loss is Medium then Voice Application QoS is Poor
19. If $E2E_{Delay}$ is High and $Jitter$ is Low and Packet loss is High then Voice Application QoS is Poor
20. If $E2E_{Delay}$ is High and $Jitter$ is Medium and Packet loss is Low then Voice Application QoS is Poor
21. If $E2E_{Delay}$ is High and $Jitter$ is Medium and Packet loss is Medium then Voice Application QoS is Poor
22. If $E2E_{Delay}$ is High and $Jitter$ is Medium and Packet loss is High then Voice Application QoS is Poor
23. If $E2E_{Delay}$ is High and $Jitter$ is High and Packet loss is Low then Voice Application QoS is Poor
24. If $E2E_{Delay}$ is High and $Jitter$ is High and Packet loss is Medium then Voice Application QoS is Poor
25. If $E2E_{Delay}$ is High and $Jitter$ is High and Packet loss is High then Voice Application QoS is Poor

Rules for Application-based Subsystems (VS)

1. If $E2E_{Delay}$ is Low and Packet loss is Low then VS Application QoS is Good
2. If $E2E_{Delay}$ is Low and Packet loss is Medium then VS Application QoS is Average
3. If $E2E_{Delay}$ is Low and Packet loss is High then VS Application QoS is Poor
4. If $E2E_{Delay}$ is Medium and Packet loss is Low then VS Application QoS is Average
5. If $E2E_{Delay}$ is Medium and Packet loss is Medium then VS Application QoS is Average
6. If $E2E_{Delay}$ is Medium and Packet loss is High then VS Application QoS is Poor
7. If $E2E_{Delay}$ is High and Packet loss is Low then VS Application QoS is Poor
8. If $E2E_{Delay}$ is High and Packet loss is Medium then VS Application QoS is Poor
9. If $E2E_{Delay}$ is High and Packet loss is High then VS Application QoS is Poor

Rules for Profiled Application QoS Subsystem

1. If Voice Application QoS is Poor and Voice Application Importance is Low then Profiled Voice Application QoS is Average
2. If Voice Application QoS is Poor and Voice Application Importance is Medium then Profiled Voice Application QoS is Poor
3. If Voice Application QoS is Poor and Voice Application Importance is High then Profiled Voice Application QoS is Poor
4. If Voice Application QoS is Average and Voice Application Importance is Low then Profiled Voice Application QoS is Good
5. If Voice Application QoS is Average and Voice Application Importance is Medium then Profiled Voice Application QoS is Average
6. If Voice Application QoS is Average and Voice Application Importance is High then Profiled Voice Application QoS is Average
7. If Voice Application QoS is Good and Voice Application Importance is Low then Profiled Voice Application QoS is Good
8. If Voice Application QoS is Good and Voice Application Importance is Medium then Profiled Voice Application QoS is Good
9. If Voice Application QoS is Good and Voice Application Importance is High then Profiled Voice Application QoS is Good

Rules for RAN QoS Subsystem

1. If Profiled Voice Application QoS is Poor and Profiled VC Application is Poor and Profiled VS Application QoS is Poor then RAN QoS is Poor
2. If Profiled Voice Application QoS is Poor and Profiled VC Application is Poor and Profiled VS Application QoS is Average then RAN QoS is Poor
3. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Poor and Profiled VS Application QoS is Good then RAN QoS is Poor
4. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Average and Profiled VS Application QoS is Poor then RAN QoS is Poor
5. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Good and Profiled VS Application QoS is Poor then RAN QoS is Poor
6. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Average and Profiled VS Application QoS is Good then RAN QoS is Poor
7. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Average and Profiled VS Application QoS is Good then RAN QoS is Poor
8. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Good and Profiled VS Application QoS is Poor then RAN QoS is Poor
9. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Good and Profiled VS Application QoS is Average then RAN QoS is Poor
10. If Profiled Voice Application QoS is Poor and Profiled VC Application QoS is Good and Profiled VS Application QoS is Good then RAN QoS is Poor

Rules for Network Configuration QoS Subsystem

1. If RAN1 QoS is Poor and RAN2 QoS is Poor then Network Configuration QoS is Poor
2. If RAN1 QoS is Poor and RAN2 QoS is Average then Network Configuration QoS is Poor
3. If RAN1 QoS is Poor and RAN2 QoS is Good then Network Configuration QoS is Poor
4. If RAN1 QoS is Average and RAN2 QoS is Poor then Network Configuration QoS is Poor
5. If RAN1 QoS is Average and RAN2 QoS is Average then Network Configuration QoS is Average
6. If RAN1 QoS is Average and RAN2 QoS is Good then Network Configuration QoS is Average
7. If RAN1 QoS is Good and RAN2 QoS is Poor then Network Configuration QoS is Poor
8. If RAN1 QoS is Good and RAN2 QoS is Average then Network Configuration QoS is Average
9. If RAN1 QoS is Good and RAN2 QoS is Good then Network Configuration QoS is Good