Sudan University of Science &Technology

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Performance Analysis of Triple play Services for IP Over WiMaX and DSL Networks

تحليل اداء خدمات التشغيل الثلاثى عبر شبكات الخط الرقمى المشترك غير المتماثل و البينيّة التشغيلية العالمية للولوج بالموجات الدقيقة

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Submitted by:

Asia Mahgoub Taha Ahmed

Supervised by:

Dr. Ibrahim Khider

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بسم الله الرحمن الرحيم

قال الله تعالى :

﴿وَقُلْ رَبِّ زِدْنِي عِلْمًا﴾

صدق الله العظيم

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DEDICATION

To the woman she gave me the power and organized my life to My *Mother*

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I would like to express my sincere gratitude to my Supervisor, **Dr. Ibrahim Khider** ***for providing his guidance and advice for the completion of this work. His strong hold & vast knowledge helped me a lot to come to the successful end of the project. Without him, the research could not have been completed.***

And of course my limitless thank and wonderful feeling to my family.

Abstract

In telecommunications, Triple Play service is a marketing term for provisioning of two bandwidth-intensive services, high-speed Internet access and television, and a less bandwidth-demanding (but more latency-sensitive) telephone service, over a single broadband connection. In this thesis, a Triple Play system performance over IP network is studied with possible techniques and mechanisms such as DSCP QoS algorithms, IP Multicast, IP Unicast techniques and VLAN mechanism, which are proposed to provide efficient Triple Play network management and improved application. Initial applications presented are entertainment video (video on demand VoD and multicast video IPTV), voice (VoIP), and best-effort data (e.g. web browsing, File sharing /downloading).

The performance of the proposed techniques and mechanisms with different broadband access network that support Triple Play services on the last mile are evaluated with simulation scenarios built using OPNET 14.5. The broadband access network used in the simulation of the proposed scenarios is composed of ADSL network to deliver Triple Play services with scalability at an optimal cost by using (IP DSLAM), and WiMax network to deliver adequate QoS to voice, video and data applications represented with a connectivity of 3.5GHz frequency bands with different radio channels bandwidth and with different path loss and multipath channel models. It explores how technologies differ and how can they be combined to provide a total last-mile access solution at the present time and in the future. Simulation results from the previous networks show that such networks can support Triple Play services including demanding real time services with the required QoS. From the results, ADSL has exhibited behavior that approach the ideal values for the performance metrics of the Triple Play services, while WiMax and wireless network have demonstrated promising behavior within the bounds of the defined metrics.

المستخلص

فى الاتصالات السلكية والاسلكية خدمة اللعب الثلاثى هى المصطلح لتوفير اثنيين من الخدمات التى تتطلب نطاق ترددى قوى ، الوصول العالى السرعة الى الانترنت والتلفزيون ،وخدمة الاقل تطلبا للنطاق الترددى (ولكن الاكثر حساسية من ناحية التأخير)، الهاتف ، عبر اتصال واحد للنطاق عريض . فى هذة الاطروحة ، تم دراسة أ داء دراسة نظام التشغيل الثلاثى عبر شبكات خط المشترك الرقمى غير المتتماثل والبينية التشغيلية العالمية للولوج بالموجات الدقيقة مع التقنيات الممكنة وآليات مثل خوارزميات جودة الخدمة ، الإرسال المتعدد عبر شبكة برتكول الانترنت وتقنيات الفريدة عبر برتكول الانترنت والية الشبكات التخيلية ، والتي اقترحت لتوفير كفاءة لادارة شبكة التشغيل الثلاثي وتحسين التطبيق . التطبيقات الأوليه المقدمه من الوسائل الترفيهيه هي الفيديو(الفيديو عند الطلب والإرسال المتعدد للفيديو) ، الصوت (الصوت عبر بروتوكول الإنترنت) ، وأفضل جهد للبيانات (مثل تصفح الإنترنت، وتبادل الملفات / تحميل).

يتم تقييم ادأ التقنيات والاليات المقترحة مع مختلف النطاق العريض للوصول الى الشبكة التى تدعم خدمات التشغيل الثلاثى على الميل الأخير مع سيناريوهات المحاكاة التى تم إنشاؤها باستخدام الاوبنت14.5 . وتتكون شبكة النفاذ عريض النطاق المستخدم في سيناريوهات المحاكاة من شبكة خط المشترك الرقمى غير المتماثل لتقديم خدمات التشغيل الثلاثي مع قابلية بتكلفة الأمثل بواسطه استخدام شبكةاجهزة الوصول لخط المشترك الرقمى عبر بروتوكول الإنترنت، وشبكة والبينية التشغيلية العالمية للولوج بالموجات الدقيقة في تقديم جود ة الخدمة كافية لتطبيقات الصوت ،الفيديو والبيانات التي تربط لنطاق ترددي 3.5 مليار ذبذبة مع قنوات إذاعية مختلفة من ناحيه فقدان المسار ونماذج المسارات المتعددة. ونستكشف كيف يمكن للتكنولوجيات ان تختلف وكيف يمكن دمجهما لتقديم مجموعه الحل للميل الأخير في الوقت الحاضر وفي المستقبل .نتائج المحاكاة من الشبكات السابقة تبين أن هذه الشبكات يمكن أن تدعم خدمات التشغيل الثلاثي بما في ذلك الخدمات للطالبين في الوقت الحقيقي مع جودة الخدمة المطلوبة. كما لوحظ من النتائج، شبكه المشترك الرقمى غير المتتماثل قد عرضت السلوك الذي اقترب من القيم المثالية لمقاييس أداء خدمات في حين ان شبكه البينية التشغيلية العالمية للولوج بالموجات الدقيقة أثبتت السلوك في حدود المقاييس المحددة.

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List of Abbreviations

Abbreviations	Meaning
3G	Third Generation
4G	Fourth Generation
AAA	Authentication, Authorization and Accounting
ADSL	Asymmetric Digital Subscriber Line
AEN	Access Edge Node
AF	Assured Forwarding
AMC	Adaptive Modulation and Coding
AN	Access Network
AP	Access Point
ASN	Access Service Network
ATM	Asynchronous Transfer Mode
BE	Best Effort
Bi-Dir PIM	Bidirectional Protocol Independent Multicast
BNG	Broadband Network Gateway
BPL	Broadband Power Line
BRAS	Broadband Remote Access Server
BS	Base Station
CAR	Commited Access Rate
CATV	Cable Television
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CE	Consumer Electronics
CO	Center Offices

CPE	Customer Premises Equipment
CPGA	Cost Per Gross Add
CQ	Custom Queuing
DiffServ	Differentiated Service
DL	Downlink
DLCs	Broadband Digital Loop Carriers
DOCSIS	Data Over Cable System Interface Specification
DS3	Digital Signal 3
DSCP	Differentiated Service Code Point
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexers
DSR	Distributed Services Router
E2E	End to End
EF	Expedited Forwarding
EPON	Ethernet Passive Optical Network
ertPS	extended real-time Polling Service
ETSI	European Telecommunications Standards Institute
FIFO	First Input First Output
FSO	Free Space Optics
FTP	File Transfer Protocol
FTTB	Fiber To The Building
FTTC	Fiber To The Curb
FTTH	Fiber To The Home
FTTN	Fiber To The Node
FTT*x*	Fiber To The any type of access network
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GPS	Global Positioning System

GSM	Global System for Mobile Communications
HD	High-Definition video
HDMI	High-Definition Multimedia Interface
HFC	Hybrid Fiber Coax
HSI	High Speed Internet
HTTP	Hyper Text Transfer Protocol
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IGMP	Internet Group Management Protocol
IGMPv1	Internet Group Management Protocol version1
IGMPv2	Internet Group Management Protocol version2
IOS	Internetwork Operating System
IP	Internet Protocol
IPTV	Internet Protocol Television
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
KPN	Koninklijke PTT Nederland
L2	Layer-2
L3	Layer-3
LAN	Local Area Network
LMDS	Local Multipoint Distribution Service
LOS	Line Of Site
LTE	Long Term Evolution
mAAA	multicast Authentication, Authorization and Accounting
MAC	Media Access Control
MAN	Metropolitan Area Network
MCU	Multipoint Control Unit

MIMO	Multiple-Input and Multiple-Output
MLR	Media Loss Rate
MMDS	Multichannel Multipoint Distribution Service
MP3	MPEG-2 Layer 3
MPEG-2	Moving Pictures Experts Group_2
MPEG-4	Moving Pictures Experts Group_4
MPLS	MultiProtocol Label Switching
NGA	Next Generation Access Network
NGN	Next Generation Network
NLOS	Non Line-of-Sight
NMS	Network Management Systems
nrtPS	non-real-time Polling Service
OC	Optical Carrier
OFDM	Orthogonal Frequency Division Multiplex
OFDMA	Orthogonal Frequency Division Multiple Access
OLT	Optical Line Terminal
ONU	Optical Network Units
OoS	Out of Sequence
OPNET	Optimized Network Engineering Tools
OTA	Over The Air
PC	Personal Computer
PCM	Pulse-Code Modulation
PHB	Per Hop Behaviour
PHY	PHYsical
PIM	Protocol Independent Multicast
PIM-SM	Protocol Independent Multicast -Sparse Mode
PLR	Packet Loss Ratio
PMP	Point-to-Multipoint

PoP	Point of Presence
POTS	Plain Old Telephone System
PPP	Point-to-Point
PPPoA	Point-to-Point Protocol over ATM
PPPoE	Point-to-Point Protocol over Ethernet
PQ	Priority Queuing
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoE	Quality of Experience
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RED	Random Early Detection
RF	Radio Frequency
RFC	Request for Comments
RG	Residential Gateway
RP	Reference Points
RSVP	Resource Reservation Protocol
RTP	Real Time Protocol
rtPS	real-time Polling Service
Rx	Receiver
SD	Standard Definition
SDU	Session Data Unit
SISO	Single Input and Single Output
SLA	Services Level Agreement
SONET	Synchronous Optical NETworking
SPs	Service Providers
SS	Subscriber Station
SSM	Source-Specific Multicast

STB	Set Top Box
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol / Internet Protocol
TELRIC	Total Element based Long Run Incremental Cost
ToS	Type of Service
TV	Television
Tx	Transmitter
UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
UL	Upload
UMTS	Universal Mobile Telecommunications System
VAD	Voice Activity Detection
VBR	Variable Bit Rate
VLAN	Virtual Local Area Network
VoD	Video on Demand
VoIP	Voice over Internet Protocol
WAN	Wide Area Network
WDM/TDM	Wavelength Division Multiplexed /Time Division Multiplexed
WFQ	Weight Fair Queuing
WFQ-LLQ	Weight Fair Queuing - Low Latency Queuing
WiFi	Wireless Fidelity
WiMax	Worldwide interoperability for Microwave access
WLAN	Wireless Local Area Network
WWAN	Wireless Wide Area Networ

CHAPTER ONE

INTRODUCTION

1.1 Preface :

Triple Play service is a concept that delivers digital television broadcast, a telephone, and an Internet access to people through a single broadband connection. It is currently a term that is well-known throughout the telecommunication world. Introducing Triple Play service has enabled incumbents to hang onto their existing customers, and the report estimates that the rate of line loss will halve to around 2.5% a year from 2008 to 2015 [1]. According to a new report from "Digital TV Research", global Triple Play subscriptions are expected to reach 400 million by 2017, nearly 300 million on the end 2011 and up by 380 million on the 2007 total [2]. Another "Triple Play Forecasts" report, estimates that in Asia Pacific region 257 million will contribute by 2017 total, and more than 200 million on the end 2011 total. China alone is expected to have 193 million Triple Play subscribers by 2017 (with only 39 million recorded at end 2011) [2].

Broadband access is now a reality in many parts of the world and launching Triple Play services are undergoing in many countries. Triple Play services will bring many benefits to the end users. E.g., they can enjoy their high quality movies at any time of the day, and their expensive, fixed telephony, is replaces with cheaper VoIP telephony service. In fact in most of the major cities, the competitive market between telecom, cable TV and green field operators has already led to restructure the Triple Play business models forcing the operators to package bundles of their Triple Play services with discounts and promotions which has led to significant price declines [3].

However, despite packaging video with voice and data services, it provides benefits beyond the cost reduction of using a common infrastructure. New features and capabilities can be implemented through the interaction between these three services. Ads customized for user or region can be inserted in a subscriber's preferred video content. Incoming voice call information can be delivered on-screen, while the video transport capabilities of the environment can be leveraged for video telephony. Today prices range from € 3.99 to € 14.99 (with or without flat rates) for telephony service or

€ 9.99 to € 59.99 for broadband Internet. be priced at discount bundles [4].

prices range from € 3.99 to € 14.99 (with or without flat rates) for telephony service, or € 9.99 to € 59.99 for broadband Internet. Alternatively, combined service plans can be priced at discount bundles [4].

One critical question facing here in this thesis is whether the architecture and capabilities of wired and wireless access networks will eventually converge. The degree of convergence will determine whether wired and wireless services function as substitutes or complements, with strong implications for market structure and regulatory policy. Wired access networks that were historically distinct, notably switched voice versus broadcast video, are evolved to a common "platform network" architecture. Wired platform networks use the high capacity of fiber-optic-rich physical networks and the general-purpose capability of IP-based protocols to support Triple Play of voice, video, and data services. To make things even more complicated, it is not always possible to provide customers with a wired Internet access network like DSL or Cable Television (CATV). This is often the case in sparsely populated areas, where it would be too expensive to set up wires. For Triple Play services, wireless access network technology has to be used in the future. In the wireless domain, albeit with a lag, there is a similar trend towards increasing capacity and providing a range of services over a common IP-centric network infrastructure. Recent wireless broadband networks such as 3G LTE and WiMax provide a general-purpose IP platform with over-the-top services at the application layer, which is similar to the design of wired IP platform networks [5].

So, in this thesis, WiMax technology is used which has become the easiest way for wireless communication and a solution to rapid requirement of Internet connection for data, voice and video services. In 2007, more than one hundred WiMax carrier trials were planned worldwide. Furthermore, the WiMax forum in March 2008 issued a press release projecting 133 million subscribers by 2012. In February 2009, the WiMax Forum has reported just under 460 WiMax fixed and mobile deployments worldwide along with over 800 million subscribers projected by 2010. Intel which has projected over 1.3 billion people has access to WiMax by 2012 [6].

The WiMax MAC layer has a connection-oriented architecture that is designe to support a variety of applications, including voice and multimedia services. One of the potential applications of WiMax is to provide backbone support for mobile WiFi hotspots. Traditionally wired connections are used as backhaul support for WiFi hotspots. However, wired infrastructure is always considered expensive and should be replaced by an alternative technology [7]. So, , the complementary nature of a hybrid wireless network (WiMax with WiFi) is explored, and an explanation of how service providers can leverage these technologies to offer wireless broadband Triple Play services connectivity and compel new services at affordable prices and in more locations. In addition, this study focuses on the synergies between the IEEE 802.11g Orthogonal Frequency Division Multiplexing (OFDM) and IEEE 802.16e-2005 OFDMA air interfaces.

In order to allow an efficiently fast and cheap access to the broadband access network infrastructure, WiMax and WiFi together with the DSL can benefit services in an inexpensive way to address the users' demand in terms of bandwidth, QoS, etc. In this work, the focus was also on studying the combination of WiMax and WiFi technologies with DSL networks. The analysis presented here is partial because the focus was on the implications of technical and architectural issues. It has been recognized that the evolution of markets, industry structure, and public policy depend on many other factors that are beyond the scope of this thesis. Other factors, include organizational aspects of operator businesses, what successful service options in the marketplace, and market responses to change in policies regulatory.

1.2 Problem Statement

The Triple Play services involve separating services & forwarding packet. The issue of isolating each service to apply diversifying topologies and infrastructure Management on a service basis is crucial for the transport network of the Triple Play architecture. Therefore, isolation can be translated as a separate packet forwarding and routing per service type. This can be enabling through the use of the distinct IEEE 802.11 (VLAN) for different services .

IPTV and VoD services need high bandwidth. Depending on the compression and coding technologies, the following transmission rates should be considered More compression means less bandwidth requirement.

1. A MPEG-2 coded SD VoD video stream or IPTV stream is 3.5 to 5 Mbit/s per TV channel.

2. A MPEG-4 coded SD VoD video stream or IPTV stream is up to 2 Mbit/s per TV channel.

3. A HD TV channel uses 8 to 12 Mbit/s when coded with H.264.

The IP platform is robust and efficient; however it is not suitable for the transmission of time-critical data streams. Delay time between packages and package loss is among the most prominent features of an IP network that caused a problem for the audio and video communication. To prevent these problems will be implemented in higher protocol layers mechanisms to monitor and support the QoS measurement.

1.3 RELATED WORK

1.2 Kampanakis *et al*., in 2006 [13] studied the basics of Triple Play and Quadruple Play services, and a few common and popular applications for the Triple Play architecture. They also included on their study the pricing model adapted to support Triple Play services followed by a simulation study on OPNET 11.5 to analyze and understand QoS parameters, i.e. end-to-end delay and jitter experienced by voice, video and data traffic as they traverse routers configured with various scheduling policies.

R. K. Kalle *et al.*, in 2007 [16] discussed network system architecture in which the 802.16e operating in PMP mode was used for last mile access and the authors investigated some of the Triple Play applications which include e-Education and Infotainment. A simulation study of the delivery of Triple Play service over 802.16e was reported and the performance evaluation for a typical emerging market scenario indicated that 6-8 simultaneous video sessions can be supported for over an 802.16e network operating in PMP mode of operation.

L. Shi *et al*., in 2008 [17] studied how to achieve Network Utility Maximization (NUM) in NGN running Triple Play services. By investigating the characteristics of most of its traffic classes, they explicitly presented their utilities as the function of allocated bandwidth. They also further formulated the NUM objective as a nonlinear programming problem with both inequality and equality constraints. Several useful results are presented on the new features of the NUM-based scheduling.

K. Ozdemir *et al*., in 2009 [18] analyzed user capacity of mobile WiMax systems for each of these three services (Triple Play) and considered various link characteristics, interference scenarios, modulations, QoS classes, and QoE requirements. He also analyzed the impact of header compression and suppression techniques and their effect on capacity. In the same year, **C. A . Papagianni *et al*.,** [14] examined in particular the performance of common packet scheduling .The performance of PQ, WFQ and WFQ-LLQ schemes was assessed, in order to find the most appropriate solution for the underlying network. Also in the same year, **F. Wan *et al*.,** [19] considered for home networks a heterogeneous wired and wireless network architecture to support Internet Protocol TV (IPTV), voice, and data, the so-called Triple Play services. To satisfy the quality of service (QoS) requirements for different traffic classes, class-based queuing (CBQ) is deployed at home gateways and routers. Simulation results over wired and multi-hop wireless paths are given which validate the analysis. The results presented provide important guidelines for the planning of future home networks for Triple Play services. They also provide important insights into how to efficiently support heterogeneous traffic with stringent QoS requirements over wireless and wired networks. In addition to the above **T. Uhl, in 2009** [20] focused on theoretical and practical demonstration of the current methods to evaluate the QoS during Triple Play services transmission.

N. Zotos *et al*., in 2010 [15] presented an experimental network infrastructure providing E2E

QoS, using a combination of MPLS and DiffServ technologies in the core network and WiMAX technology as the wireless access medium for high priority services (VoIP, High Quality Video Streaming) transmission. Also in 2010 **Y. Zhang *et al*.,** [21] offered a good solution for Triple Play integrated automation network. It integrates the surface network, underground network and wireless communication network in the same unified platform, so as to get maximum information sharing function and achieving stronger spot monitor, in addition to control the capabilities of the network. The application on Xinglong Coal Mine turned out to be a big success of the Triple Play integrated automation network in practice

I. Papapanagiotou and M. Devetsikiotis, in 2010 [22] the choice of an appropriate architectural approach and sizing model for the aggregation network is studied through cost optimization models, which encompass aspects of non-stop delivery, service flexibility, policy management and cost allocation for Triple Play services.

M. Baldi and P. d. Torino, in 2011 [23] presented a low complexity solution for proving Triple Play services that enables integrating different applications, so that each received required service is received, possibly in a guaranteed fashion, while ensuring high resource utilization efficiency. In the same year, **F. Khan *et al*.,** [24] presented an enhancement to our proposed hybrid (WDM/TDM) architecture to support Triple Play services and employs EPON within last mile to address the access bottleneck issue. Also, he used digital transmission for data and voice services with analog video broadcast service is realized using RF overlay model. In addition to the above **Y. Hao-wei *et al*., in 2011** [25] briefly introduced the position of Triple Play system in the whole government emergency management system, and discussed the development aim of Triple Play system, the flow of police alert receipt and dispatching, the construction content and the functions to be achieved.

