

DISSERTATION TITLE
**PERFORMANCE EVALUATION OF WIMAX FOR RURAL
BACKHAUL**

SUBMITTED BY

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IN PARTIAL FULFILMENT OF THE DEGREE

Master of Science in Engineering with specialization
in the field of Telecommunications and Information Technology
from the University of KwaZulu-Natal



DATE OF SUBMISSION

23 February 2007

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DECLARATION

I, Verosha Singh, Student Number 200101020, hereby declare that the dissertation entitled **Performance Evaluation of WiMAX for Rural Backhaul** is a result of my own investigation and research, and presents my own work unless specifically referenced in the text. This work has not been submitted in part or full for any other degree or to any other University.

Friday, 23 February 2007

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ABSTRACT

Technologies such as WiFi and WiMAX, can be a powerful driving force for increasing rural connectivity. This research proposes use of WiMAX as backhaul to WiFi hotspots in order to provide cost effective and easy to deploy rural telecommunication services. The hotspots will take the form of telecentres. The IP-based network will provide voice, data and Internet services. The work evaluates the performance of WiMAX as a backhaul technology based on a typical South African rural region as a case study. Further, telecommunication traffic estimates for the region are determined. Multiple simulations are carried out in OPNET to determine the quality across different services, based on these user requirements. The future demands on the proposed network are also considered. Quality measurements will include IP packet loss and delay, as well as perceived voice quality during peak traffic periods. In, addition the cost benefits of using a mesh-based solution to cover the entire geographic region of Nongoma are considered. The ability of IEEE 802.16 mesh mode to extend coverage is also investigated.

ACKNOWLEDGEMENTS

To my family, words can't express my appreciation, for the continuous love, support and prayer throughout my career.

I acknowledge my Supervisor, Mr. R Sewsunker, for his insight, forward thinking and inspiration throughout my MSc degree by dissertation and coursework.

Further, I acknowledge Professor T.J.O. Afullo for his conscientious support during this degree. Also to the School's IT manager, Mr Bruce Harrison, thank you.

I recognize and appreciate the financial endorsement by Telkom South Africa for my postgraduate studies.

To my friends and fellow colleagues from the Centre of Excellence, in particular Karuna, Charlene and Deshree, thank you.

To Ananda, thank you for your love and continuous encouragement. I would not have done this without you.

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LIST OF ABBREVIATIONS AND ACRONYMS

ACK	Acknowledgement
ARQ	Automatic Repeat Request
BE	Best Effort
BPSK	Binary Phase Shift Keying
BS	Base Station
BSS	Basic Service Set
BW	Bandwidth
BWA	Broadband Wireless Access
CID	Connection Identifier
codec	coder/decoder
CPS	Common Part Sublayer
CRC	Cyclic Redundancy Check
CRTP	Compressed Real Time Transport Protocol
CS	Service Specific Convergence Sublayer
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear to Send
DCF	Distributed Coordination Function
DFS	Dynamic Frequency Selection
DHCP	Dynamic Host Configuration Protocol
DIFS	Distributed Inter-Frame Space
DIUC	Downlink Interval Usage Code
DL	Downlink
DSL	Digital Subscriber Line
DSSS	Direct Sequence Spread Spectrum
EDGE	Enhanced Data for Global Evolution
ESS	Extended Service Set
ETSI	European Telecommunication Standards Institute
FDD	Frequency Division Duplex
FDM	Frequency Division Multiplex
FEC	Forward Error Correction
FHSS	Frequency Hop Spread Spectrum

FTP	File Transfer Protocol
GPC	Grant per Connection
GPRS	General Packet Radio Services
GPSS	Grant per Subscriber Station
GSM	Global System for Mobile Communications
HiperLAN	High Performance Local Area Network
HiperMAN	High performance Metropolitan Area Network
IBSS	Independent Basic Service Set
ICT	Information and Communication Technology
IE	Information Element
IEEE	International Institute of Electrical and Electronic Engineers
IP	Internet Protocol
ISRDP	Integrated Sustainable Rural Development Programme
ITU	International Telecommunications Union
kbps	kilobits per second
LOS	Line of Site
MAC	Media Access Control
Mbps	Megabits per second
MBS	Mesh Base Station
MDRR	Modified Deficit Round Robin
MOS	Mean Opinion Score
MPDU	Media access control Protocol Data Unit
MRTR	Minimum Reserved Traffic Rate
MSDU	Media access control Service Data Unit
NLOS	Non-Line of Sight
nrtPS	Non-Real Time Polling Service
OFDM	Orthogonal Frequency Division Multiplex
OSI	Open System Interconnection
PC	Personal Computer
PCF	Point Coordination Function
PDA	Personal Digital Assistant
PDU	Protocol Data Unit
PHY	Physical Layer
PMP	Point-to-Multipoint

PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
rtPS	Real Time Polling Service
RTS	Request to Send
SAP	Service Access Point
SDU	Service Data Unit
SFID	Service Flow Identifier
SNMP	Simple Network Management Protocol
SNR	Signal-to-Noise Ratio
SS	Subscriber Station
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplex
TDMA	Time Division Multiple Access
TFTP	Trivial File Transfer Protocol
TIA	Telecomms Industry Association
UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
UL	Uplink
VoIP	Voice over Internet Protocol
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
WISP	Wireless Internet Service Provider
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network
WPAN	Wireless Personal Area Network
WWAN	Wireless Wide Area Network

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1 INTRODUCTION

1.1 Wireless Technology for Rural Application

There is need for cost-effective telecommunication systems in the rural and remote regions of South Africa that are capable of providing reliable voice and data services with adequate capacity for growth. People in these regions are presented with many challenges, whilst trying to participate meaningfully in the global economy. The most significant challenge is the inability to access information that can promote education, employment, trade and wealth. Information and communication technology (ICT) is regarded as an empowering force that can help overcome these challenges, and thus bridge the existing digital divide.

The diverse range of wireless technologies available has the potential to provide delivery of multimedia applications to these regions and, more importantly, has the potential to bridge the digital divide by providing low-cost broadband Internet connectivity to rural under-served areas. Its deployment can become a useful tool in reducing isolation, increasing economic prosperity and improving productivity. It can offer many of the advantages of wired broadband Internet access with much less investment in infrastructure. It is a feasible means of extending Internet connectivity to more areas in the developing world, Africa, and South Africa in particular. This is essential in order to transform society.

There are many types of wireless deployments available today, each with its own technology and characteristics. These evolving technologies can be classified by the hierarchy shown in Figure 1.1 below.

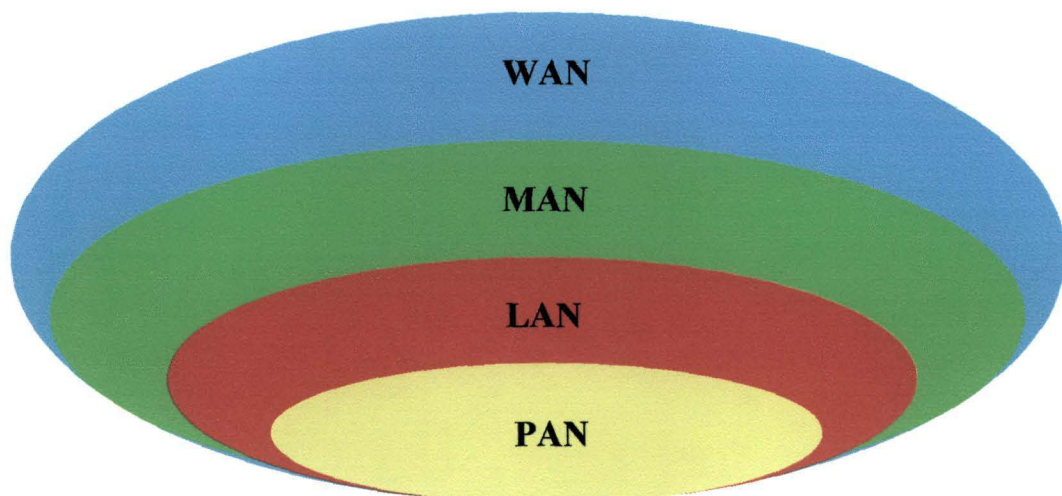


Figure 1.1 - Wireless Hierarchy

A wireless personal area network (WPAN) is the smallest scale network, allowing devices to communicate over short distances. WPANs connect devices, such as personal computers (PCs), personal digital assistants (PDAs), cell phones, around a person's workspace. It has a maximum range of approximately 10m. A popular type of WPAN technology is IEEE 802.15, Bluetooth. The European Telecommunications Standards Institute (ETSI) high performance personal area network (HiperPAN) standard is also another type of PAN.