O. Schilke *et al*., in 2012 [26] focused on the bundle of broadband Triple Play. Technological advances have given telecommunications service providers a way to offer a full array of Internet services, which they often combine into one bundled package. Triple Play has emerged as a term in the business press that describes such offerings; it denotes the bundling of three broadband services: Internet access, telephone, and television. Also, **A. E. Garcia *et al.*, in 2012** [27] proposed a cost model which considers QoS parameters for Triple Play services. This model is based on the “Total Element based Long Run Incremental Cost” (TELRIC), which is applied to the wholesale access and interconnection paradigm. Three traffic engineering methods were considered and studied for network dimensioning. Hereby the aim was to guarantee QoS of different services: complete traffic segregation under virtual tunnels, complete traffic integration by over engineering, and partial traffic integration using a priority queuing scheme. The proposed method enables the development of a specific cost scheme based on a complete scenario taking into consideration different types of users. The variety of used IP applications supposes direct implications over different levels of interconnection, mainly at the low-level Metro access and the high-level edge node ,

1.4 Objectives:

The Objectives of this thesis are to:

- To discuss in more detail the Triple Play services delivered over ASDL and WiMax and the possible subservices offered with the required QoS.
- To Study the possible techniques and mechanisms for allow in on have efficient Triple Play network scheme and for improving application response times by solving applications network congestion and supporting smooth application deployments. Examples of techniques and mechanisms are DSCP QoS algorithms, IP multicast, IP unicast techniques and VLAN mechanism.
- Design proposed model for Triple Play system using the OPNET v14.5 Network Simulator. Comparing and analyzing its performance has been tried under various scenarios, including well accepted Triple Play network models by applying different mechanisms, applications and broadband access network.
- To discuss briefly the simulation results for each scenario.

1.5 Methodology :

OPNET Modeler 14.5 simulator is chosen for simulation to examine the performance of the broadband infrastructures and their support for Triple Play services under two separated scenarios.

- Examining and delivering Triple Play services using ADSL technology. This examines the capacity, in addition, to the availability of ADSL links specifically for video over twisted pair, which is a booming technology called IPTV. To compare between two compress methods, video service is streamed to the subscriber using the MPEG-2and MPEG-4 format.

- Examining and delivering Triple Play services using WiMax technology. The best needed bandwidth for a carrier frequency of 3.5GHz with different distances between BS and SS. is tested. The pathloss and multipath models was simulated .

1.6 Outlines

The remainder of this work is organized as follows:

Chapter2 explains Triple Play system with a basic overview of the Triple Play network architecture and the possible services that can be offered as part of the Triple Play system. In addition, to study the requirement of these services towards broadband access network and how it can be used to support Triple Play services. Finally the chapter ends with the technical aspects of service delivery such as IP multicast techniques and how they can support some applications of Triple Play.

Chapter3 gives an introduction to the network simulation. Network architecture, simulation strategy and models to different broadband infrastructures simulated scenario with Triple Play services has been designed using the principles presented in chapter 2 in this thesis.

Chapter4 discusses and analyses the simulation results for all simulated scenarios.

Chapter5 presents the conclusion of this thesis and the future work future .

CHAPTER TWO

TRIPLE PLAY SYSTEM

2.1 Introduction

This chapter introduces the Triple Play system and explains how its network architectures work. The types of Triple Play services and some of the well-known services are then described. It also demonstrates how the system can delivers its services over different broadband access connection with different requirements. Advantages and disadvantages of the access broadband and the techniques that are required to deliver these services are also discussed in this chapter.

2.2 Next Generation Triple Play Network

Telecom operators are now employing new strategies to deliver thrilling new services using next generation Triple Play networks. The full package of this service includes: line rental and telephony with a combination of Internet access, IPTV, VoD, entertainment applications and, eventually, cellular phones. In other words this means: multiple services, multiple devices, but one network, with different vendor and one bill. This manoeuvre is much more than just a new commercial product. It is a consequence of the important changes the industry is undergoing, such as technological innovations, social changes and new regulations.

Furthermore, next generation Triple Play networks are capable of connecting wired and wireless subscribers, providing flexible and fast service, provisioning high bandwidth with high QoS, and reduced overall service cost. The major components of telecommunication networks that are constructed for various types of wired and wireless architectures to support services of Triple Play network that are shown in Figure 2.1 are [28]:

- Core networks that handle high volume aggregated transmissions between the network and the backhaul.
- Distributed networks that extend the line of sight (LoS) coverage area of the core

network.

- Local networks that interface directly with the end users.

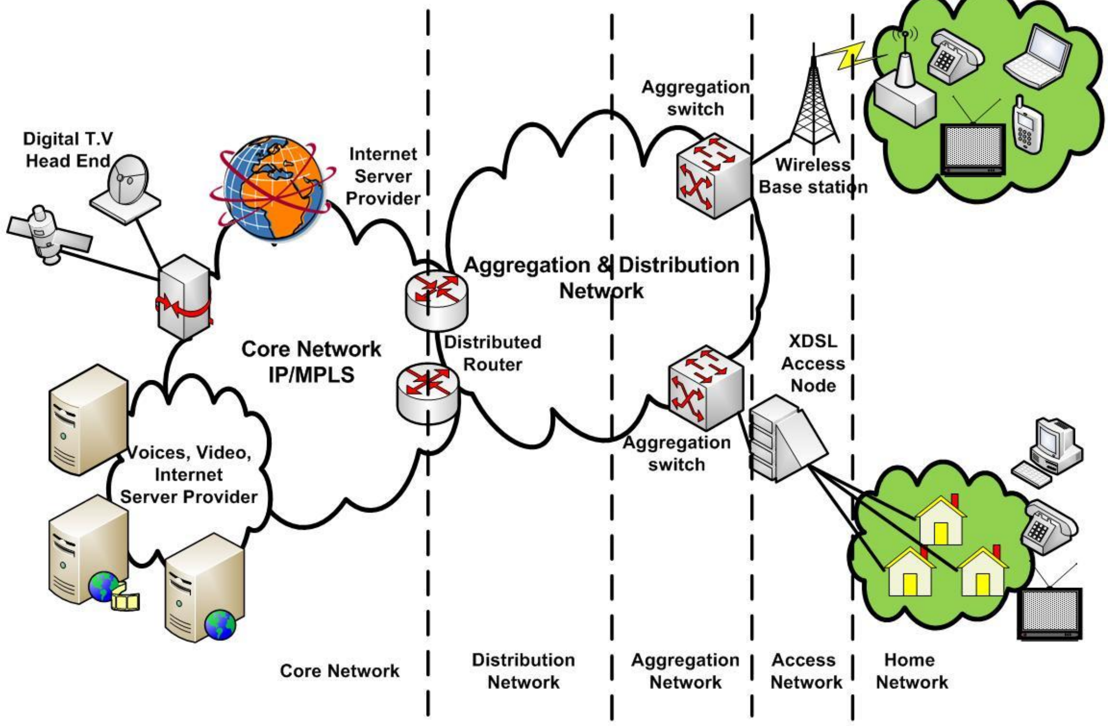


Figure 2.1: Overall next generation network environment [28]

2.3 Triple Play Services Delivery Architectures

Triple Play faces some new network challenges. Deploying the aforementioned services require some significant changes in today's network architecture. Triple Play services are not like traditional Internet services. Traditional Internet is "best-effort" service. This means that due to the fact that users are not online the same time, aggregating traffic from multiple users increases network efficiency. However, Triple Play includes real-time services, like voice and video. These kinds of services have strict end-to-end delay, jitter and bandwidth requirements. Therefore, Triple Play services are not treating as "best-effort" services. If a real-time service confronts packet delay or packet loss the connection is instantly dropped. From the above discussion it is obvious

that network delivery architectures are required for Triple Play services. The solution for voice is trivial though; since voice has small bandwidth requirements, all the problems could be alleviated by just reserving some network capacity for it.
However, with the video service which includes IPTV, VoD and HDTV, things are not so apparent. Users may now need up to 20Mbps to satisfy their needs. As a result, the Triple Play is postulates a new kind of delivery infrastructure architecture as can be shown in Figure 2.2 [13].

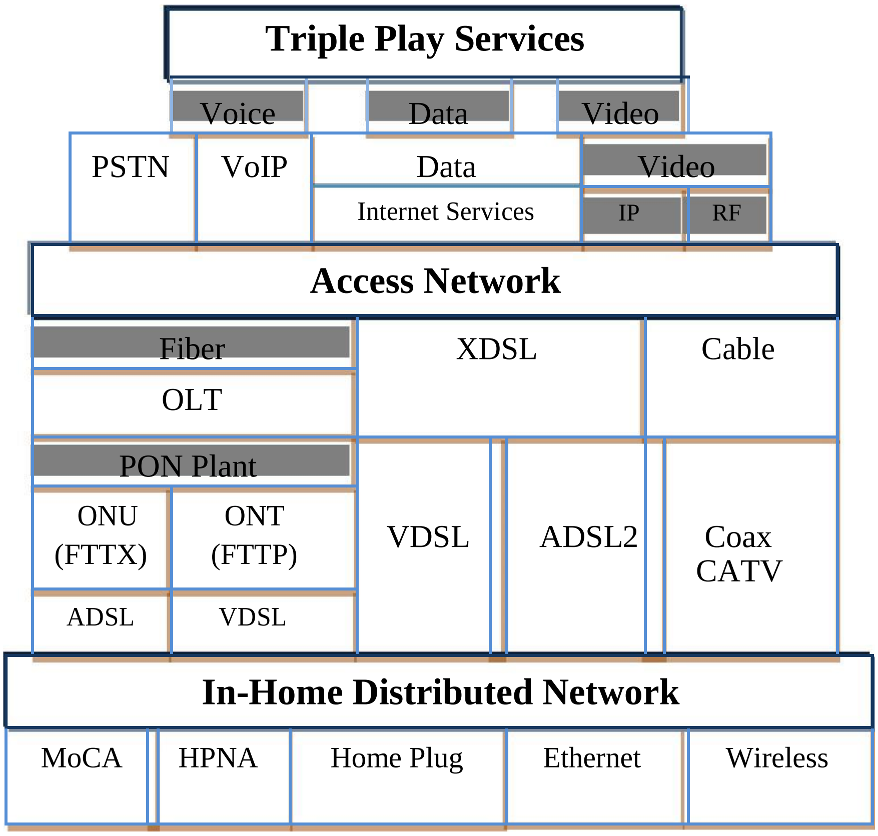


Figure 2.2: The building blocks of the Triple Play services delivery infrastructure [29]

After presenting the basic architecture of the Triple Play, it is worth analyzing some architecture proposed in the market. It will be useful to get an idea of how technology providers try to address the challenges of multiservice. Triple Play is considered to become a "killer application" as soon as its requirements are satisfied and market matures enough to provide it in a low cost. Increasing revenues lead many big companies to enter to the fields of IPTV and Triple Play in general. There are a number of different architectural approaches which claim to be able to integrate all the types of services in a flexible and scalable manner [22].

2.3.1 Centralized Edge Design

In this type of architecture, the L2 metro Ethernet aggregates the traffic from multiple access points before the IP edge network, as shown on Figure 2.3. Some of the characteristics of this architecture are:

- All types of traffic are backhauled to the Broadband Network Gateways (BNGs) and then to a single P-router or PoP location, which is connected to the ISP backbone.

- Subscriber termination functionality, multicast replication and IP QoS policies are executed in the BNG deeper in the network.

- IP multicast traffic for broadcast video is transmitted from the edge router over L2 multicast Virtual Local Area Networks (VLANs) to all customer premises.

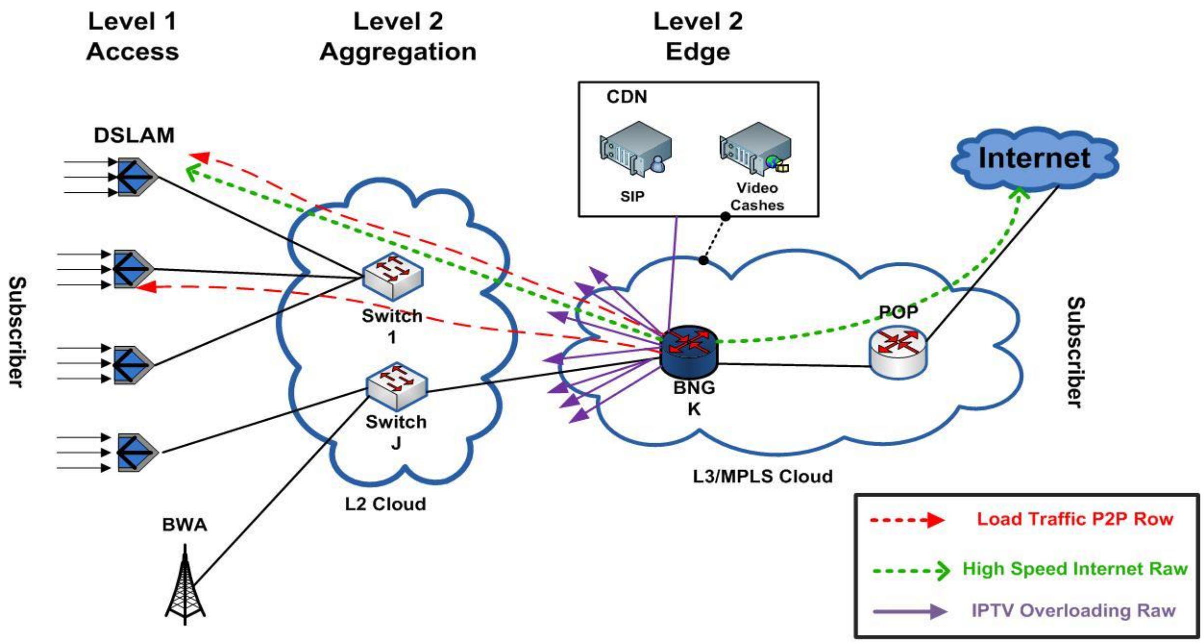


Figure 2.3: Centralized single edge overlay architecture [22]

2.3.2 Distributed IP Edge Design

A distributed IP edge approach is being considered by many SPs as an alternative architecture to satisfy the bandwidth requirements for future applications. As shown in Figure 2.4, the edge network is comprised by both L2 and L3 routers. decreased (less subscribers are terminated per BNG) and IP QoS is enforced closer to the last mile.

IP multicast routing is used across the L2/L3 carrier Ethernet network for delivery of broadcast video services.

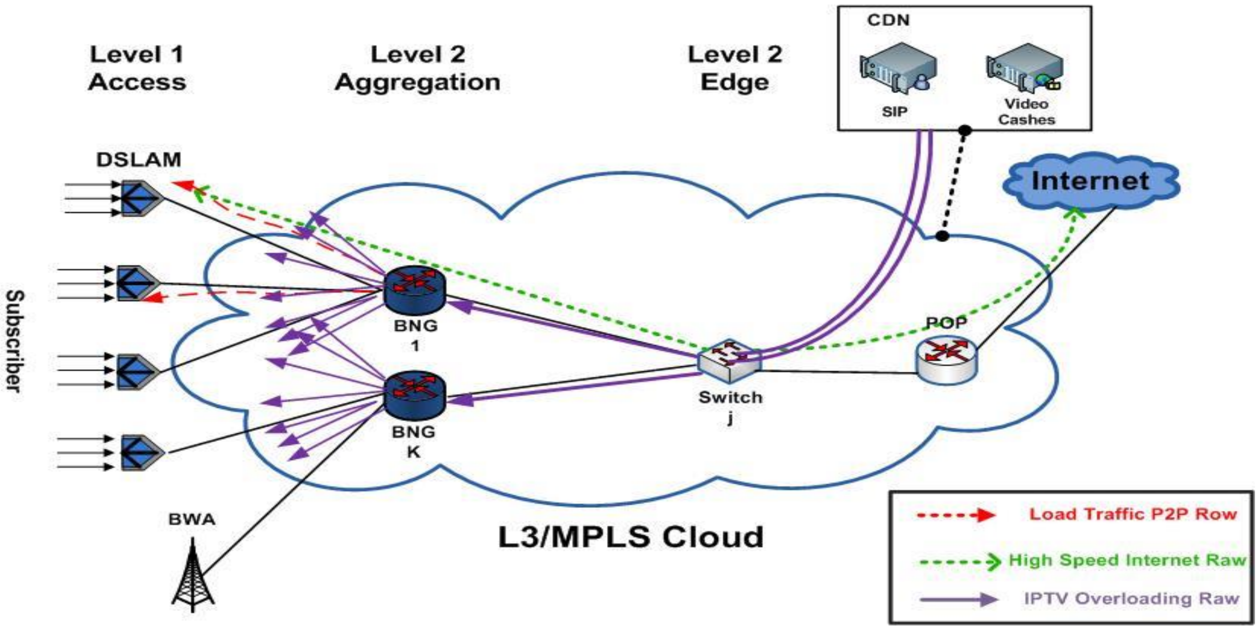


Figure 2.4: IP distributed single edge overlay architecture [22]

2.4 Types of Triple Play Services

The term "Triple Play service" covers a large collection of voice, video and data services, including: Video telephony / IPTV, which is multicast video (T.V channel) / Video on Demand (VoD), which is unicast video / Voice over IP (VoIP) / Gaming /

Internet access (HTTP, FTP traffic) / E-mail / etc..

In the following subtopics a description for each service of the Triple Play services is given. However, our attentions focus on delivering TV service. As a result, research analysis is focused on IPTV; believe that effectively transmitted video is the greatest challenge. Later, a brief review of the two other services, i.e. VoIP and data are continued. Finally, introductory section to Quadruple Play services is discussed.
Delivering high-quality video content is one of the latest and most demanding

2.4.1 IPTV

Delivering high-quality video content is one of the latest and most demanding challenges faced by the IP. With the deployment of these new IPTV services, the existing network infrastructures will be pushed to their limits. To accommodate the needs of IPTV services, networks must be able to scale to millions of customers, maximize bandwidth resources, and provide QoS and security on an end-to-end basis. For these and other reasons, network intelligence is critical when deploying video over broadband [29].

IPTV services may be classified into three main groups [31]:

- Live television, with or without interactivity related to the current TV show.
- Time-shifted television: catch-up TV (replays a TV show that was broadcast hours or days ago), start-over TV (replays the current TV show from its beginning).
- Video on Demand (VoD): browse a catalog of videos, not related to TV programming.

IPTV is distinguished from Internet television by its on-going standardization process (e.g., ETSI) and preferential deployment scenarios in subscriber-based telecommunications networks with high-speed access channels into end-user premises via set-top boxes or other customer-premises equipment [32].

However, IPTV is a sensitive service to the packet loss and delays for unreliable streamed data. IPTV has strict minimum speed requirements in order to ensure delivering the right number of frames per second for moving pictures. This means, limited connection speed and bandwidth available for a large IPTV customer can reduce the quality of delivered service. Their impact is briefly summarized below:

- The latency inherent by the use of satellite Internet is often held up as a reason why satellites cannot be successfully used for IPTV. However, in practice latency is not an important factor for IPTV. An IPTV service does not require a real-time transmission, as is the case with the telephony or videoconferencing services [33].
- Bandwidth requirements, for high-speed data transfer the needed bandwidth for the viewer is increased. For example, at least 2 Mb/s is needed for web-based applications on computer. Additionally, 64kbps is required for using landline telephone. In a minimal usage, 13 Mb/s is requires to process in a household with an IPTV Triple Play service.
- Privacy implications due to limitations in bandwidth, an IPTV channel is delivered to user one at a time, as opposed to the traditional multiplexed delivery. Changing a channel requires a request from the head-end server to provide a different broadcast stream; much like VoD (for VoD stream is delivered using Unicast, while for normal TV signal multicast is used). This could enable the service provider to accurately track each and every program watched with its duration for each viewer. Broadcasters and advertisers could then understand their audience better, subsequently programming with accurate data and targeted advertising [33].
- Service bundling for residential users, IPTV is often provided in conjunction with video on demand and may be bundled with Internet services such as Internet access and VoIP telecommunications services. Commercial bundling of IPTV, VoIP and Internet access is sometimes referred in marketing as Triple Play service. When these three services are offered with a cellular service, the combined service may be referred to as Quadruple play.

- Regulation historically, broadcast television has been regulated differently than telecommunications. As IPTV allows TV and VoD to be transmitted over IP networks, new regulatory issues arise [34]. Professor Eli M. Noam highlights in his report [35], "TV or Not TV: Three Screens, One Regulation?" some of the key challenges with sector specific regulation that is becoming obsolete due to convergence in this field.

Despite the challenges that are faced by the transfer of IPTV service, the number of global IPTV subscribers was expected to grow from 28 million in 2009 to 83 million in 2013. Europe and Asia are the leading territories in terms of the overall number of subscribers .But in terms of service revenues, Europe and North America generate a larger share of global revenue, due to very ARPU in China and India, the fastest growing and ultimately, the biggest markets is Asia. The global IPTV market revenues are forecast to grow from US$12 in 2009 to US$38 billion in 2013 [36].

2.4.2 Voice over Internet Protocol

VoIP is a technology that allows users to make telephone calls over an IP data network (Internet or Intranet) instead of traditional PSTN. Therefore; VoIP provides a solution that merges both data and voice, which gains benefits include; cost savings, high quality and value added services. There are many different audio codec's available for voice applications. The simplest and most widely used codecs are G.711, G.723 and G.729. The simplest encoder scheme is G.711 (64 kb/s). The acceptable packet loss factor of G.711 is up to 0.928% [37]. So, the encoder scheme G.711 (64 kb/s) is used for VoIP service in the simulation project.

Furthermore, it is important to state that Ethernet and IP layers are critical to the deployment and troubleshooting of VoIP service. They have significant impact on the overall QoE parameters, including dropped calls and call quality. Items that affect the overall transport quality of VoIP service are; IP packet delay, loss, jitter, and out of sequence (OoS) packets. These items are briefly summarized below [29]:

- Packet delays can have varying effects on voice quality, so it is important to measure the delay at the time of service installation to provide a benchmark for verification against potential problems.
- Packet loss can occur for a variety of reasons inside a network. Periodic losses in excess of 5 to 10 percent of all transmitted voice packets will degrade voice quality significantly.
- Packet jitter will make speech choppy and difficult to understand. For high-quality voice, the average inter-arrival packet time at the receiver should be nearly equal to the inter-packet gaps at the transmitter and the standard deviation should be low.

iv. Out of sequence packets have a similar effect as those of lost packets because voice codec's often discard them. It is also important to note that lost packets and OoS packets are always measured on the local link and can help in the segmentation of the overall problem.

codec's often discard them. It is also important to note that lost packets and OoS packets are always measured on the local link and can help in the segmentation of the overall problem.

However, in the enterprise market, IP is opening up major new VoIP-based opportunities. Forrester Research [38] predicts that by 2015, 95 percent of enterprise voice calls will be VoIP-based. However, the fastest growing VoIP market is "hosted IP voice" or "IP Centrex" which is expanded from about US$60 million in 2004 to more than US$7.6 *billion* in 2010, representing a CAGR of 282 percent. By that time, VoIP technologies are expected to be handling 45 percent of the total voice telephony market [38].

2.4.3 Data Service

Data service refer to activities familiar to all of us like web browsing, file downloading, e-mail, electronic purchases, electronic games and other applications using HSI. The Internet has a major and a successful cultural phenomenon in the past few years. The drivers to its success is the convenience and benefits of applications like e-mail and web browsing, combined with low cost and wide reach. However, today Internet needs something else to remain successful and maintain its promise as a ubiquitous and universal means of accessing any type of information. It must be a profitable business or, at least, economically self-sustainable. In order to achieve this objective, the services that people are willing to pay, must be offered over the Internet and the cost of delivering them must be low. In order to be able to properly bill for such services, they must be reliable, and feature high and persistent qualities. Moreover, a Triple Play solution with a low technological complexity is needed to support single network infrastructures, which include traditional Internet application (e.g., web, e-mail, and file sharing), telephony, and video. Current solutions for integrating traffic of heterogeneous nature on the same network infrastructure are leverage on over provisioning to ensure satisfactory quality.

This leads to poor utilization of network resources, thus conflicting with the objective of minimizing costs [23]. This thesis presents an optimal solution for supporting Triple

Play through IP networks, and specifically the Internet, as it offers:

- Low complexity, hence high scalability of routers.
- QoS guarantees (deterministic delay and jitter, no loss) for (UDP-based) CBR and VBR streaming applications.
- The service received by elastic, e.g., TCP-based applications, and current employed network design and traffic engineering practices and models.

2.3.4.Quadruple Play

A Quadruple play service is the little brother of the Triple Play service. It combines the Triple Services (HSI, voice and video) with wireless services. In this thesis, when wireless services are mentioned, rather than just having a cellular-like service, it is the ability to have wireless access to all the aforementioned services. Given the advancements in WiMax, the ability to transfer information over a wireless link at combinations of speeds, distances and NLOS conditions is rapidly improved. WiMax is a wireless technology that has the ability to provide broadband connections over long distances. Moreover, advances in both CDMA and GSM standards, and utilizing 3G, 4G or UMTS allow the service operators to enter into Quadruple Play and gain competitive advantage against other providers [13].

2.5 The Requirements of the Investigated Triple Play Services

There are many different services that are currently available in home, and even more when looking towards the near future. These services can be audio/video entertainment, games, news, education, personal communications, telemedicine, home control, financial management, asset management and teleworking. These numerous (end-user) services each have their specific requirements [39]. In the following subtopics, the types of requirement for Triple Play services towards broadband networks are discussed. The focus was on the requirements of Triple Play related services (telephony broadcast TV, on demand video and Internet related services) towards the broadband telephony-, cable-, wireless- and fiber-networks.

2.5.1 Bandwidth Requirements

Next generation services of Triple Play (data/voice/video) require far more bandwidth than the existing technologies like dial-up, ISDN, cable, and DSL that could offer. The gulf between the access and metro networks is widening due to bandwidth difference. This problem is termed as "access bottleneck". The most apparent requirement is the speed which is needed by the service to transport efficiently over the network. However there are more challenges for traffic/speed related requirements to deliver Triple Play services efficiently with cost effectively, these are explained below [40]:

- **Data rate and traffic direction: Symmetric/Asymmetric**
  The most important bandwidth requirement of services is data rate, this may differ in the up- or downstream direction from narrowband (<128 kbit/s) to mid-broadband (128kbit/s – 10Mbit/s) and super broadband (10Mbit/s-1Gbit/s) or even ultra-broadband (>1 Gbit/s). If a service uses more / less in either the up- or downstream direction this connection is called asymmetric (e.g. watching a video which uses more downstream capacity), other services use approximately the same speed in both up as downstream (like placing a telephone call over IP or video-conferencing).
- **Traffic Duplication: Point-Point, Point-Multipoint, Multipoint-Multipoint**
  Services can communicate information purely to one user (Point-Point) like on demand video (specifically requested by one person), other services can communicate information to multiple users (Point-Multipoint) like broadcast television over IP, or gaming (Multipoint-Multipoint). Traffic pattern: constant, variable or burst services sometimes require a CBR connection (like a 128 kbit/s MP3 audio stream) or generate traffic varying between a certain range of bit rate (like certain VBR streaming video content), or generate 'bursty' traffic at a high data rate (like a query on a web page) or a file download.

2.5.2 Network Quality of Service Requirements

QoS is a concept which promises to compensate the conflict between different services. To achieve this goal, it is necessary to priorities individual service and to allow a grouping in the so-called class of service. The idea of QoS is to assign related service requirements to a group of applications or users in order to equate service features, hence isolating them from other applications. Nowadays, there are several mechanisms available for QoS techniques; the most common mechanisms include [19]: IEEE 802.1p/Q standards, MPLS, DiffServ and RSVP.

These mechanisms implement the tagging (marking) of certain types of traffic (e.g. voice, video, Internet-traffic) and use it to serve, e.g. voice traffic with a higher priority than the video and Internet traffic (which have to wait a bit longer than the voice-traffic to be transmitted) [40]. It must be emphasized that these mechanisms act on different layers of the logical network model, which is shown in Figure 2.5 [20].

Application Layer
Transport Layer (RSVP)
Network Layer (Differentiated Services)
MPLS
Network Access Layer (IEEE 802.1p/Q)

Figure 2.5: QoS techniques in the TCP/IP model [20]

In an IP transport platform, there are a number of parameters that influence the quality of service, which are used in the project to evaluate the different behaviors experience in the simulation. In relation to the Triple Play services, these parameters are: packet loss, network delay, jitters, throughput, FTP downloads response time, HTTP page response time, and buffer overflow percentage. These parameters are explained below.

i. **End to End Delay:** it is the time duration in seconds from the instant when a packet arrives at the source node until it is received at the destination node. This metric can be calculated as follows [41]:

$$d_{enden\ d} = Q(d_{proc} + d_{queue} + d_{trans} + d_{prop}) \qquad (2.1)$$

Where: ***(Q)*** is the number of network elements (routers, switches and firewalls) between sender and receiver. ***(d_{proc})*** is the processing delay at a given network element. ***(d_{queue})*** is the queuing delay at given network element. ***(d_{trans})*** is the transmission time of a packet on a given link and ***(d_{prop})*** is the propagation delay across a given network link

ii. **Jitter:** is defined to be the variation of delay between consecutive packets. If two consequetive packets leave the source node with time stamps *t1* and *t2* and arrive the destination at time *t3* and *t4* after reassembly and play back, then the jitter is represented by equation (2.2) [42]:

$$Jitter = (t4 - t3) - (t2 - t1) \quad (2.2)$$

Where :*(t4 - t3)* is the expected packet reception time and *(t2 - t1)* is the actual packet reception time Negative jitter means that the packets where received in different time range i.e (t4 - t3) < (t2 - t1).

iii. **Packet Loss Ratio:** PLR is the number of corrupted, dropped, or excessively delayed packets in relation to the total number of packets expected at the client

iv. **Throughput:** it is defined as the traffic load that the media stream will impress upon the network. It can be measured in bytes/sec (Bps) or bits/sec (bps). with variable bit rate encoder, the traffic loading is dynamic in nature and a function of the scene complexity and the associated audio content. Consequently, VBR traffic loads are typically quoted in throughput ranges.

v. **FTP downloads response time:** is the time elapsed between sending an FTP request to the FTP server and receiving the complete response packet. It includes the signaling delay for the established and terminated connection. The download response time is an indicator of the user-perceived latency of the downloaded FTP file. In this experiment all time specific information regarding download has been the signaling delay for the established and terminated connection. The download response time is an indicator of the user-perceived latency of the downloaded FTP file this experiment all time specific information regarding download has been collected. Later the downloaded time has been considered according to equation (2.3) [46]

Download Time = Total Time – (Name Lookup Time + Connection Time) (2.3)

vi. **HTTP page response time:** the page response time is the time elapsed between sending the HTTP request to the HTTP server and receiving the complete response of the entire web page with all contained embedded objects. The main page object and the embedded objects are downloaded using a single TCP connection with HTTP 1.1. The HTTP paging response time is an indicator of the user-perceived latency of the web page retrieval [47].

v. **Buffer Overflow Percentage:** this measure shows the fraction of time for which of the time of arriving packets the buffer at the SS station is full. This measure implicitly reflects the percentage of time, where (video, voice and data) packets are lost due to buffer overflow [43].

Various research efforts have published thresholds for one or more of these four metrics, these have been summarized Table 2.1. The target PLR, delay and jitter thresholds for this project are reported in the final column. It reflect the recommended Y.1541 QoS that adopted by the IPTV and voices service providers.

Table 2.1: Performance metrics [41]

Metric	Video	Voices	Target
PLR **Average < 10-3** **Ideal < 10-5**	$4.9 \cdot 10^{-4}$ $< 10^{-3}$ $3.5 \cdot 10^{-3}$ $4.0 \cdot 10^{-3}$ $< 5.0 \cdot 10^{-2}$ $3.3 \cdot 10^{-2}$ (MS) $< 10^{-2}$	10^{-2} (fixed) $2.5 \cdot 10^{-2}$ $5.5 \cdot 10^{-2}$ (mobile)	10^{-3}
Delay (ms) **Average < 400** **Ideal < 150**	5 - 300	< 125 < 150 < 250 < 400	< 400
Jitter (ms) **Average < 60** **Ideal < 20**	< 40	< 0.4 < 31	< 50
Throughput **(kbps)**	16 – 4096 707 – 1264 34 - 5100 (Matrix)	384 (Dolby 5.1) 16-256 125 (stereo) 80 – 128 187 – 211 (Matrix)	221- 5311

In order to ensure the quality of service to the end-user, the expected delay (variation) and packet loss requirements can be fulfilled for specific traffic flows under network congestion conditions. For example, a telephony service have more stringent quality of service requirements then a more "general" Internet access service (browsing websites) [42]. In addition, latest developments in the area of video and audio encoding impose new challenges in the content of delivery systems. In this thesis, the DiffServ architecture is chosen because it is preferred over "Hard QoS" architecture [14]. Moreover it is perfectly applied to the Triple Play, as it satisfies a different QoS requirement which is described in the following subtopic.

- **QoS Management in DiffServ**

The creation of a wide use new high quality demanding services (VoIP, High Quality Video Streaming) and delivering them over already saturated core and the access network infrastructures have created the necessity for E2E QoS provisioning. Network providers use at their infrastructures several kinds of mechanisms and techniques for providing QoS. Most known and widely used technology is DiffServ. Differentiated Services are a computer networking architecture that specifies a simple, scalable, and coarse-grained mechanism for classifying, managing network traffic and providing QoS guarantees on modern IP networks [15].

2.5.3 Network Security Requirements

Security in general is important for services that carry some privacy sensitive information (like the content of a telephone conversation or transaction information of a home banking) or content that has a certain commercial value (like on demand movies). These services need some level of protection, such that only the intended recipient (s) can receive and use the transmitted information. This type of service is usually implemented on a "higher level" than by the network itself, often this is done by the (service related) client and server (like a software player or STB for the on demand video). The network encryption function is often used by the broadband wireless systems (like WLAN and Wireless Local Loop) to provide a network security that is "equivalent" to point-point broadband wire line infrastructures like the telephony network (as a signal sent over the air can be easily received by others, while a signal sent over the dedicated telephony cannot be received unless an explicit physical ‘tap’ is placed). The segregation of traffic function is often used to make sure that traffic of customers served by one service provider do not interfere with the traffic generated by customers of a different service provider and is especially important for “open broadband networks” which give access to multiple service providers [40].

2.5.4 Network Availability Requirements

Most current Internet based services have low availability requirements as they are often free of charge and do not perform "critical" services. However, this is changed now, since the existing analogue services like regular telephony and broadcast TV will Most current Internet based services have low availability requirements as they are often free of charge and do not perform "critical" services. However, this is changed now, since the existing analogue services like regular telephony and broadcast TV will be also offered over the Internet (VoIP, Digital TV over IP) via different types of broadband networks. For telephony this might even go as far that even after a power failure, the telephony service should remain active ("lifeline" functionality) as this functionality is offered by KPN for the regular telephony service (by remotely powering the telephone from the CO, which has an emergency power supply) [40].

2.6 Broadband Infrastructures and Their Support for Triple Play

As the broadband revolution continues, the ever increasing competition in the broadband service market is forcing broadband service suppliers to plan their strategies for delivering of "Triple Play" services, with voice, data and video provided by a single connection. In this thesis the latest developments in the leading broadband access technologies are reviewed and the ability of those technologies to meet the future requirements of the broadband consumer is assessed. Figure 2.6 shows several types of broadband access technologies that are used to support Triple Play services bandwidth [41].

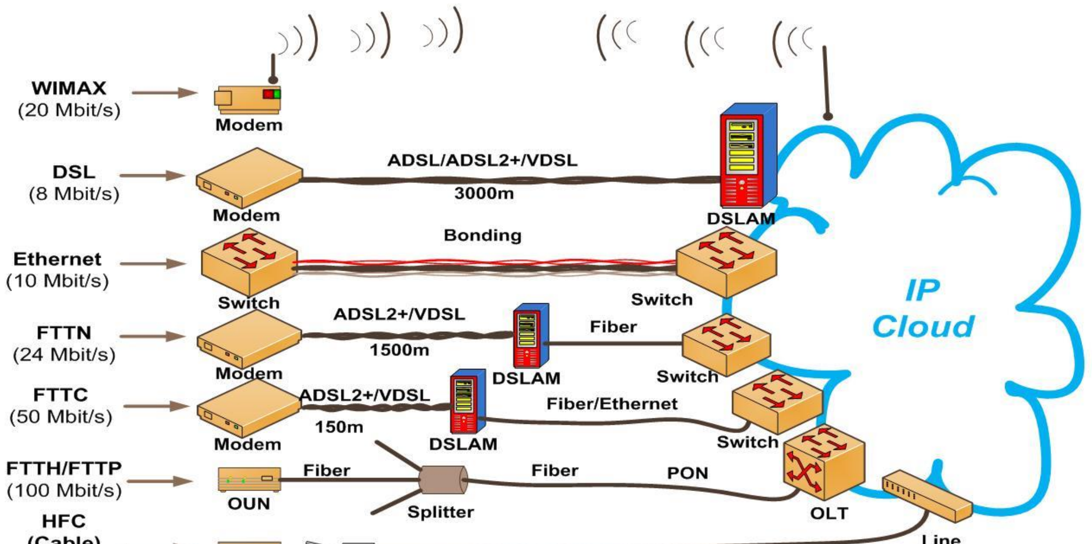


Figure 2.6: Broadband access technologies. Triple Play supports bandwidth hungry applications that require bandwidth of many Mbit/s [28]

In general broadband solutions can be classified into two groups: fixed line technologies and wireless technologies. The fixed line solutions communicate via a physical network that provides a direct "wired" connection between customer and service supplier. The best example is the POTS, where the customer is physically connected to the operator by a pair of twisted copper cables. Wireless solutions use radio or microwave frequencies to provide a connection between the customer and the network operator's; mobile phone connectivity is a prime example. These two types are explained in the following subsection.

2.6.1 Fixed Line Technologies

Fixed line broadband technologies rely on a direct physical connection from services supplier to subscriber's residence or business. Many broadband technologies such as cable modem, xDSL and broadband power line have evolved to use an existing form of subscriber connection as the medium for communication. Cable modem systems use existing hybrid fiber-coax cable TV networks. xDSL systems use the twisted copper pair traditionally used for voice services by the POTSIn general, all three aforementioned technologies strive to avoid any upgrades to the existing network due to the inherent implications of capital expenditure. By contrast, FTTH or FTTC networks require the installation of a new (fiber) link from the local exchange (CO) directly to or closer to the subscriber. Consequently, although fiber is known to offer the ultimate in broadband bandwidth capability, the installation costs of such networks have, up until recently, been prohibitively high. The fixed line technologies evaluated of wire include: Hybrid Fiber Coax (Cable TV & Cable Modems), xDSL, BPL and Fiber to the Home/Curb [43].

However, in this research, the term 'wire' applies only few technologies which are designed for access to Triple Play networks requirement, such as xDSL and FTTx that are described in the following subtopics.

A. Digital Subscriber Line (xDSL)

DSL is a very high-speed connection that uses the same wires of a regular telephone line. DSL allows making regular phone calls and surfing the Internet simultaneously. It is a point-to-point medium that manages to squeeze more information through a standard copper wire. However, the capacity and moreover the availability of DSL depends on the length of the local loop. DSL is a distance sensitive technology, user's distance from the closest CO should be less than 18000 ft, see Figure 2.7.

The most important advantage of Triple Play services through DSL, the bandwidth of copper wires is capable of carrying more than the phone conversations. DSL exploits this property. In particular, Asymmetric DSL (ADSL) divides up the available frequencies in a line such that upstream and downstream data as well as voice can travel together in the same copper wire. Delivering video using DSL over twisted pair is a booming technology called IPTV. A device called STB is used by the subscriber to control and order video services,like video on demand. Internet data can be delivered using ATM or DOCSIS. Voice can be distributed in two ways; using the traditional POTS interface or using VoIP [13].

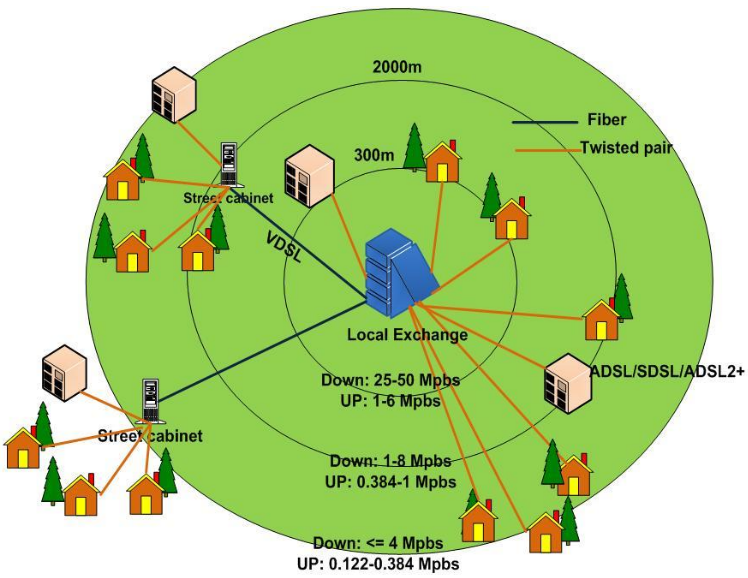


Figure 2.7: Network architectures for various forms of xDSL, note the xDSL bandwidth is dependent on distance from the local exchange/central office

B. Fiber to the x (FTTx)

Fiber to the *x* (FTT*x*) is a generic term for any broadband network architecture that use optical fiber to replace all or part of the usual metal local loop used for the last mile telecommunications. The generic term was initially a generalization for several configurations of fiber deployment (FTTN, FTTC, FTTB, FTTH...), all starting by FTT but differentiated by the last letter, which is substituted by *x* in the generalization [44].

Today the world is heading for employ new strategies to deliver Triple Play services to customers using fiber optical links called next generation Triple Play network. However, fiber which virtually has unlimited bandwidth, it is more economical than copper in terms of maintenance, upgrade, and manpower requirements. Optical networks have been seen by many as a promising solution

for the last mile problem. Figure 2.8 shows a fiber optical network that consists of one optical line terminal (OLT) located at the provider central office (head end) and multiple optical network units (ONUs or stations) at the customer side [40].

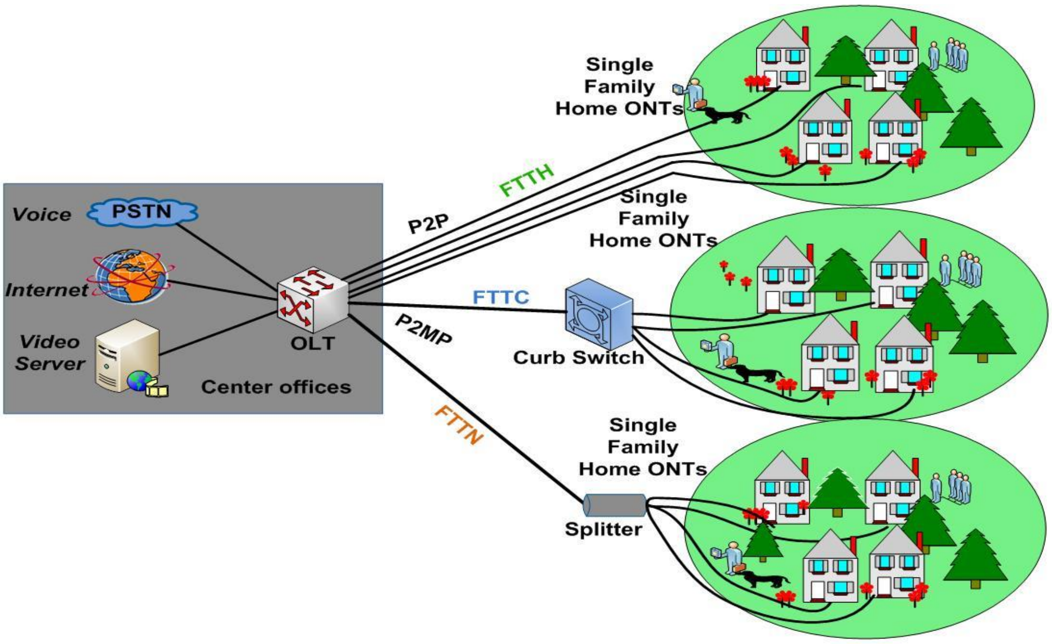


Figure 2.8: Triple Play over FTTx

2.6.2 Wireless Technologies

All communications based on a transmission media other than copper wire, optical fiber, or any other kind of cable, are wireless. This definition includes communications based on satellite microwave links, terrestrial radio links and free space optical communications [27]. The technologies evaluated of wireless include: Microwave links, MMDS, LMDS, FSO, Wi-Fi, WiMax, Satellite, and 3G.

However, in this research, the term 'wireless' applies only to a few microwave technologies designed for indoor and outdoor access to Triple Play networks. This kind of wireless communication is currently dominated by some technologies defined by the IEEE. The most important one is the IEEE 802.11 family (Wi-Fi) and the IEEE 802.16 family (WiMax), which is described in the following subtopics.

2.6.3 Wireless WAN-WiMax

The WiMax technology, based on IEEE 802.16–2004 standard [25], defines a fixed broadband wireless metropolitan area network. Mobile WiMax, based on IEEE 802.16e-2005, adds functions and features to the original standard to support mobility. The most current IEEE 802.16–2009 standard [24] is a revision of IEEE 802.16–2004. It also consolidates material from IEEE 802.16e–2005 and other previous 802.16 standards. Mobile WiMax has a target transmission range of up to 31 miles and a target data rate exceeding 100Mbps [45]. Compared to mobile WiMax, 3G data services provide a relatively low bandwidth and high price while WiFi suffers from limited transmission ranges and security issues.

WiMax is part of the evolution from voice-only wireless communications systems to ones that provide additional services like web browsing, streaming media, gaming, instant messaging, and other content. Being able to deliver a wide variety of services requires also a delivery system that is flexible and can efficiently allocate system resources. The 802.16 standard offers adjustable data rate to and from each user while maintaining the required QoS. Certain applications require higher error resilience and latency requirements that directly factor into the QoS However, WiMax is a serious contender for delivery of Triple Play services. With advanced antenna techniques, it offers data rates up to 70 Mbps and ranges up to 50 km, ensures secure delivery of content, and supports mobile users at vehicular speeds of up to approximately 100 km/hr. Figure 2.10 shows the system model of a WiMax deployment for Triple Play services over all IP backbone network. Triple Play content can either be classified at the IP layer and enter the WiMax system as Ethernet payload or each of the three types of content be individually provided for further classification and optimization, e.g., through Access Service Network (ASN) gateway as shown on the left part of Figure 2.9. [18]

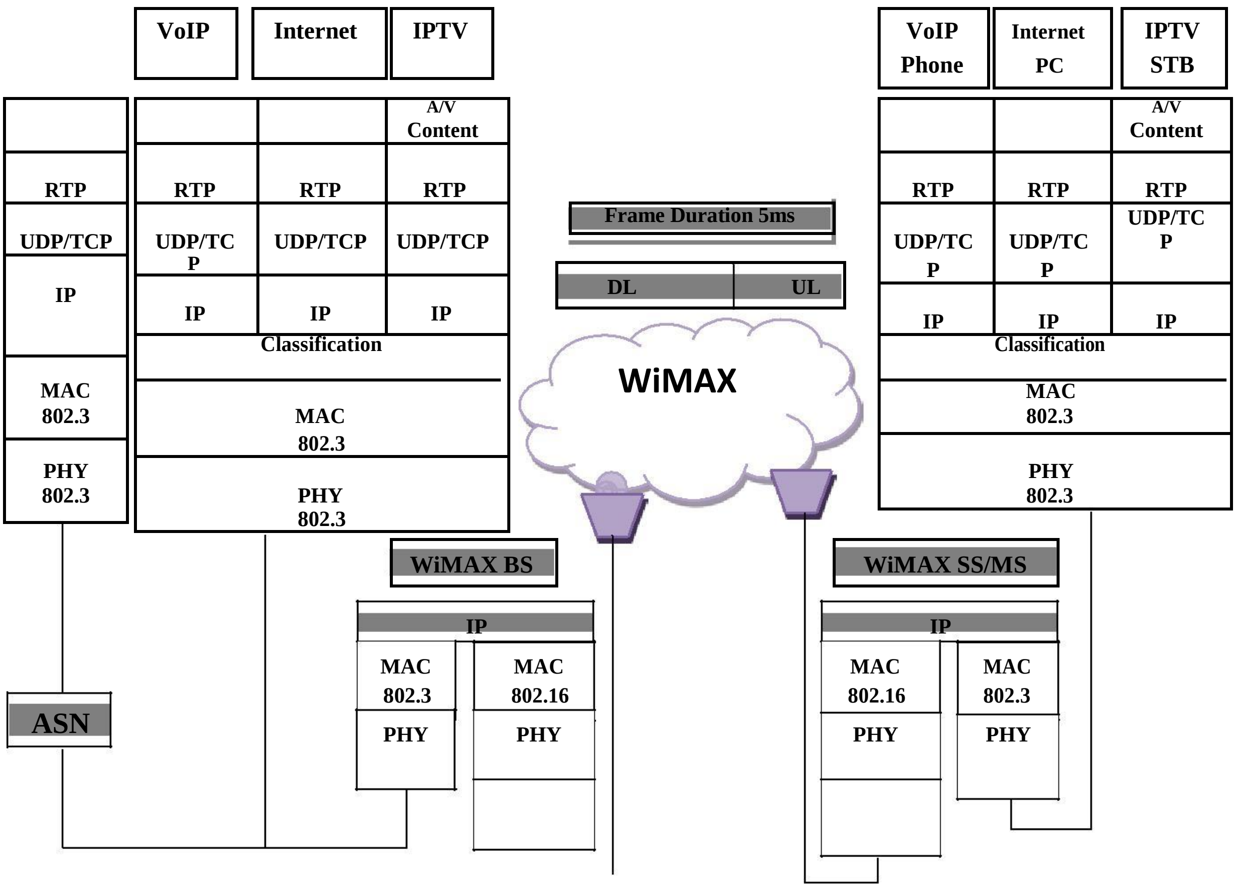


Figure 2.9: Triple Play services over WiMax system model [18]

CHAPTER THREE

TRIPLE PLAY SERVICES SIMULATED SCENARIO

3. 1 Introduction

After the detailed theory of Triple Play system presented in chapter two; it is the time to go through the details of the project simulation.

One of the main objectives of this thesis is to examine the performance of the broadband infrastructures and their support for Triple Play services under different separated scenarios. In addition, to compare the output results to study the effect of wire and wireless heterogeneous broadband connectivity under an IP network using OPNET Modeler 14.5. As discussed earlier, OPNET was found to be a better platform that can help to integrate a complete communication node with a different access broadband model. OPNET can provide a platform to create and test an analytic and practical video, voice and Internet model. It can also provide the ability to integrate the model into a different environment [42]. A clarification to the network architecture, simulation strategy, modeled topologies, examined parameters and type of used frequency and bandwidth are all illustrated in the following sections. The details results of each simulation project are then shown in the next chapter.

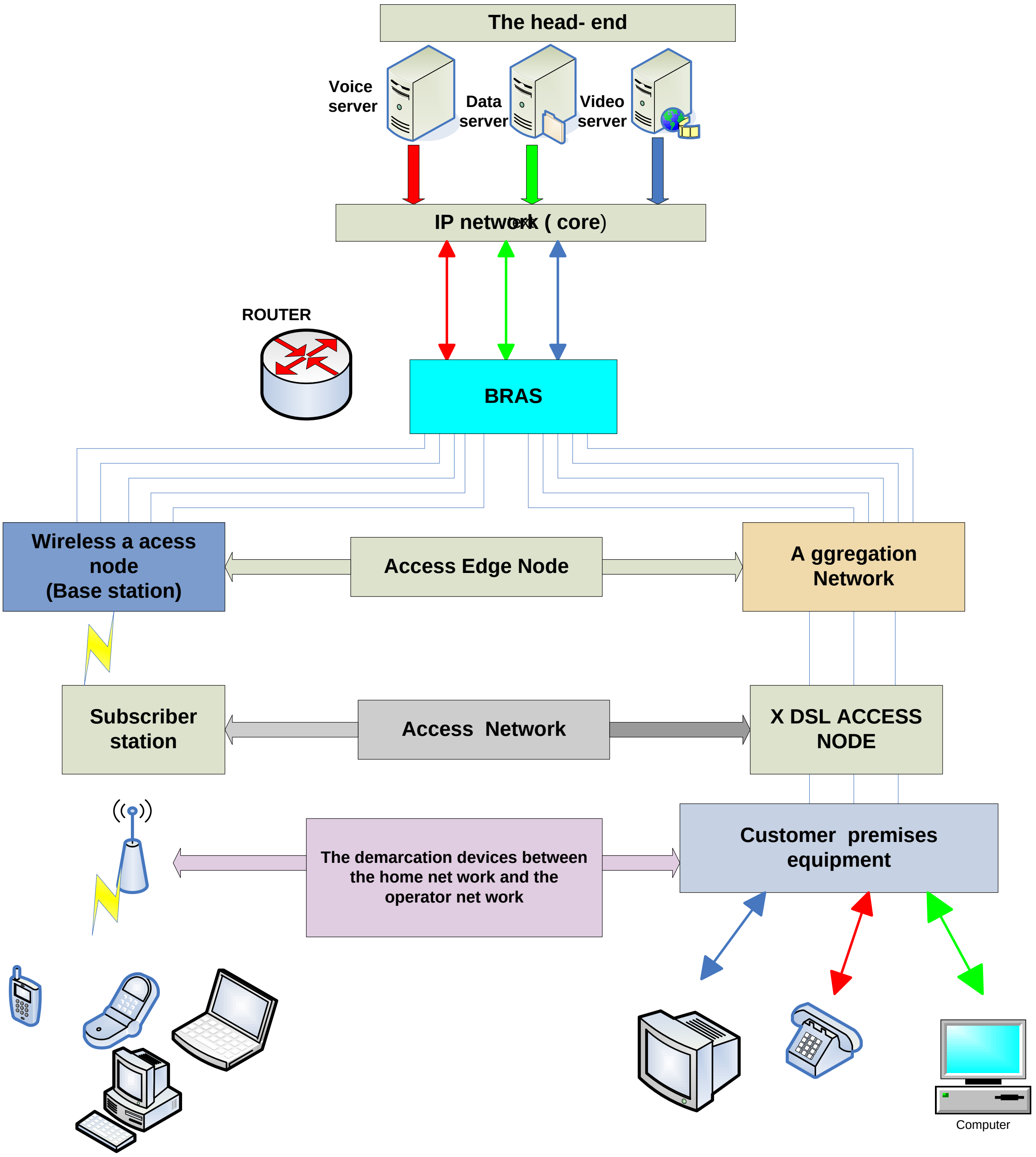
3.2 Network Architecture

A key challenge is to build an infrastructure that is optimized for today's offering and that can scale for future subscriber and service growth. The proposal in this project is to decompose delivery architecture for the Triple Play system to perform the essential functions of Triple Play network including non-stop service delivery, assured & optimized IP video delivery, service quality: assurance, and congestion avoidance. Table 3.1 lists the proposed subsystem of Triple Play system architecture. While Figure 3.1 shows the suggested system architecture and how the next generation network foundation that; leverages access, optics, carrier Ethernet and IP technologies, and couples them with robust management and capabilities to deliver a seamless user experience .

Table 3.1: The suggested subsystems of Triple Play system architecture

Subsystem	Function
The head-end	One or two main data centers house that consist the infrastructure servers, , and log servers. Often collocated at these same data applications. These servers are connected to switches by use 100BASE-T or 1000BASE-T electrical connections, which are then fed into the core via optical fiber links.
The core network	The core network, which is a high capacity fiber backbone network, uses ATM or IP/Ethernet over SONET.
Broadband Remote Access Server (BRAS)	The BRAS sits at the core of an ISP's network, which is a business or organization that provides to customers access to the Internet and related services. The specific tasks of a BRAS include: • Aggregation point: the BRAS aggregate the output from multiple DSLAMs in the access network. • Router: the BRAS routes traffic into an ISP's backbone network. • Session termination: the BRAS provide the logical termination of PPP sessions. These may be PPPoE or PPPoA encapsulated sessions. • Subscriber management functions: the BRAS provide the interface to (AAA) systems. The BRAS is also responsible for assigning session parameters such as IP addresses to the clients. The BRAS is the first IP hop from the client to the Internet.

	• Policy management and QoS: at the BRAS, an ISP can insert policy management and IP QoS.
Access Edge Node	The AEN is also responsible for enabling the equal access concept. This is possible due to the creation of service bindings, which can be created by any service provider requesting resources in the network. The role of the AEN is to manage and control the service bindings by forwarding the frames from the customer premises equipment's to a specific service provider using the aggregation network. In this sense, the AEN can be seen as a bridge between environment
Access Network	The Access Node is the element where individual physical connections to the user domain are attached and maintained. It provides a bridge between the Ethernet switching domain of the access network and the end-user’s network domain. Therefore, the Access Node acts as a gateway between the physical technology used by the user devices to connect to the Access Node on the first mile (Wi-Fi, xDSL, fiber, WiMAX) and the Ethernet technology used by the aggregation network. It is the first point in the access network where traffic coming from multiple user ports is aggregated towards the AEN. The Access Node has the role to forward and regulate traffic to/from the user ports and to/from the Access Edge Node through the Ethernet aggregation network. It also has the responsibility to perform traffic marking (802.1P field) on the upstream of service bindings based on defined policy rules, and to perform traffic policing on the upstream of service bindings against defined policy rules.



The head- end
Voice server
Data server
Video server
IP network (core)
ROUTER
BRAS
Wireless a acess node (Base station)
Access Edge Node
A ggregation Network
Subscriber station
Access Network
X DSL ACCESS NODE
The demarcation devices between the home net work and the operator net work
Customer premises equipment
Computer

Figure 3.1 Architecture for suggested Triple play system

3.3 Simulation Strategy

The outlines of simulated projects are as follows:

- Examining and delivering Triple Play services using ADSL technology. This examines the capacity, in addition, to the availability of ADSL links specifically for video over twisted pair, which is a booming technology called IPTV. To compare between two compress methods, video service is streamed to the subscriber using the MPEG-2 (and soon MPEG-4) format.
- Examining and delivering Triple Play services using WiMax technology. The best needed bandwidth for a carrier frequency of 3.5GHz is tested. The pathloss for urban, suburban and rural is calculated, as well the multipath with pathloss for ITU channel model of pedestrian in different environments.

3.4 Simulation Parameters

Each object in the Triple Play network models project has a specific set of parameters. Some configurable attribute parameters are set according to their default values other are changed depending on the scenario and required to examine. In general, parameters that remain constant at all scenarios are classified into:

3.4.1 Application Parameter

Application attribute definition is used to specify/choose the required application among the available applications, such as FTP, HTTP, Video, Voices, and Print. It is possible to create a name and give a relevant description in creating a new application. In this thesis, the following applications are defined:

- **Voice Application:** Voice service is simulated by setting up a VoIP application characteristic as shown in Table 3.2 between residential users and an Ethernet server. Concerning the simulation of VoIP service over the broadband connection, and G.711 encoded data streams are utilized. G.711 is the international standard for encoding telephone audio on a 64 kbps channel [14].

Table 3.2: Residential VoIP application characteristics

Encoder Scheme	Voice Frame Per Packet	DSCP Value	Compression Delay(s)	Decompression Delay(s)
G.711	1	EF	0.02	0.02

- **HTTP Application:** Web browsing application is defined with different characteristics as shown in Table 3.3 for residential users. Residential users may browse the Internet for images, socialize in face book, search for information, read an electronic version of a newspaper, etc...

Table 3.3: Residential HTTP application characteristics

Http Specification	Object Size (Kbyte)	DSCP Value
Http 1.1	Constant(1) Media Image (0.5-2) Large Image (2-10)	BE

- **FTP Application (file sharing /downloading):** In order to simulate users downloading and sharing files, an FTP application is setup with the characteristics shown in Table 3.4. Users request to download a file of 5MB in size in random time windows from a server, which is located at the ISP's (backbone network).

Table 3.4: Residential FTP application characteristics

Download	Size File (MB)	DSCP Value
100%	5	AF13

- **IPTV Application:** IPTV is a platform that delivers Internet TV. It exploits the architecture of an IP network instead of radio frequency broadcast, satellite signal and cable television. To simulate an IPTV service, a video traffic is applied, which is generated by the OPNET Modeler video conferencing application using a CBR configuration, as shown in Table 3.5.

Table 3.5: IPTV application characteristics

Frame Rate	Incoming Frame Size (Byte)	Outgoing Frame Size (Byte)	DSCP Value
15 Fps	17280	17280	AF33

- **VoD Application:** VoD service is much more difficult to engineer into network given the uncertainty of when and how many users would request this service simultaneously. Technically, when customer selects movie, a point-to-point unicast connection is set up between the customer's decoder (STB or PC) and the delivering streaming server. To simulate VoD service, the video trace (10-minute MPEG-2 movie clip of Terminator 2) or (2-hour MPEG-4) provided as shown in Table 3.6 [45].

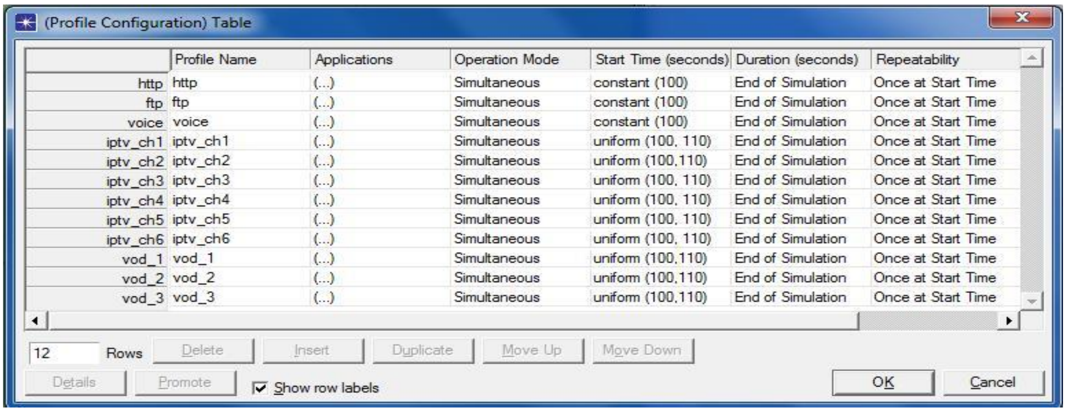
Table 3.6: VoD application characteristics

Parameters	**T2**	**Matrix III**
Resolution	**1280x720**	**352x288**
Codec	MPEG-2	MPEG-4 Part 2
Frame Compression Ratio	58.001	47.682
Minimum Frame Size (bytes)	627	8
Maximum Frame Size (bytes)	127036	36450
Mean Frame Size (Bytes)	23833.792	3189.068
Display Pattern	IBBPBBPBBPBB	IBBPBBPBBPBB
Transmission Pattern	IPBBPBBPBBIB	IPBBPBBPBBIB
Group of Picture Size	12	12
Frame Rate (frames/sec)	**30**	**25**
Number of Frames	324,000	180,000
Peak Rate (Mbps)	30.488	7.290
Mean Rate (Mbps)	**5.720**	**0.637**
DCSP	**AF33**	**AF33**

3.4.2 Profile Parameter

Profile attribute definition is used to create user profiles; these profiles can be specified on different nodes in the designed network to generate application traffic.

Figure 3.2 shows the profiles that are used in this thesis for the simulated model. These profiles are: ftp profile, http profile, voip profile, iptv_ch1 profile, iptv_ch2 profile, iptv_ch3 profile, iptv_ch4 profile, iptv_ch5 profile, iptv_ch6 profile, vod_1 profile, vod_2 profile and vod_3 profile.

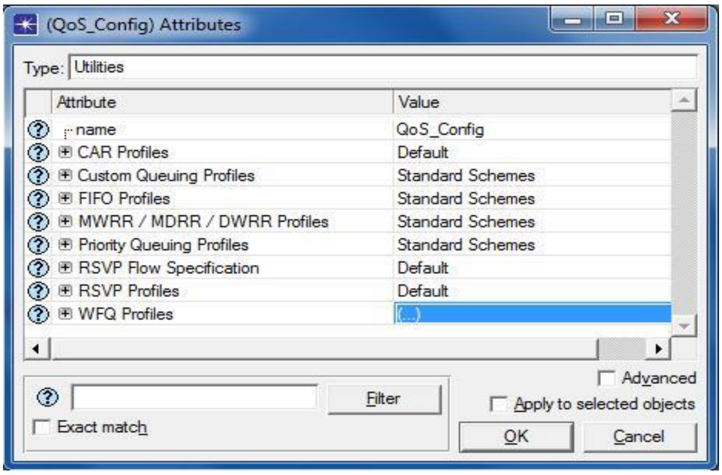


	Profile Name	Applications	Operation Mode	Start Time (seconds)	Duration (seconds)	Repeatability
http	http	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
ftp	ftp	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
voice	voice	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
iptv_ch1	iptv_ch1	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch2	iptv_ch2	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
iptv_ch3	iptv_ch3	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch4	iptv_ch4	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch5	iptv_ch5	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch6	iptv_ch6	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
vod_1	vod_1	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
vod_2	vod_2	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
vod_3	vod_3	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time

Figure 3.2: The profile attribute configuration

3.4.3 IP QoS Parameter

The QoS attribute configuration object defines the following technologies: CAR, FIFO, WFQ, CQ, and PQ, see Figure 3.3. Each technology contains a table, such that each row represents a single queue. Each queue has a number of parameters, e.g. queue size, classification scheme, RED parameters, etc...



Attribute	Value
name	QoS_Config
CAR Profiles	Default
Custom Queuing Profiles	Standard Schemes
FIFO Profiles	Standard Schemes
MWRR / MDRR / DWRR Profiles	Standard Schemes
Priority Queuing Profiles	Standard Schemes
RSVP Flow Specification	Default
RSVP Profiles	Default
WFQ Profiles	(...)

Figure 3.3: The QoS technologies

Network infrastructure consists of: a service provider, core, and access networks. Accordingly, the IP DiffServ mechanism needs to be implemented at each network node and application. This is discussed in the following subtopics.

- **Configuration QoS Classes with Different Applications:** There is large number of applications that need a level of assurance from the network. However, it is difficult to define a specific level for each. For that reason, several categories called "QoS classes" are exist, which classify applications by giving higher priority to the delay-critical voice and video packets over other data packets, such as email messages, webpages, etc.. The method of classifying applications is done related to the applications attributes. Where, "Application Configuration" attribute defines for each application the types of services. In this thesis DSCP with priority of HTTP/Best-effort, FTP/AF13, IPTV/AF33, VoIP/EF and VoD/AF43 are defined ,
- **Configuration QoS in fixed and mobile node (Server, Router, and Workstation):**

  QoS specification parameters are available on a per interface basis on every router or workstation node. The sub-attribute called "Interface Information" in the "IP QoS Parameters" attribute is used to specify this information such as QoS scheme, Buffer size; Maximum Reserved Bandwidth, etc. see Figure 3.4

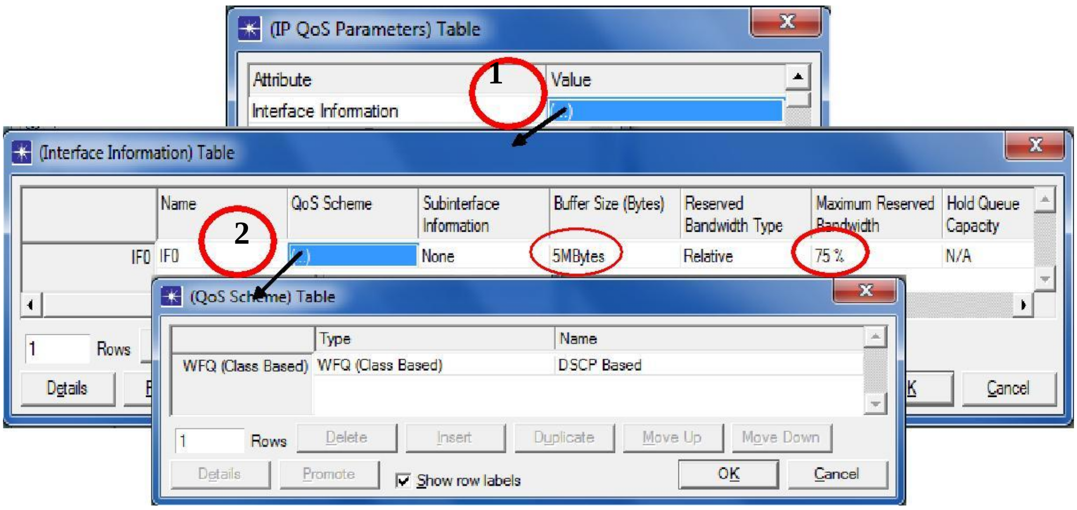


Figure 3.4: Configuration QoS in fixed and mobile node

3.4.4 Server Parameters

In each server, supported services are based on the user profiles that may support FTP, HTTP, VoIP, Video, etc..., on the client.

3.4.5 WiMax Configuration Parameter

In WiMax configuration, the global configuration object is used to configure parameters such as PHY profiles, efficiency mode and MAC service class definitions. Three efficiency modes are used: "Framing Module Enabled", "Physical Layer Enabled" and "Mobility and Ranging Enabled". When attribute is set to "Framing Module Enabled", the simulation processes a frame-by-frame modeling of allocations on the UL and DL (the scheduler is invoked periodically - once per frame - when the Map's for each frame are built), this leads to more accurate delay results. When the efficiency mode is set to "Physical Layer Enabled", the simulation accounts for physical layer effects and these modes can be used for fixed work station. For mobile node, the "Mobility and Ranging Enabled" mode is set and the simulation accounts for mobility and ranging effects (physical layer effects and frame-by-frame modeling is also performed).

3.4.6 Nodes Parameters

The network parameters are set for all nodes, such as work stations, with client server applications run over TCP/IP and UDP/IP.

3.5 Simulation Model Design

The following sections describe the various aspects of model design and its configuration.

3.5.1 Simulation Triple Play Services over ADSL Technology

This simulation gives a high-level overview of a generalized network topology, protocols and devices, that are used in the design of Triple Play services over ADSL technology with data rate downstream 12Mpbs and upstream 1.3Mpbs (ITU G.992.1 Annex A, ADSL over POTS, 2001) [26]. In the current project, the performance of different service classes provided in the wire network have been evaluated by measuring several QoS parameters including average jitter, packet loss and end-to-end delay. The networks are tested for the cases when approximately the availability of network bandwidth is 50% (services are full at each home at a time, 50% traffic in core links), 30% (services are full at each home at a time, 70% traffic in core links) and 30% (services are not full at each home at a time, 70% traffic in core links).

1. Network Topology for ADSL

The general network is for geographical areas and sub network is assigned for any region, The major components are: server head end, external Internet peers, IP core, (BRAS) edge, Ethernet aggregation network, DSLAMs, local loop, and customer household. Those are explained in the following subtopics.

A Backbone Subnet: Figure 3.5 shows the "Backbone" subnet which is designed with 4 servers configured to stream stored audio and video contents, HTTP and FTP. It contains a 100Mbps IP network and access routers for both IP multicast traffic load (R3, RP, and R4) and IP unicast traffic load (DSR). These routers are connected to switches (Source and PPP), which divide traffic into VLAN. Then, these routers are connected to the BRAS router at the aggregation subnet through a PPP_SONET_OC24 with data rate (1244.16 Mbps) WAN link. The approximate distance between the backbone subnet and aggregation subnet is 5.44km, which corresponds to approximately 18.133ms propagation delay. The multicast and unicast techniques are configured with VLAN parameter to get efficient Triple Play services;

these are explained in the following paragraphs

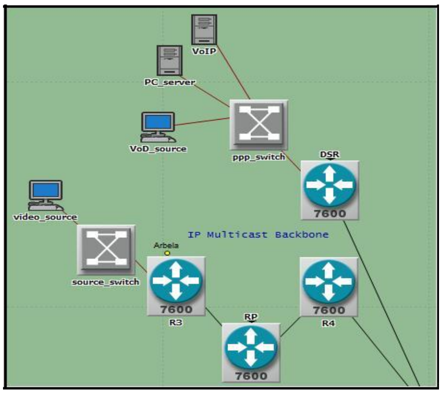


Figure 3.5: Backbone subnet inside ADSL network topology

First used IP Multicast technology, To support the IPTV services, multicast technology and protocols must be presented in the core, edge, and access layers of an IP ADSL Triple Play network. This can be achieved by, enabling multicast on routers. It can be configured using the attribute " IP multicast -> IP multicast parameter -> multicast routing". Then, the multicast traffic is forwarded, using IP multicast group address. A multicast group address is a single IP address taken from a reserved range (224.0.0.0/4 for IPv4, FF00: /8 for IPv6) to uniquely identify a group of hosts desiring to receive certain traffic [64]. The sender (video_source server, see figure 3.9) uses IP multicast address to set up a video conference (IPTV) session with all receivers. A router can listen to a particular group by setting the group address in the IGMP static membership information table. In this project, the specified group address is added from 224.0.6.1 to 224.0.6.6 for six IPTV channel under "IGMP Parameters-> Interface Information-> <interface> <Membership Groups", see Figure 3.6.

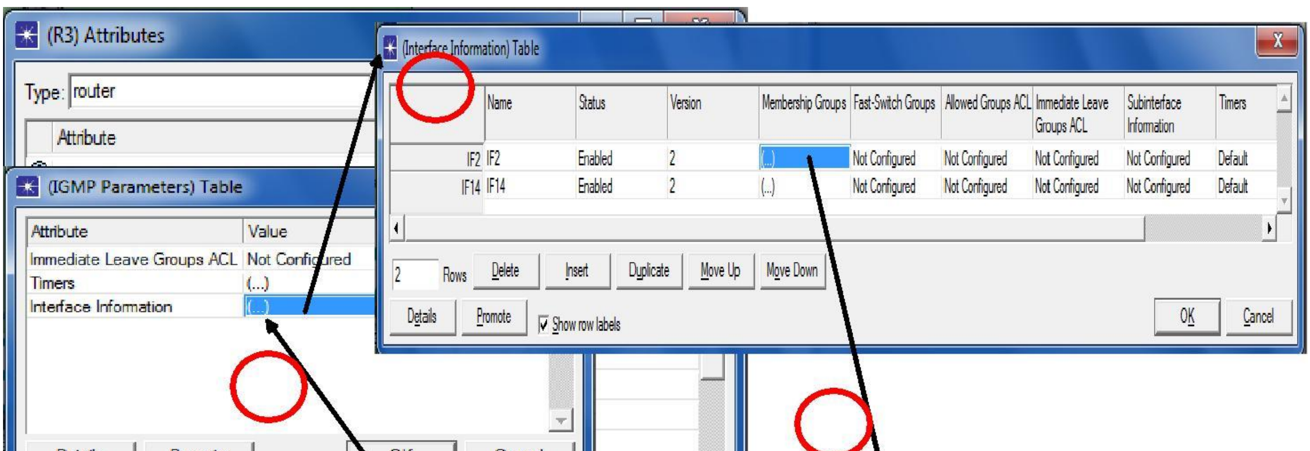


Figure 3.6: Configuration IP address for multicast group on routers

PIM-SM is a protocol that distributes the routing information. This protocol is called *protocol-independent* since it is not dependent on any particular unicast routing protocol for topology discovery.]. In each router, PIM_SM is a configured by using the attribute" IP multicast ->IP multicast parameters ->" Interface Information-><Routing Protocol>" as shown in Figure 3.7.

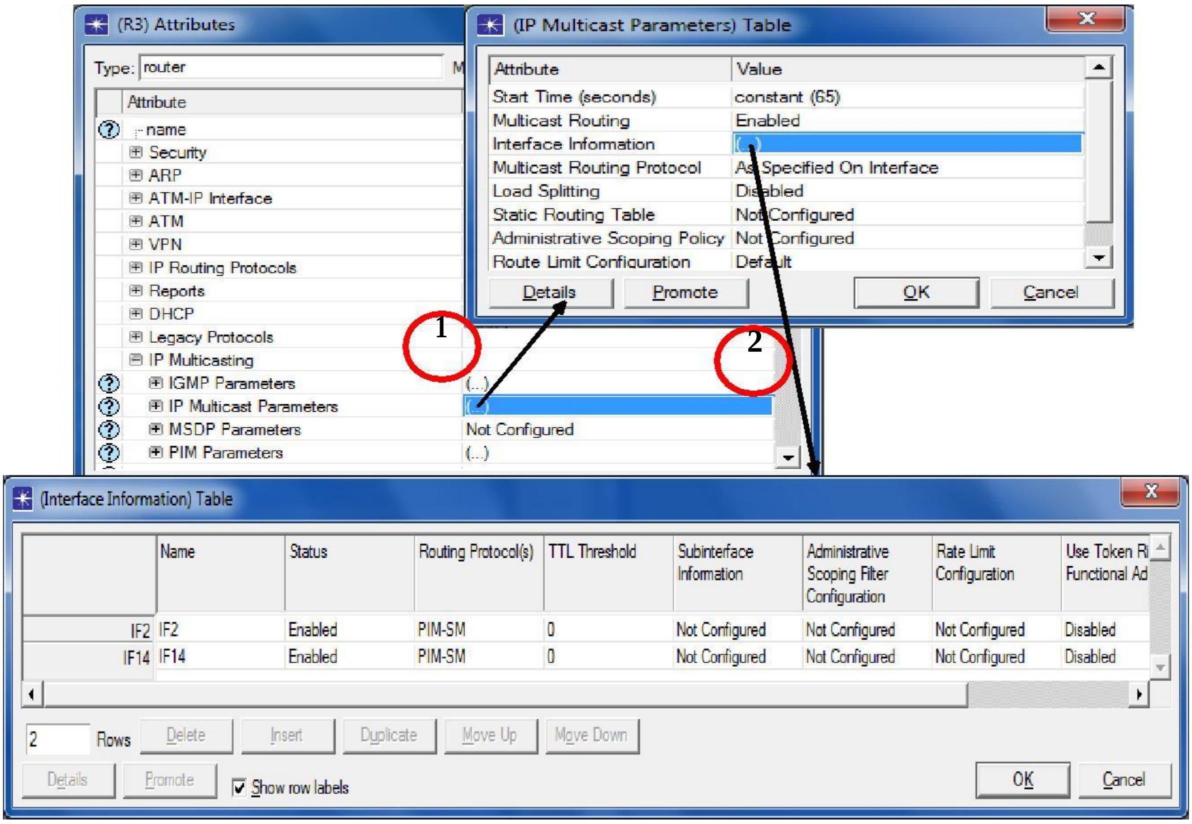


Figure 3.7: Configuration PIM-SM multicast protocol on routers

The Rendezvous Point (RP) router, see Figure 3.8, which is responsible for knowing all multicast sources must be added, and a complicated shared must be built to the RPs. PIM-SM creates a single multicast tree rooted at a core RP for all multicast group members within a domain. Sources send their data to the RP which forwards the data down the shared tree to the receivers. A receiver joins a multicast group via IGMPv1 or IGMPv2 and receives data sent by the multicast group source via the shared tree. RP is configured either with Auto-RP Bootstrap or Static protocol. In this project static RP under "PIM-SM Parameters" is used, it is shown in Figure 3.7

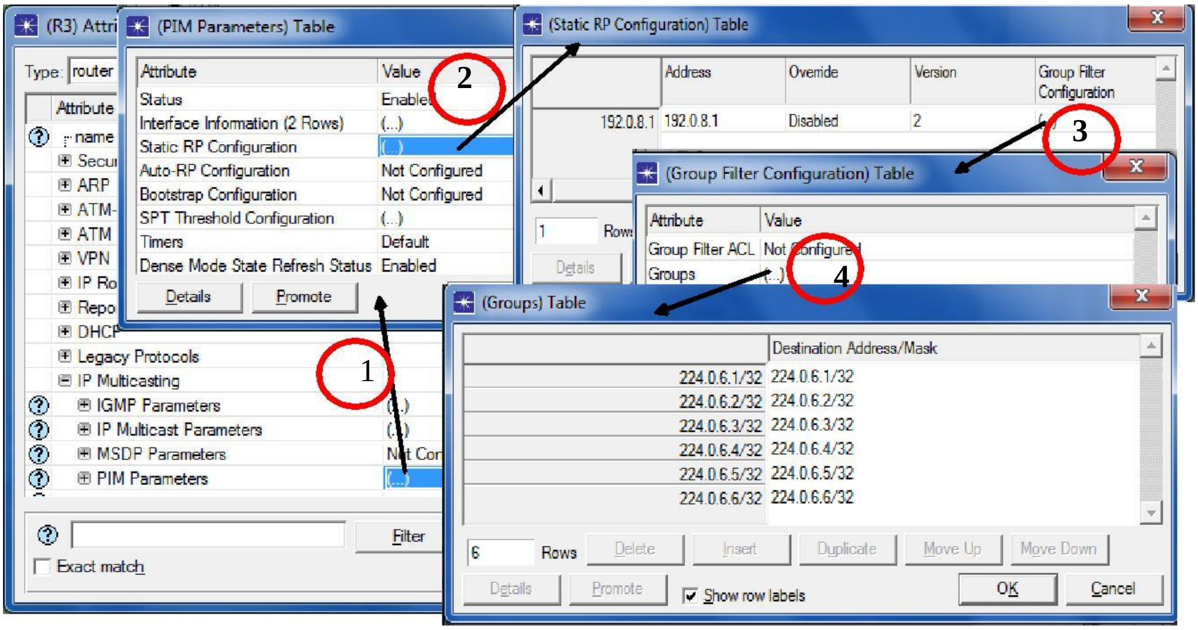


Figure 3.8: Configuration static RP distribution trees

In the edge where ADSL household presented IP, multicasting can be enabled or disabled for each IP interface. This can be configured using the attribute "IP Host Parameters->Multicast Mode". On IPTV channel (from CH1 to CH6), application can joins an IP group address in T.V by specifying the group address in the attribute "Application: Multicasting Specification" that are shown in Figure 3.9, in which 2 IPTV channel support CH1 and CH2.

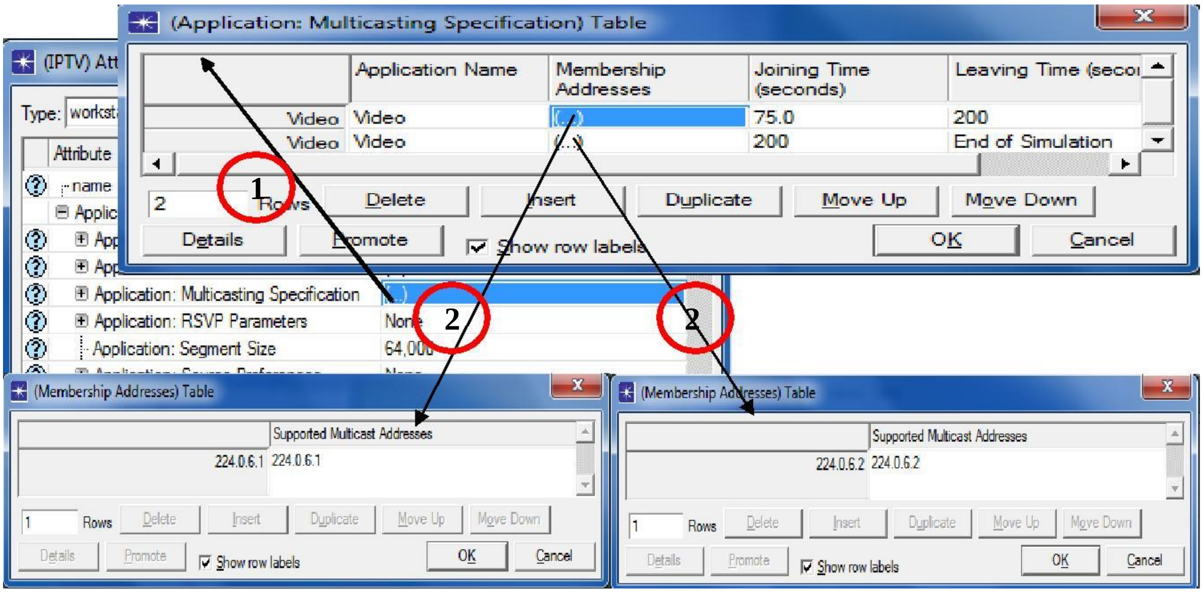


Figure 3.9: Joining a group for IPTV channel at T.V in Home_network

Second using IP Unicast technology, the router DSR, see Figure 3.9 in the backbone subnet is used in the unicast mode for delivery unicast traffic (VoIP, Internet & VoD) to the BRAS at the aggregation subnet.

Third using VLAN per Service technology, a separate VLAN is used for each service. This architecture is frequently used when IPTV service is introduced onto a broadband network. Putting new services into a different VLAN lowers the risk of disrupting the existing service. This approach is shown in Figure 3.10 [9].

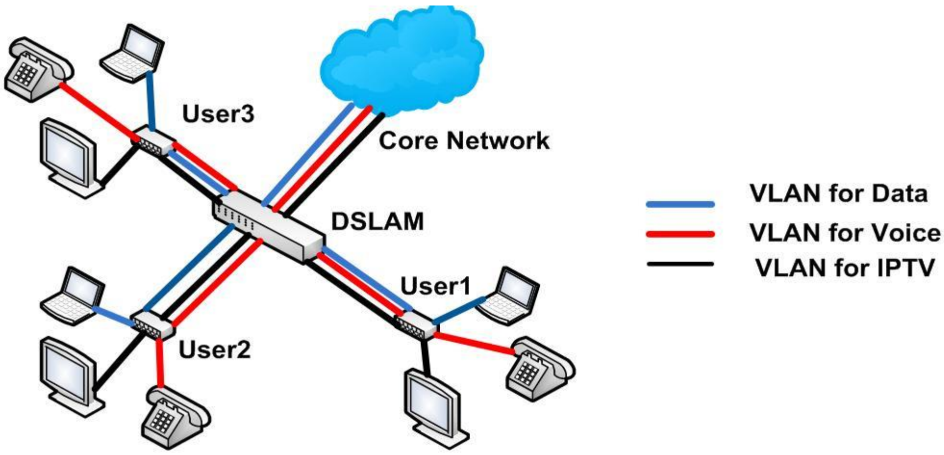


Figure 3.10: VLAN per service [9]

One of the easiest ways of delivering a triple- or multi-play service is to allocate a single VLAN (VLAN7 is used with packet traffic) with multiple modes and allow all customers to share that VLAN, as shown in Figures 3.11 and 3.12. This model can be used to study the basic performance of telecommunication network architecture for multi-service. It can be easily extended with VLANs (VLAN per service).

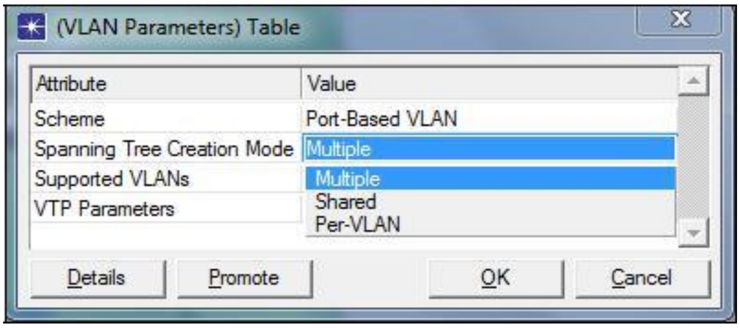


Figure 3.11: Configuration single VLAN with multiple modes

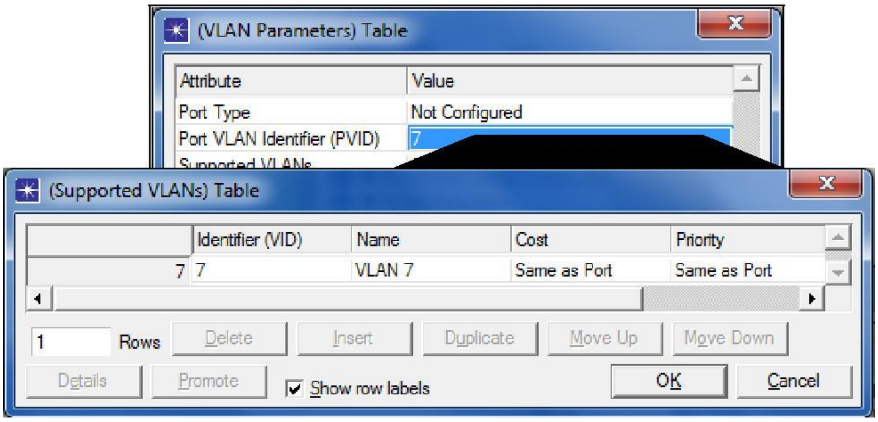


Figure 3.12: Configurations VLAN on switch

B. Aggregation_network Subnet: The aggregation subnet, receives all traffic from backbone network using PPP_SONET_OC24 with data rate of 1244.16 Mpbs. Then, delivers it to access network at a customer side using "Metro Ethernet Network", this subnet has several components These components are discussed in the following subtopics.

1. BRAS: The "BRAS" is a Broadband Remote Access Server router that forwards packets between the core and customer. It is a complex router that implements dynamic per-subscriber IP policies, QoS profiles, rate limiters, packet manipulation, address assignment, session termination, and forwarding.

2.Center Offices (CO): The CO router relays the data to the access network, which consists of DSLAMs and DLCs. CO router and remote DSLAMs are supplied with "Gigabit Ethernet" links. The capacity of this link was set to 1Gbps. A switch for "Metro Ethernet Network" (AGS1_1) is used to separate a traffic received from BRAS and CO router using VLAN technology.

C. Access Network (Region Subnet): It simulates end-to-end communications between residential customers and backbone subnet network. Figure 3.13 shows subnets allocated inside region, subnet is used for the coverage of three areas with 1Km. Each has several components as shown in the same figure and discussed in the following subtopics.

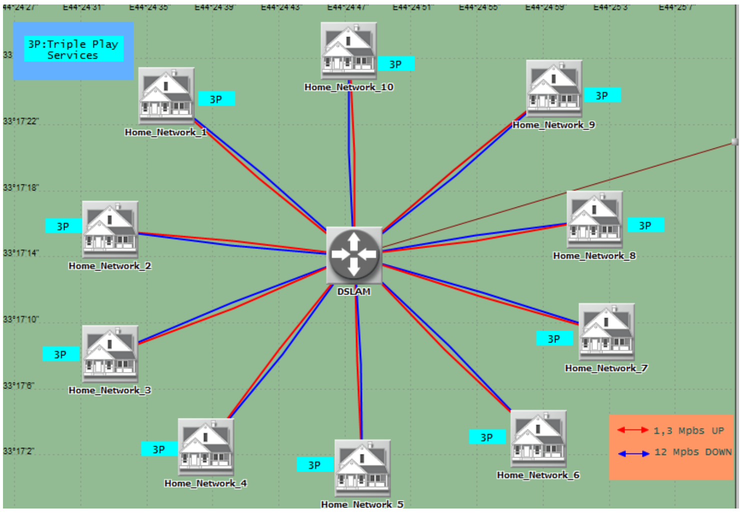


Figure 3.13: Access network inside region subnet in ADSL network topology

- **DSLAM**: DSLAMs terminate subscriber copper local loops and provide a DSL modulation service to the CPE. The DSLAM is Ethernet capability; it forwards traffic toward the BRAS as Ethernet frames with at least one level of VLAN tagging. Ethernet-capable DSLAMs are an important aspect of most of the designs in this thesis. The DSL forum uses the terms Level 1, Level 2, and Level 3 to describe how many upstream layers of ANs a DSLAM are connected. When Level 2 DSLAMs are used, a capacity is limited only to tens subscribers, this is due to their smaller sizes.

- **The customer household:** The customer household is the network's demarcation point between the service provider and customer. Services are delivered to the demarcation point, which may be a piece of equipment managed by the service provider. Or, in a simpler architecture, customer may provide his or her own hardware. In both cases, they are called the RG, which is the CPE. Most CPE are multi-play networks involves complex pieces of equipment provided by the service provider that integrate a DSL modem and router. Most providers have only a single STB today, which is connected to the RG. STBs take VoD and IPTV traffic coming from the network and send it to a TV connection via composite, SCART, HDMI, or RF outputs. STBs are controlled with a remote control, which is used to switch channels. To let other devices connect to the broadband network, RG router has several Ethernet ports to connect PCs, telephone and other devices in the home as shown in Figure 3.14.

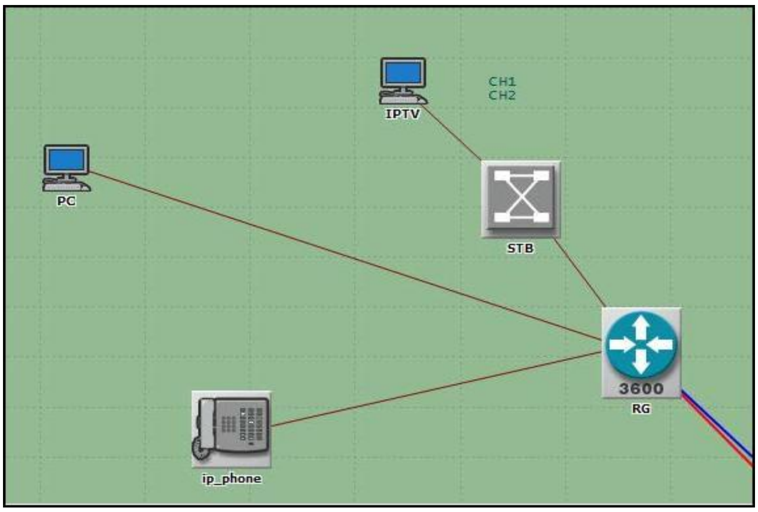


Figure 3.14: RG, STB, PC, IP Phone device inside each Home_network subnet

Figure 3.14: RG, STB, PC, IP Phone device inside each Home_network subnet

2. QoS Management in ADSL Simulation Scenarios

Looking at the ADSL network topology in the simulated architecture, the access links between IP backbone and BRAS, and the links between DSLAMs and BRAS router can become congested. Furthermore, the combination of video, data, and voice on a single medium requires more sophisticated QoS features. The QoS simulated architecture is based on the IETF Differentiated Services (DiffServ) architecture which is described in RFC 2475; this is used in project simulation. Therefore, the DiffServ domain must be included in the IP backbone, BRAS, CO and subscriber's side (configured in the same way as described in the subsection 3.4.3).

Finally, provide QoS for multicast traffic: this thesis proposes a solution to support real-time multicast traffic with QoS constraints over (DiffServ) IP networks. The proposed solution allows multicast users to dynamically join and leave multicast tree. Moreover, this allows a multicast user who has already negotiated the best-effort session to upgrade to a QoS-enabled session.

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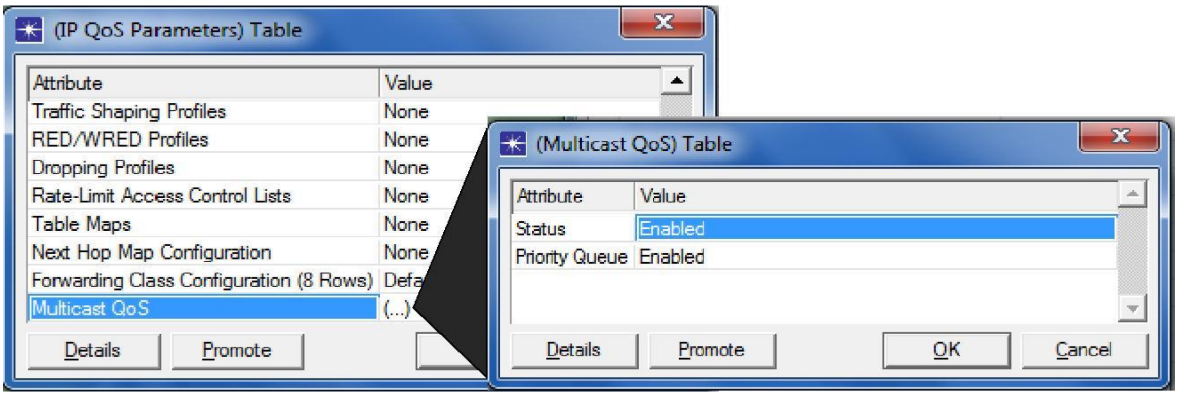


Figure 3.15: Multicast QoS configured

3. Simulation Scenarios for ADSL:

The above network topology is used to provide five example scenarios as pictured in Table 3.7. The execution process is for 500 seconds for each scenario to enable the OPNET simulator to simulate all the underlying layers. The profiles and outlining Modeler specific parameters are configured in section 3.4. The QoS guidelines and the objective performance recommendations in these scenarios are presented according to the context described in subsection (3.5.1, point 2). The result and analysis are given in chapter four.

Table 3.7: Simulation Scenarios for ADSL Network Topology

#	Scenario Name	Saved	Results	Sim Duration	Time Units
1	30 home users with 100 availability_ MPEG4	saved	out of date	500	second(s)
2	30 home users with100 availability_ MPEG2	saved	out of date	500	second(s)
3	30 home users with 50 AvL_ Full Services	saved	out of date	500	second(s)
4	30 home users with 30 AvL_ Full Services	saved	out of date	500	second(s)
5	30 home users with 30 AvL_ NO Full Services	saved	up to date	500	second(s)

Scenarios **# 1, 2**: In these scenarios the networks are tested with a configuration for both video streams: MPEG-2 1280x720 at 30 fps that has a high resolution format and encoding frame rate, and MPEG-4 352x288 at 25 fps. The mean and peak rates for both video streams shown in Table 3.6 are more realistic for modeling access network video streaming. In addition, a comparison for video delivery services using IP multicast and IP unicast is performed. In these scenarios, the total number of customer household is 30 houses; all have received full Triple Play services simultaneously at 100% bandwidth availability with guaranteed QoS.

Scenarios # **3, 4, 5:** The objective of these scenarios is to compare the performance of Triple Play applications over ADSL in cases of approximately 50% and 30% availability for access broadband networks at end users with full services at a time. In scenario #1, when the network availability is 100% (0% traffic in core links), many IPTV data and voice connections are supported simultaneously to 30 ADSL household with a guaranteed QoS. But this is not the real case; sharing bandwidth which needs to be managed can create scalability and manageable issues. Even though, these networks may not near term concerns, they need to be considered to ensure investments made now in the access network by saving bandwidth which are suitable for the evolution to true (saleable and manageable) multi-service networks. The not complete objects and parameters of these scenarios are arranged in a similar manner as in the previous case with a difference related to the network's performance, which has been observed by assuming that all network links has had, 50% load (50% availability) and 70% load (30% availability), in both cases (full services at a time and no full services at a time) simulating high utilization in the core network.

3.5.2 Simulation Triple Play Services over WiMAX Technology

After simulating Triple Play services over ADSL technology which is considered to be the major residential broadband network used for supporting Triple Play services in the world, it is a time to go through wireless communications. This simulation network uses several features to study the performances of WiMAX on Triple Play services, 3.5GHz band is chosen in this study because it is a widely used band all over the world. Moreover, this band is licensed, so that interfere is under control which allows using higher transmission power. Furthermore, it supports the NLOS condition, resulting better coverage range than that of 2.5GHz and 5.8GHz bands. WiMax does not offer a compelling reason to switch from other forms of residential broadband. When bundled with a broadband Internet access and VoD, WiMax Triple Play becomes a very attractive to residential subscriber. In this simulation scenario QoS, security and reliability mechanisms are combined into WiMax, providing good service to subscribers.

1. Network Topology for WiMAX

First the "Backbone" subnet, see Figure 3.16 initially the data, voice server and

VoD sources are defined. DSR router is used in the unicast mode for delivery unicast traffic (VoIP, Internet & VoD) to the BRAS at the aggregation subnet. Also, in this subnet PPP_switch is used for VLAN model, which has been visualized in Figures 3.10 and 3.11.

Figure 3.16: Backbone subnet inside WiMAX network topology

Second the "Aggregation" subnet, in this subnet all traffic from backbone subnet is received using PPP_SONET_OC24 with a data rate of 1244.16 Mpbs. Then the traffic is delivered to the CO router using the same link. Finally, it is delivered to the access network via a 45 Mbps DS3 WAN link at a customer side.

Finally WiMAX network subnet, in this subnet the BS and SS are changed from scenario to other depending to what is needed to be examined which is discussed later.

2. QoS Management in WiMAX Simulation Scenarios

Specifically, in all scenarios considered along this simulation, a WiMax segment is included inside an IP network using the DiffServ protocol for QoS management (that was configured in subsection 3.4.3). For a proper QoS transfer between the DiffServ and WiMAX domains, it is necessary to map the traffic classes defined by DiffServ over the appropriate MAC IEEE 802.16 scheduling service. There are four scheduler types: UGS, rtPS, nrtPS, and BE. The available bandwidth resources are first allocated to the UGS, then rtPS, next to the nrtPS flows. Finally, any remaining resources are then assigned to the BE flows. Based on the different features of traffic classes, the traffic mapping is proposed and depicted in Table 3.8. However, it is necessary to mention that this classification could be adapted to the requirements other scenarios, traffic profiles, and applications.

Table 3.8: Basic traffic classification according to QoS requirements

Traffic	**DiffServ PHB**	**WiMax Service**	**Example Application**
Real time, highly intolerant to delay and jitter. Fixed packet size and rate.	EF	UGS , ertPS	VoIP
Real time, highly intolerant to delay and jitter. Variable packet size and rate.	EF	UGS	Videoconference
No real time restrictions, high bandwidth, medium tolerance to delay, highly tolerant to jitter.	AF43	rtPS	Video-On-demand (HD,SD)
No real time restrictions, medium bandwidth, medium tolerance to delay, highly tolerant to jitter.	AF33	rtPS	Video Streaming
No real time restrictions, high tolerance to both losses and delay.	AF31	nrtPS	FTP
No real time restrictions, small bandwidth, medium tolerance to delay, highly tolerance to jitter, medium tolerance to losses.	BE	BE	E-mail, Web browsing HTTP

3. WiMAX MAC and PHY Characterization (Global Configuration)

In this section, the MAC and PHY system parameters are designed to reflect a practical WiMax deployment that maximizes video, voice and data content traffic. Unlike other wireless standards including WiFi (IEEE 802.11) which use a connectionless MAC layer, WiMax has adopted connection-oriented MAC that provides centralized resource allocations similar to DOCSIS v1.1 in many ways [40].

WiMax service classes capture the QoS requirements of service flows, where service flows represent the traffic flows between the base station and subscriber stations. Service flows from the base station to the subscriber station are called downlink flows, while service flows from the subscriber station to the base station are called uplink flows. For a given service class, the key parameters are minimum sustainable data rate (minimum guaranteed over the air (OTA) rate) and the media access control (MAC) scheduler type, which enables WiMax to provide QoS capabilities, thereby supporting delay sensitive traffic such as voice and video services.

In this study, four service classes are created for the downlink, it is achieved by first using Gold-DL, which is configured with UGS scheduling, 96Kbps for a minimum sustainable data rate and a 96Kbps maximum sustainable data rate for voices services. Then a Silver-DL is used, which is configured with rtPS scheduling, 1Mbps for a minimum sustainable data rate and a 5Mbps maximum sustainable data rate for VoD services. Next, Bronze-DL, this one is configured with nrtPS scheduling, 96Kbps for a minimum sustainable data rate and a 384Kbps maximum sustainable data rate for FTP services. Finally, Platinum-DL, which is configured with BE scheduling, 96Kbps for a minimum sustainable data rate and a 384Kbps maximum sustainable data rate for HTTP services.

Another, four service classes are created for the uplink, it is achieved by first using Gold-UP, which is configured with UGS scheduling, 96Kbps for a minimum sustainable data rate and a 96Kbps maximum sustainable data rate for voices services. Then a Silver-UP is used, which is configured with rtPS scheduling, 1Mbps for a minimum sustainable data rate and a 0.5Mbps maximum sustainable data rate for VoD services. Next, Bronze-UP, this one is configured with nrtPS scheduling, 96Kbps for a minimum sustainable data rate and a 200Kbps maximum sustainable data rate for FTP services. Finally, Platinum-DL, which is configured with BE scheduling, 96Kbps for a minimum sustainable data rate and a 200Kbps maximum sustainable data rate for HTTP services. Figure 3.17 shows the WiMax MAC service class configuration used for this simulation model. The derived AMC is also configured into the WiMax configuration nodes, which is explained in the next section. The OFDM PHY profile configuration for a different channel bandwidth. It should be noted that the frame structure for a given channel bandwidth is varied according to the design. In addition, frame duration of 5ms is used with a base frequency 3.5GHz for all 20, 10 and 5 MHz channel bandwidth.

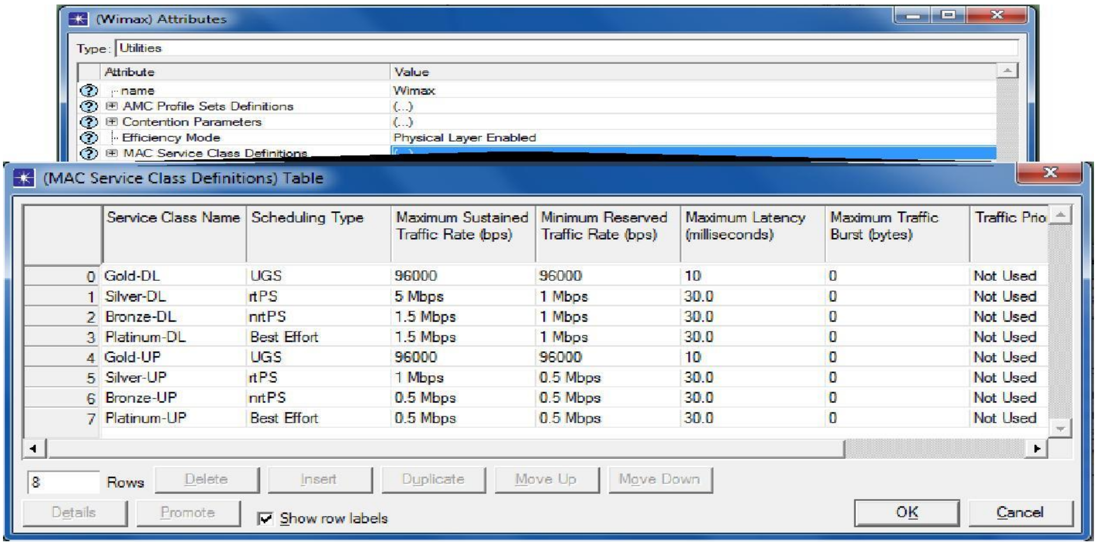


Figure 3.17: MAC service class configuration

4. Configuration WiMAX Base Stations

This section describes the configuration used in the WiMax BS for all scenarios. A single sector BS node model is used to model all BSs with antenna gain=15 dBi, and a max transmission power=20W. BSs are then configured to map the higher level of Triple Play application traffic to a service class by setting service class to "Gold-DL" if type of service (DCSP) parameter matches the "EF" , "Silver-DL" if type of service (DSCP) parameter matches the "AF43", "Bronze-DL" if type of service (DSCP) parameter matches the "AF13", "Platinum-DL" if type of service (DSCP) parameter matches the "BE". Figure 3.18 shows the configuration of service class mapping at the base station. A similar operation is also performed on the SS in the next section.

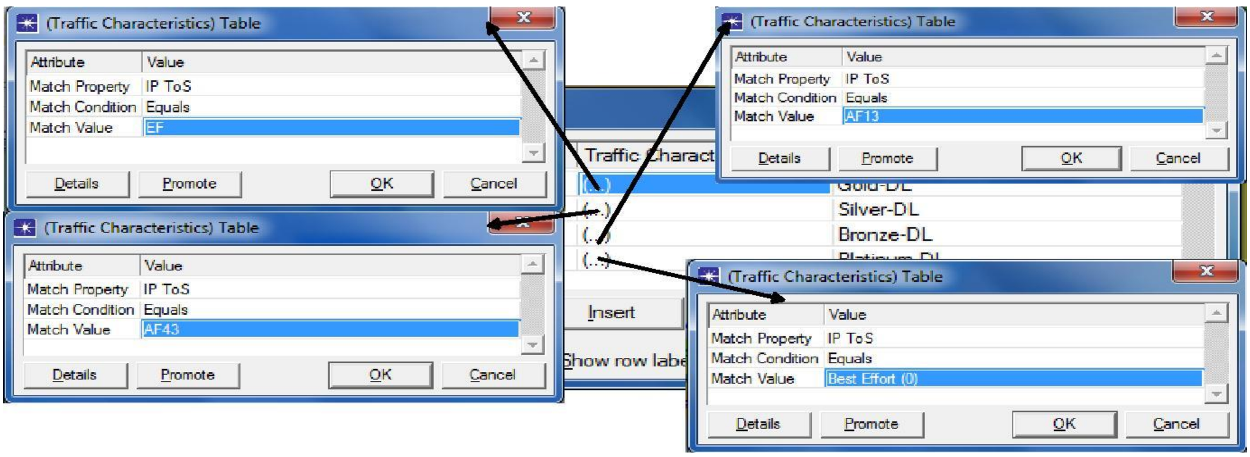


Figure 3.18: DL service class mapping

5. Configuration WiMAX Subscriber Stations

This section describes configuration for the subscriber stations. Firstly, each SS is configured with antenna gain=14 dBi and max transmission power=0.5W. Then SS is configured to map the higher level of Triple Play application to the "Gold-UL", "Silver-UL", "Bronze-UL", and "Platinum-UL" service classes in a similar manner as with the BS performed in the previous section..

In this model, the UL channel is assumed to have similar properties to the DL channel, so the same modulation and coding scheme are configured on UL and DL service flows. WiMax client stations are manually configured with more robust modulation/coding schemes and increased distance from the base station. The DL and UL for all service flows in 2Km, 4Km and 6Km configurations are pictured in figure 3.19. This configuration also incorporates the modulation and coding parameter, average SDU size in byte, buffer size in byte, ARQ parameter, etc...

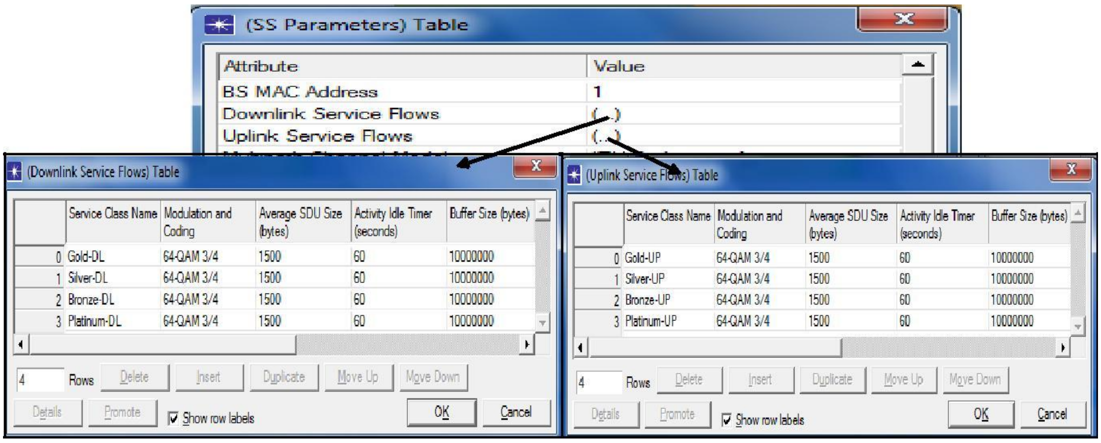


Figure 3.19: Configured DL and UL for all service flow in 2Km

6. Simulation Scenarios for WiMAX

Simulated scenarios are discussed individually. Types of application, the profile configuration and different parameters are configured into the network topology have been, derived and explained in the previous subsections. At these scenarios the behavior of various propagation models in WiMax network at 3.5GHz with respect to pathloss, packet end-to-end delay, delay-variation (Jitter), FTP download response time and HTTP page response for various terrain models with Triple Play applications is studied. It should be noted that the modeler provides three distinct coordinate systems to model node distances and corresponding wire and wireless path lengths. In these models, the geocentric coordinate system is adopted by using the latitude and longitude. The specifications of model components in each scenario are discussed In general, each scenario consists of N SS nodes and one BS to cover the geographical represented area

. The result and analysis are expressed in chapter four, and the main settings and parameters of these scenarios presented below:

- Simulation Time: Simulation time is set to (1000 sec).
- Efficiency Mode: Physical layer enable.
- Bandwidth is ranging from 5 to 20 MHz (scenario dependent).
- Base Frequency: 3.5 GHz.
- MAC Service Class Definition: as shown in Figure 3.17.
- BS configuration: as shown in subsection (3.5.2, point 4).
- SS's configuration: as shown in subsection (3.5.2, point 5).

Scenarios # 1, 2, 3: Figure 3.20 shows the arrangement of objects in these scenarios. These scenarios contain one BS serving with three SS's located at 2, 4, and 6 km from the WiMax base station, since the objective of these scenarios is to examine the effect of a different channel bandwidth for the base frequency 3.5GHz with different distance environments that incorporate the modulation and coding parameter.

Figure 3.20: The arrangement of objects in scenario # 1, 2, 3

Scenarios # 4, 5, 6: The objective of these scenarios is to investigate the combined effect of terrain and multipath channel model (ITU Pedestrian_A) on Triple Play services. At these scenarios 5 SSs are used in the range of BS as shown in Figure 3.21. The terrains are simulated by choosing the terrain type C in OPNET which is selected, based on location (2Km, 4Km and 6Km) and surrounding terrain of transmitter-receiver pair.

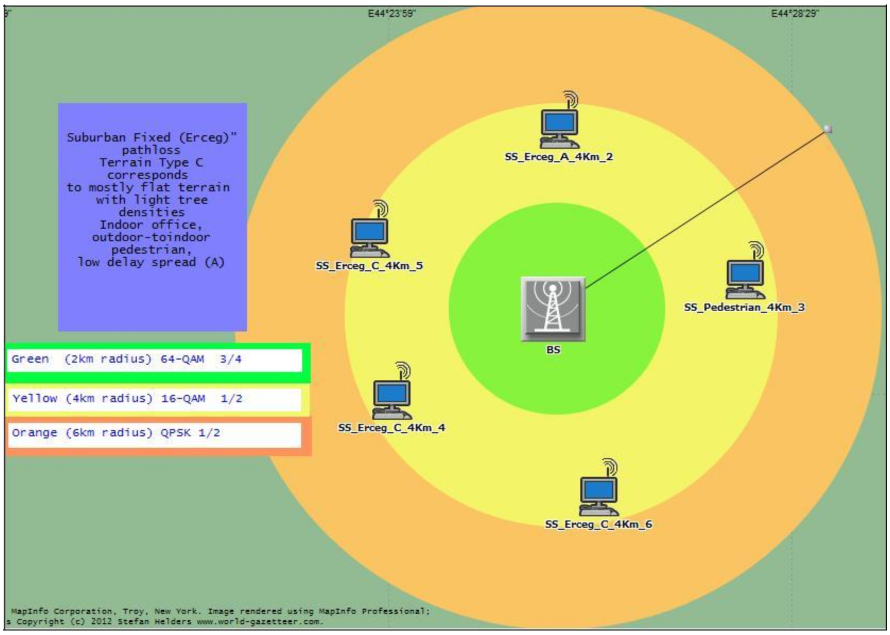


Figure 3.21: The arrangement of objects in scenario # **4, 5, 6**

CHAPTER FOUR

SIMULATION RESULTAND DISCUTOIN

4 SIMULATION RESULTAND DISCUTOIN

In this chapter the results of simulation scenarios presented in chapter three are introduced to evaluate the performance of sending Triple Play services over a hybrid wire-wireless networks scenarios . Consequently, in order to determine the requirements of each application independently, the terms of QoS performance metrics (packet loss, packet delay, packet jitter, etc.) are included within the following section, which are evaluated and discussed for each scenario.

4.1 Triple Play over ADSL Network Simulation Scenarios

As mentioned in section (3.5.1), the simulation process is divided into 5 scenarios that are depicted in Table 3.7. The results related to these scenarios are presented and analyzed in the following subsections.

1. Scenarios # 1, 2

This simulation project is trace driven using actual video traces that configured in Table 3.6. Video traffic is a key aspect of the study as its intensive inherent bandwidth and delay sensitive properties, stress the access links further than most of other types of application traffic such as voice and data.

Figures 4.1, 4.2 and 4.3 shows the performance metrics of packet loss ratio, packet jitters and end-to-end delay, respectively, which are used to quantify video streaming that is compressed by using MPEG-2 and MPEG-4 format.

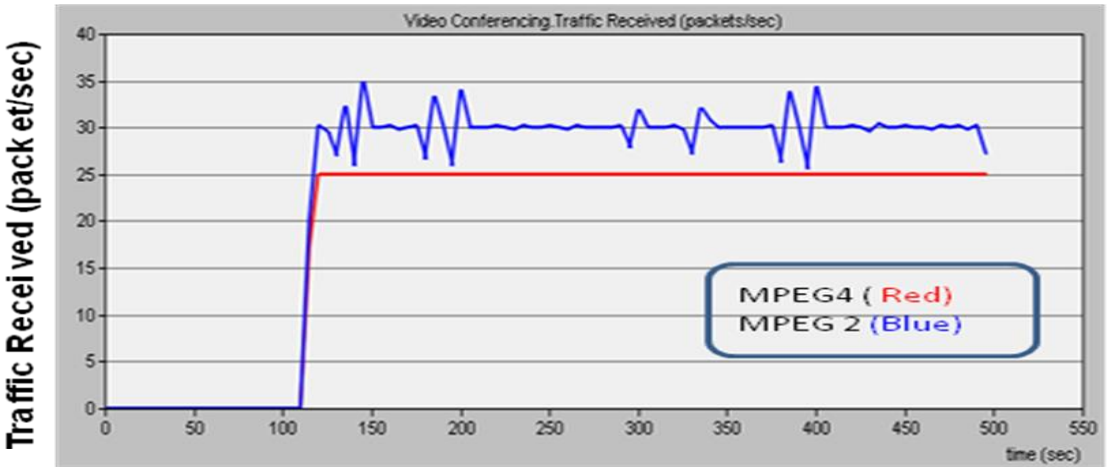


Figure 4.1: The MPEG-2 and MPEG-4 video PLR

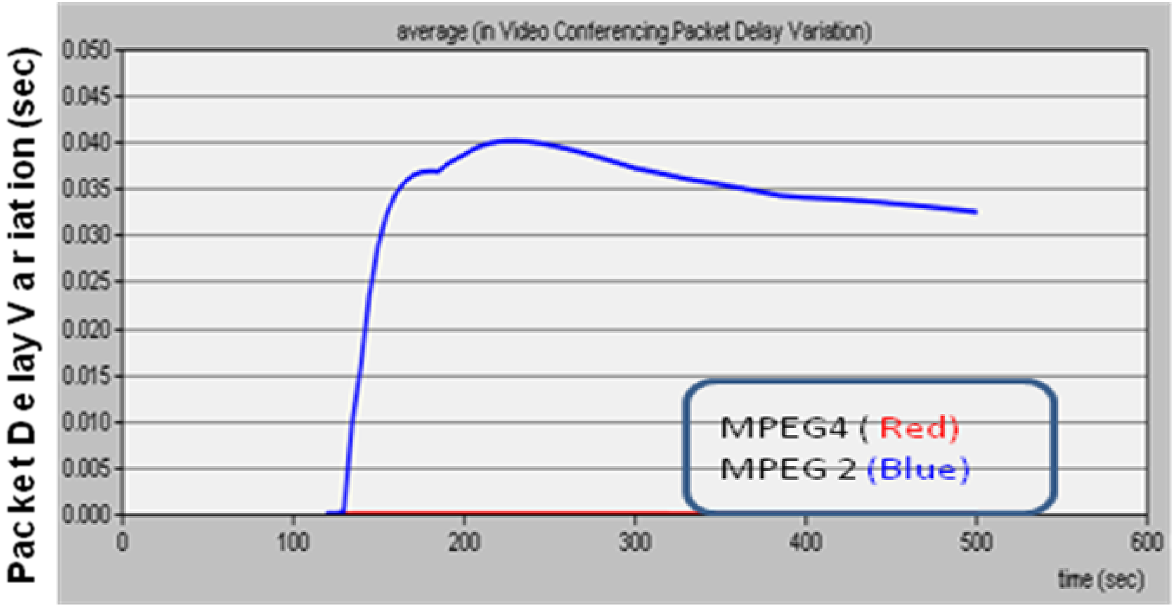


Figure 4.2: The MPEG-2 and MPEG-4 video packet jitter

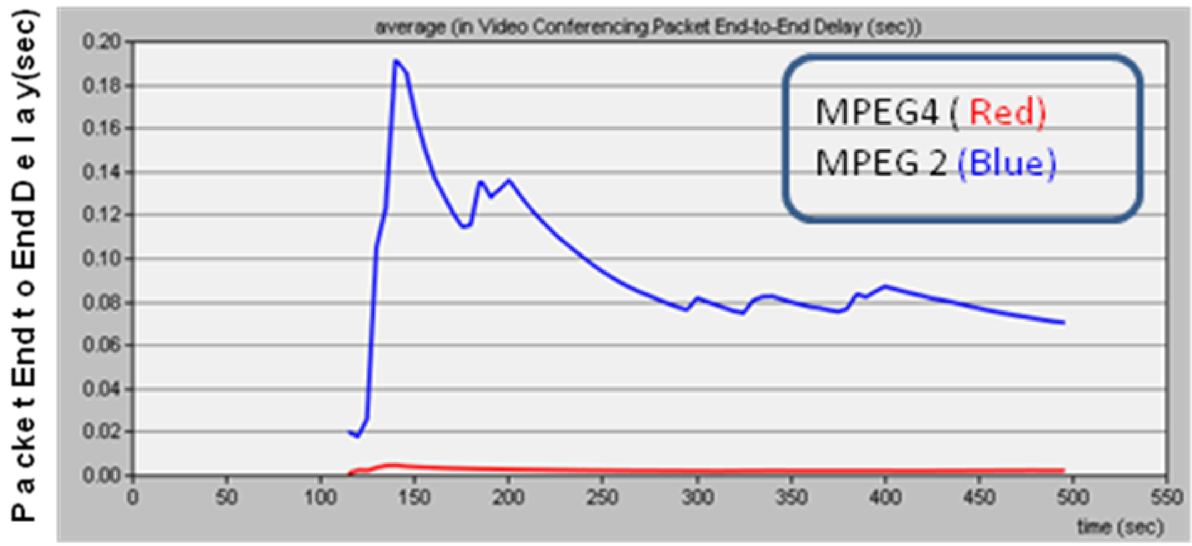
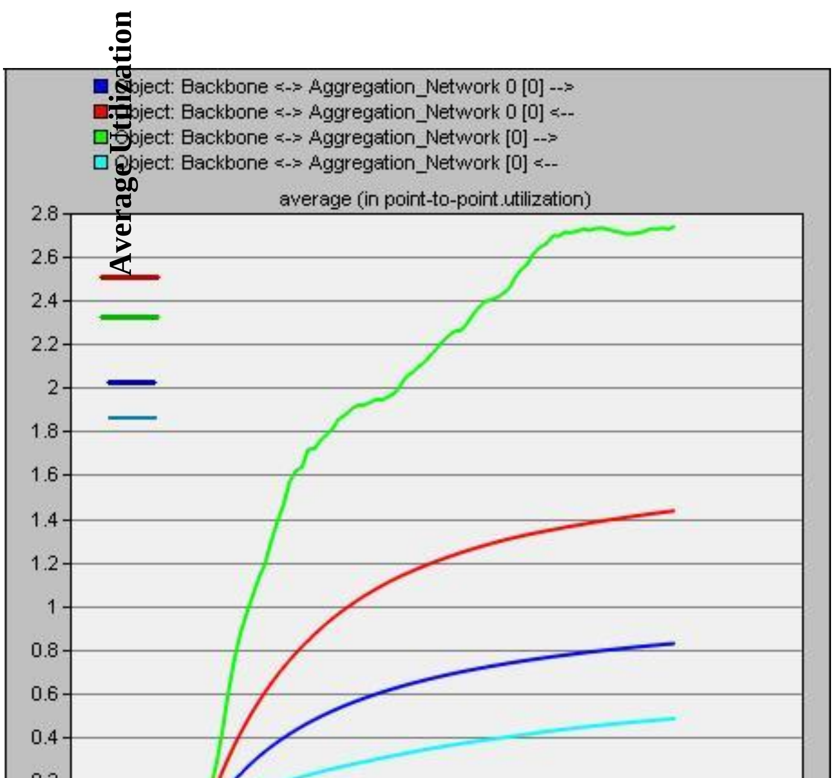


Figure 4.3: The MPEG-2 and MPEG-4 video end-to-end delay

At these scenarios, also sender uses IP multicast technology to set up a video conference session for IPTV services with all receivers. In multicast, only a single session is set up for all receivers. Therefore, the sender sends only one copy of each video packet. Figure 4.4 shows the interface level as an example highlighting link utilization with and without IP multicasting.



P Multicast traffic

IP Unicast traffic

Upstream Core Link Usage

IP Multicast traffic

IP Unicast traffic

Downstream Core Link Usage

I

Time (sec)

Figure 4.4: IP Multicast and IP Unicast core link usage

2. Scenarios # 3,4,5

In these scenarios, the network is tested for the cases when the network availability is approximately 50% (services is full at each home at a time, 50% traffic in core links), 30% (services is full at each home at a time, 70% traffic in core links) and 30% (services is not full at each home at a time, 70% traffic in core links), as shown in Figure 4.5. Furthermore, the number of subscribers for all cases is the same for these network topologies. The same services, profiles are applied in order to compare between these network availability in the different test.

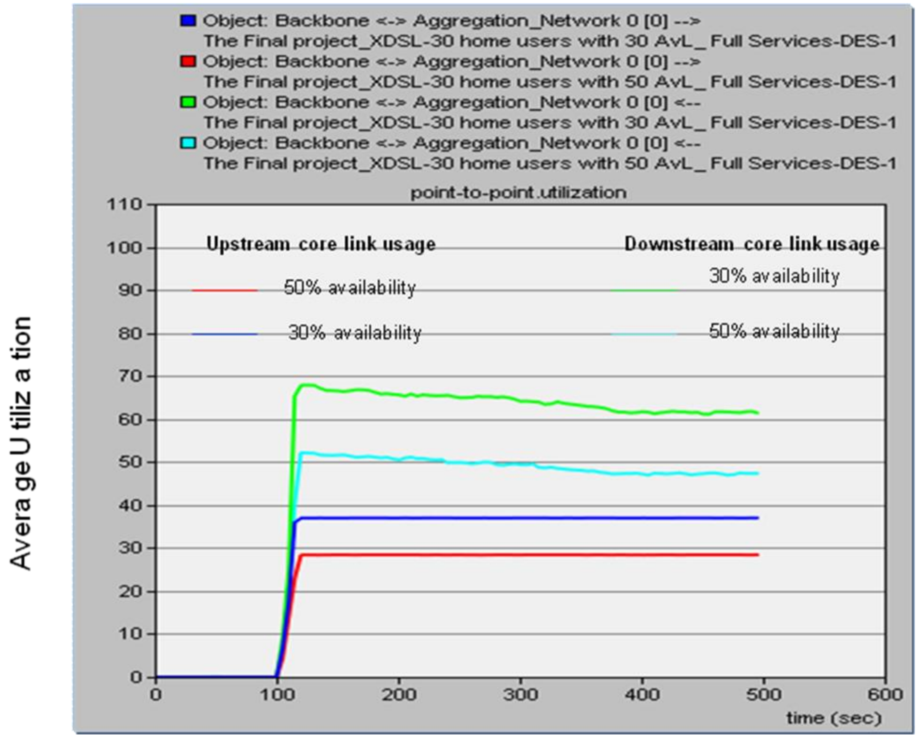
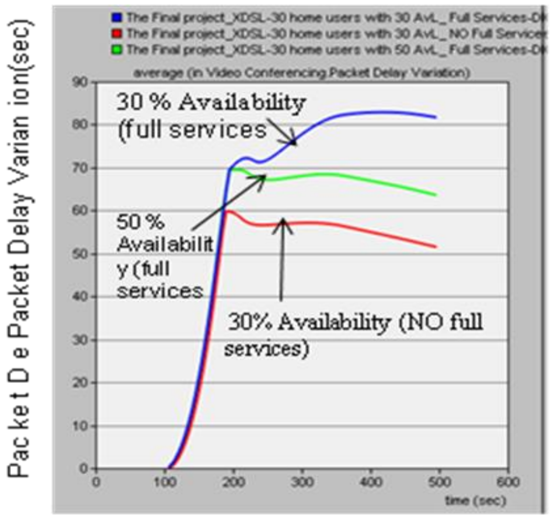


Figure 4.5: Core link usage

Figures 4.6 and 4.7 show the average delay-variation (Jitter). These results represent the first performance metric used to quantify voice and video services content traffic over ADSL for the three cases of bandwidth availability.



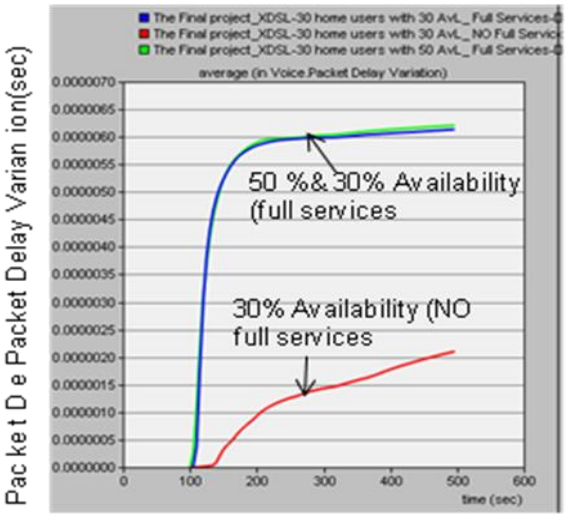


Figure 4.6: Video packet delay-variation **Figure 4.**7: Voice packet delay-variation

HTTP object response time and FTP downloads response time are another performance metric used in the quantification of the HTTP and FTP services content traffic over ADSL for the three cases of bandwidth availability, as shown in Figures 4.8 and 4.9.

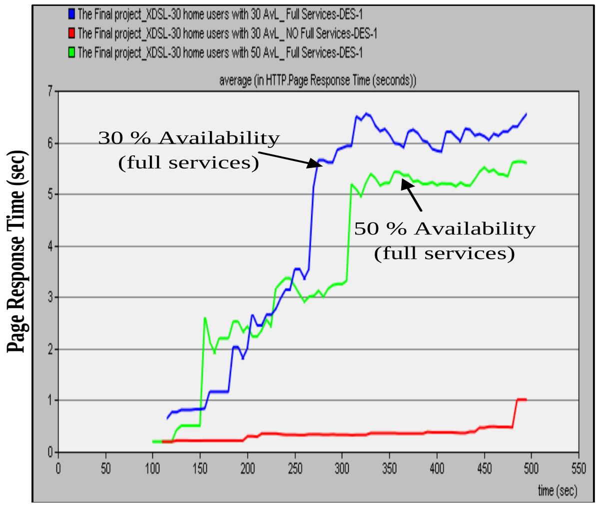


Figure 4.8: HTTP object response time

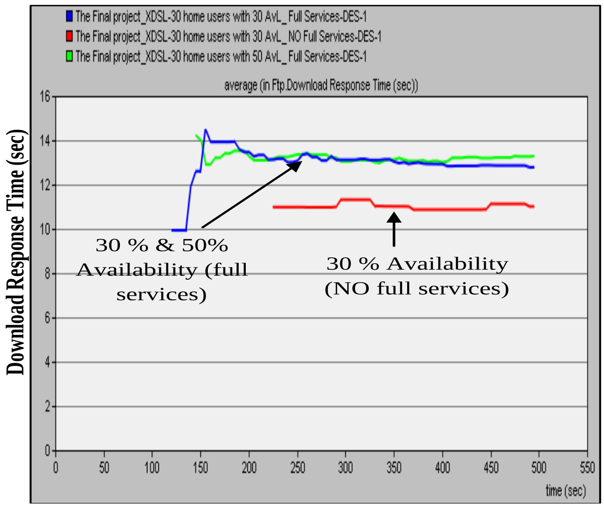


Figure 4.9: FTP downloads response time

3. Discussion

In this section, what is understood from these results is that:

Due to high bandwidth demand of video services, it is important to use more active compression formatting, i.e., MPEG-4. It is clear from the results, the effect of video type into network parameters such as packet loss, jitter and reordering problems, detects more problems when MPEG-2 codec format is used. Consequently, MPEG-2 video content is abandoned and instead in focused on the MPEG-4 video content is adopted.

In such system which consists of different types of services, when using unicast technology transmitting the same data multiple times a lot of resources will be weakened which is not an economical use of the network capacity. So the efficient transmitting is achieved by using multicast technology in which only a single data is setup for all receivers.

The results also show that, when network availability is 50% (full services at each household) delay-variation is low. When the available bandwidth decreases to 30% delay-variation for voice and video services is increased exponentially. For the same available bandwidth, but without full services at each household, delay-variation is considered excellent and best performing scenario can be achieved. As observed, HTTP is required to be downloaded to user terminal from (6.5sec - 5.5sec) when network availability is 30% and 50% (full services at each household), while if the available bandwidth is 30% but no full services at each household it requires (1sec). Also, FTP download response time has a downloading time from (13-14sec) when network availability is 30% and 50%, the downloading time became (10.5sec) when network availability is 30% without full services at each household, but this will not be taken into consideration by the final user. However, if the server serves more users or located into another ISP's premises, this value will increase more, probably reaching a delay value estimated in minutes.

4.2 Triple Play over WiMax Network Simulation Scenarios

As mentioned in section (3.5.2), the simulation process is divided into 9 scenarios, as depicted in Table 3.9. The results for each scenario are explained in the following subsections.

1. **Scenarios # 1, 2, 3**

In scenario #1, the channel bandwidth 20 MHz and frequency 3.5GHz, the system is tested at 6 km, 4 km and 2 km. The packet delay and delay-variation (Jitter) are the first performance metric used to quantify video services for the three WiMax stations, as shown in Figures 4.10 and 4.11.

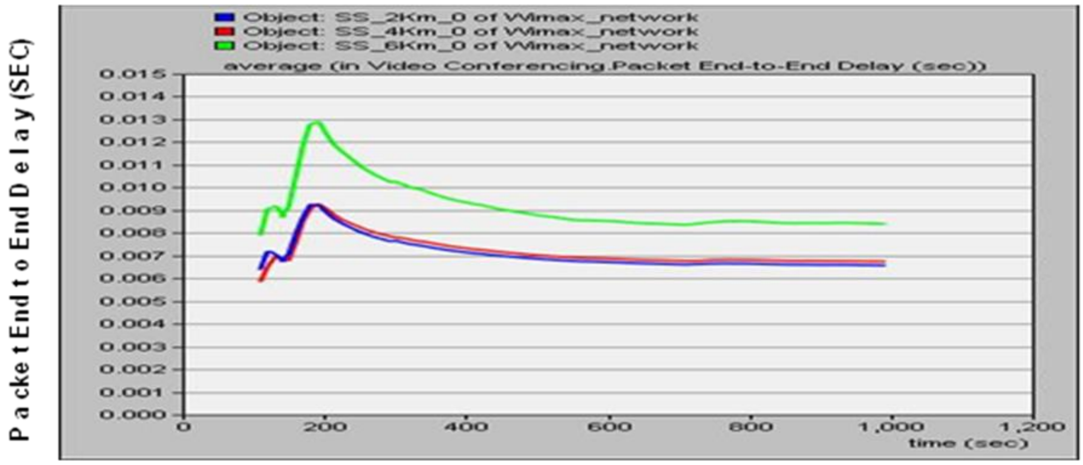


Figure 4.10: SS video packet delays for 20 MHz channel bandwidth

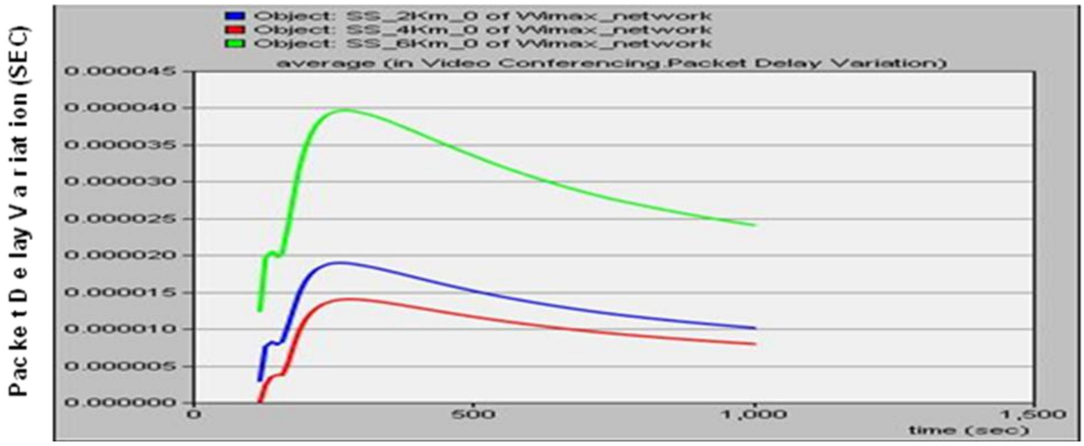
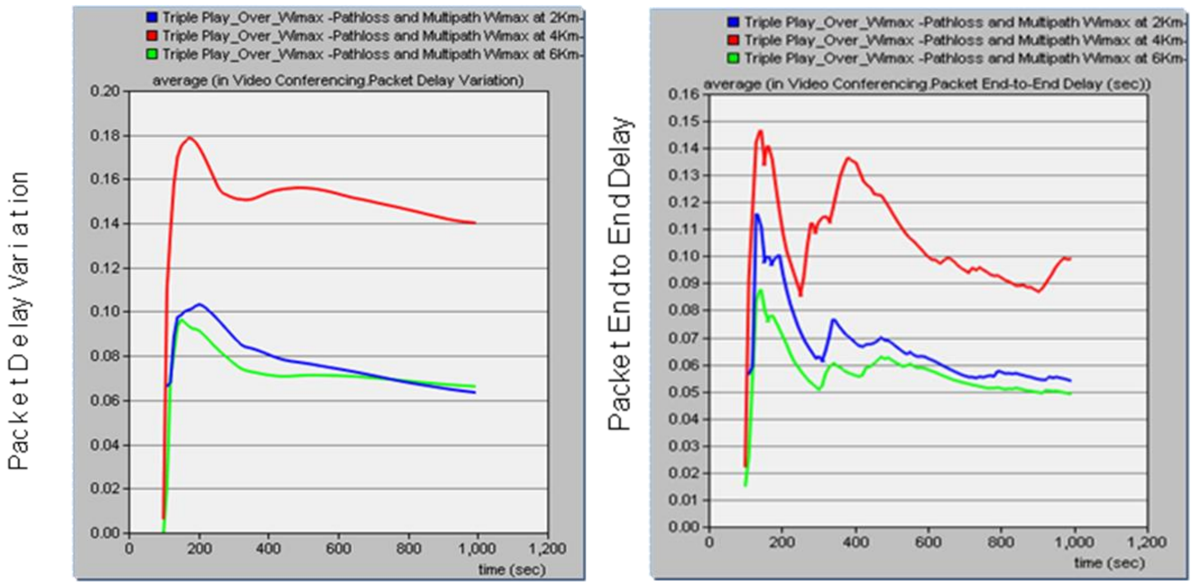


Figure 4.11: SS video delay-variation for 20 MHz channel bandwidth

Packet end-to-end delay, delay-variation (Jitter), FTP downloads response time and HTTP page response time are the next performance metrics used to quantify Triple Play services content streaming over WiMax in scenarios # 1, 2, 3. Observing that these 3 scenarios have been simulated successfully, using a configuration of 5ms frame duration and frequency 3.5GHz without MIMO and retransmissions. A comparison is made for 5MHz channel bandwidth and 10MHz channel bandwidth, for the suggested distances. The mean traffic rate is specified corresponding to the metrics listed in Table 2.1

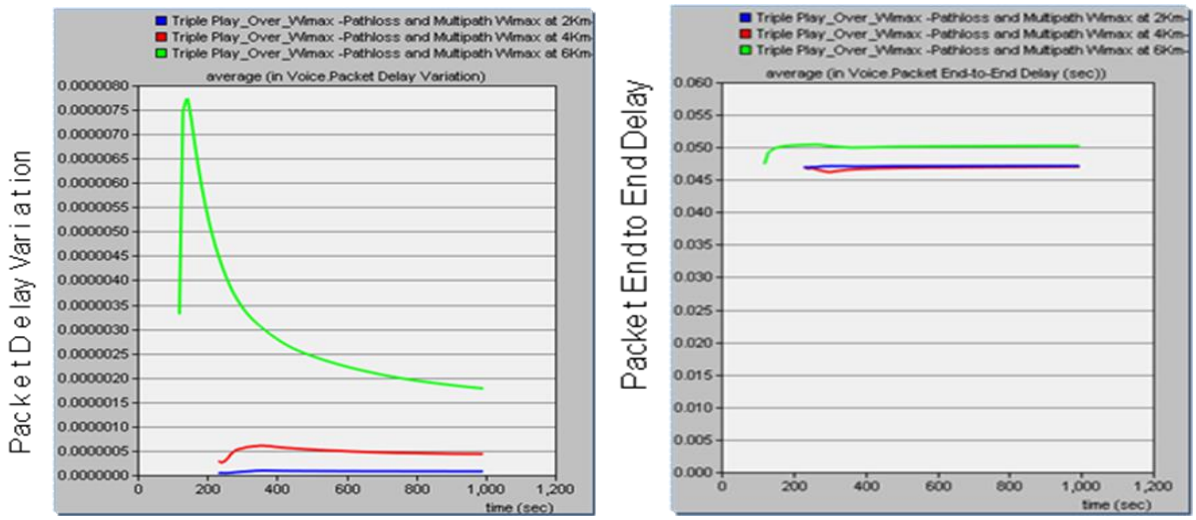
2. Scenarios # 4,5,6

Finally, for the purposes of this study, it was important to factor pathloss and multipath channel model into the performance of our WiMax clients with Triple Play services, so at these scenarios, also a fixed suburban (Erceg) model and multipath channel model which is indoor office, outdoor-to-indoor pedestrian, low delay spread (A) was employed with a conservative terrain model The measured results are shown in Figures 4.12, 4.13 and 4.14 for packet end-to-end delay, delay-variation (Jitter), FTP download response time and HTTP page response time, which are the performance metrics (Global statistics) used in the quantification of voice, Internet and video services in pathloss with multipath channel model effect at a different distance.



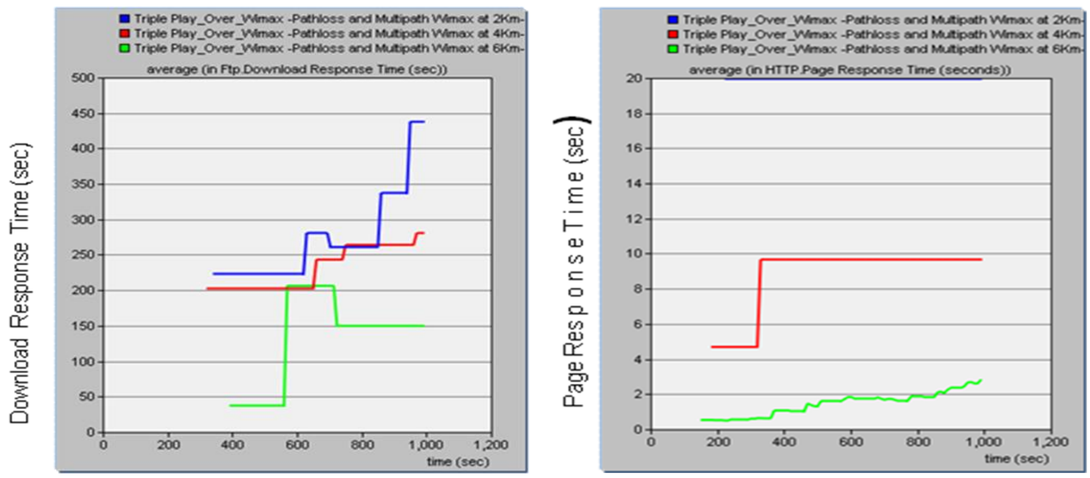
(a) **(b)**

Figure 4.12:(a) Video packet delay and (b) delay variation at both pathloss & multipath effect



(a) **(b)**

Figure 4.13 (a) Voice packet delay and **(b)** delay variation at both pathloss& multipath effect



(a) **(b)**

Figure 4.14: (a) FTP & **(b)** HTTP download response time at both pathloss& multipath effect

4. Discussion

This study explored the technical details and performance of WiMax broadband access technology. The results of simulation indicate that the slope of propagation losses is constant because of fixed SS and:

The packet delay and delay-variation of video service for the three WiMax stations are shown in Figures 4.10 and 4.11, respectively. It can be noticed from these results, 6km

WiMax station exhibits a much higher loss rate than the 4km and 2km stations over the 1000sec interval. A packet end-to-end delay, delay variation, FTP downloads response time and HTTP page response time have been studied for frequency of 3.5GHz with different channel bandwidth and distances between BS and SS. It can be noticed in Table 4.2 that the results are largest for small channel bandwidth and also when the distance between BS and SS is increased.

The pathloss is small for free space and large for SS in outdoor to indoor and pedestrian environment. Where, the SS in terrain type C, the pathloss will be more than that in free space. The SS in terrain type A the pathloss will be less than that in outdoor to indoor. The SS in terrain type B, the pathloss will be between terrain types A and C. Also, the pathloss is largest when the distance between BS and SS is increased.

It should be noted that if multipath channel modeling is enabled, the transmitted signal has been distorted due to the multipath and dispersion effects of the channel. It becomes more difficult to understand if changes (in delay, delay-various, FTP download response time and HTTP page response time) are attributed to interference or to the direct path between Tx and Rx which is a function of the distances (2km, 4km and 6km). Since the multipath reflections are not easy realized also are random in nature.

CHAPTER FIVE

CONCLUSIONS AND RECOMMENDATION

5.1 Conclusions

In this project, it is found from the initial system test that H.264 (MPEG-4) codec must increasingly be used instead of the MPEG-2 codec for Triple Play services over ADSL access technology. To decrease consumption in network video streaming, the video is send either via IP multicast in case of live TV (IPTV) or via IP unicast in case of video on demand. Also, the link congestion must be avoided at all costs, and links should operate at 75% of their total capacity (25% availability), in order to avoid Triple Play services degradation, then they should program ADSL network upgrade, in order to decrease their network bandwidth availability and maintain QoS.

In order to maintain a standard quality for delivered Triple Play services to keep the customer satisfied with that current follow ITU-T G.114 recommendation and the Y.1541 QoS recommendations for PLR, end-to end delay and jitter, the channel bandwidth for system performance over WiMax at 3.5GHz frequency bands should be chosen to be between 10 and 20MHz. Radio pathloss and ITU multipath models in various environments was simulated where one of the main target of this work is to check the variation of the network performance due to varying pathloss and terrain models.

Fundamental and persistent differences are identified between wired and wireless networks, which propel wired and wireless access network architectures on divergent evolutionary paths. Where as it is expected wired broadband access networks to continue to evolve toward common general-purpose platform architecture. Therefore, it is expected wireless networks to remain heterogeneous. The inherent scarcity of radio frequency spectrum emerges as the key reason for this prediction

5.2 Recommendation

Triple Play to N play, conduct comprehensive analysis on all Triple Play services and encapsulates video traffic in RTP, in addition, a special packetization scheme is used, which is based on RTP/UDP/IP encapsulation. This configuration would more accurately model the actual protocol overhead associated with video streaming services. The current video traces in this project do not account for audio content. Accordingly, incorporating audio data would make the model more realistic.

Opportunity window for wireless to provide wireless access network for capacity hungry applications. 4th generation wireless protocols, in order to perform a more complete analysis of the next generation access network implementation. It has combined all essential networking functions broadband access to Internet, show T.V and staying mobile.

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