A wireless local area network (WLAN) typically aggregates personal area networks, and includes technologies such as the Institute of Electrical and Electronic Engineers (IEEE) 802.11 standards and ETSI HiperLAN. These technologies permit devices to share information up to approximately 100m. WLAN standards can currently provide theoretical data rates of up to 54Mb/s. There has to be line-of-sight (LOS) between the transmitting and receiving devices.

The next step on the networking scale is the wireless metropolitan area network (WMAN), which allows networks the size of cities to be connected. Metropolitan area networks aggregate local area networks. Reach limits for WMANs are significantly larger than WLANs and may be up to 50km or more [1]. WMAN standards such as IEEE 802.16 can attain data rates of up to 70Mbps [2]. Other WMAN technologies include ETSI HiperMAN.

The final step in the area network scale is the wireless wide area network (WWAN). This is a network that spans a relatively large geographical area. Examples of wireless wide area network technologies are global system for mobile communications (GSM); general packet radio services (GPRS), and enhanced data for global evolution (EDGE) networks. Included in this segment of wide area networks is the proposed IEEE 802.20 standard, which specifies the air-interface for mobile users travelling at high speed.

The usage for each of the wireless network segments; PAN, LAN, MAN, and WAN depends on numerous variables. These include bandwidth and distance needs. Research into improving the performance of the current wireless technologies is unending.

1.2 Motivation for Research

Since few people in rural Africa, and South Africa can afford computers, and thus Internet access, a practical approach is public telecommunication access facilities, such as telecentres. There is a rapidly growing use of public access services such as telecentres, and cyber cafés in Africa, that are

promoting use of the Internet and voice over IP (VoIP). Telecentres encourage local economic and social development. The telecentre solution, though, needs to be sustainable, affordable, low-maintenance, and capable of providing service for multimedia applications.

The Integrated Sustainable Rural Development Programme (ISRDP) is a government poverty relief program that has identified 13 District Municipalities within South Africa that are underserved in terms of basic service, communication infrastructure and economic activities. This research investigates the use of wireless technology integrated with telecentres as a solution for one of these nodes, the Zululand District municipality, and in particular the Nongoma locality.

In order for the telecentres to be viewed as an opportunity for the Nongoma community to gain access to vital news and information, they need to be part of a development strategy that includes investment in all primary sectors, a development strategy such as the ISRDP.

The complementary wireless LAN and MAN technologies, WiFi and WiMAX, respectively, can provide such a solution. Wireless Fidelity, WiFi denotes a set of standards for WLAN products that are compliant with IEEE 802.11 specifications. Worldwide Interoperability for Microwave Access, WiMAX, is the IEEE standards industry consortium, which promotes 802.16 compliant technologies.

Generally, it is accepted that WiMAX will be used to deliver bandwidth to Wi-Fi. This allows the mature WiFi standard to reach the end user, while WiMAX can be used as backhaul. Wireless Internet Service Provider's (WISPs) greatest cost is considered to be backhaul, thus using WiMAX as backhaul can significantly reduce costs. This wireless technology also has the ability to provide differentiated service to applications, and thus quality of service (QoS). This is necessary for real-time applications such as voice and video, which have stringent delay requirements. These factors are key drivers for WiFi/WiMAX coexistence in a rural setting.

The envisioned architecture for the telecentre solution is that WiFi is used to distribute Internet service within the telecentre, and WiMAX is used as backhaul to interconnect WiFi hotspots, as seen in Figure 1.2 below:

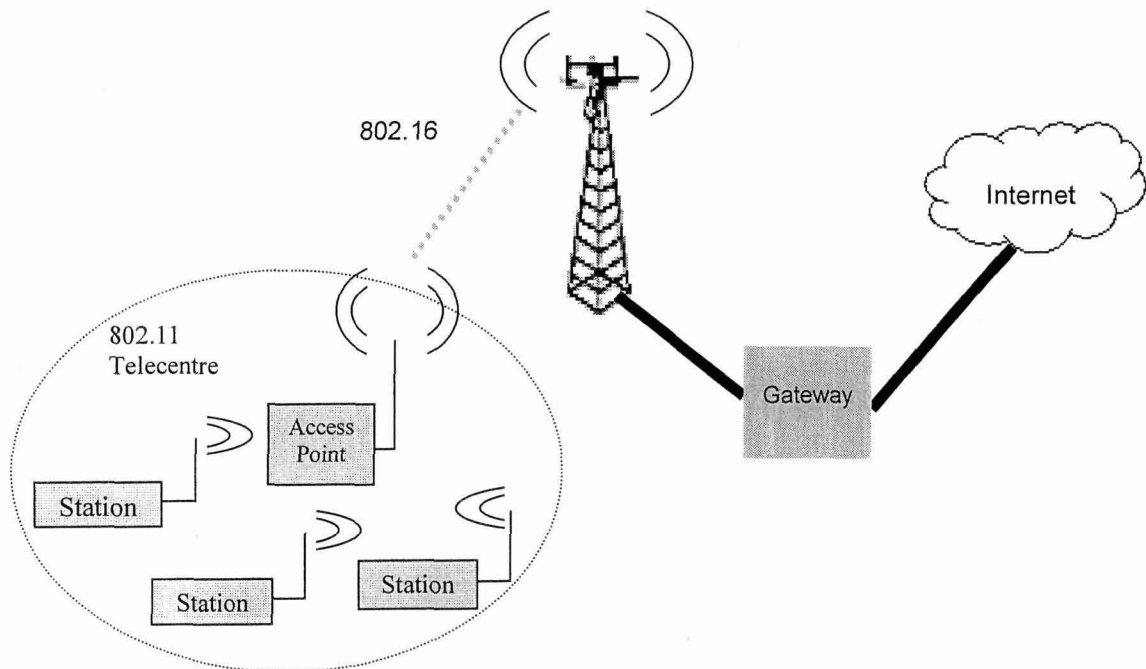


Figure 1.2 – WiMAX as backhaul for WiFi Hotspot

1.3 Scope of Research

This research takes a particular rural region in South Africa as a case study, and investigates the suitability of an IP based wireless access infrastructure for delivering Internet connectivity including multimedia applications such as VoIP and telemedicine.

Towards this goal, this research provides a dimensioning solution for the Nongoma scenario, taking into account specific traffic requirements. The traffic profile that is established for the telecentres comprises of various types of applications. These include data and multimedia traffic. The number of possible subscribers for the region is determined, and the data and multimedia traffic bandwidth requirements are predicted over an eight year period.

The quality of service (QoS) for the IP based solution is considered. This research investigates the ability of WiMAX to provide performance guarantees to delay sensitive applications, such as VoIP and telemedicine. This work endeavours to evaluate the performance of applications based on specific user requirements, in a rural IEEE 802.11 WLAN and IEEE 802.16 backhaul context.

The cost benefit of using IEEE 802.16 standard mesh mode to cover the entire geographic region of Nongoma is also considered. In addition, this research investigates the ability of mesh mode to extend the coverage of the WiMAX footprint.

1.4 Dissertation Outline

This chapter has provided the background and motivation of the research. Chapter 2 provides a brief description of the Nongoma region and presents the telecentre profiling and traffic analysis.

Chapter 3 examines the IEEE 802.11 and IEEE 802.16 standards. A concise description of the standards is provided with a focus on the 802.16 media access control (MAC) layer. A discussion on the operation and performance of 802.16 QoS characteristics is presented.

The literature survey results of 802.16 MAC layer investigations as well as the results obtained from the simulation experiments that were carried out are presented in Chapter 4. A theoretical result on adaptive modulation is also presented.

Chapter 5 looks at mesh mode in the IEEE 802.16 standard, as a cost-effective and robust means of communication as compared to point-to-multipoint (PMP) mode. Two scheduling methods used in mesh mode are also investigated.

Chapter 6 summarizes the conclusions and outlines additional work to be done in terms of providing QoS for the Nongoma region using wireless technology.

1.5 Contributions

The major contribution of this research is the performance evaluation of the WiMAX backhaul network for the rural Nongoma scenario (presented in Chapter 4).

The following paper was accepted at the Southern African Telecommunications Networks and Applications Conference (SATNAC) held in Mauritius in September 2007, as a result of this work:

- V Singh, R Sewsunker, "Performance Evaluation of WiMAX for Rural Backhaul,"

2 A RURAL CASE: THE NONGOMA REGION

2.1 Nongoma Region

Nongoma is situated in the western region of Zululand in the KwaZulu-Natal Province. It extends over an area of 2184 km² and has a population in excess of 237 000. It comprises of 19 wards, and is Zululand’s second largest Municipality in terms of population and area. Rural communities are located around Ward 19 in Nongoma which is regarded as their primary service centre. Ward 19 and its immediate vicinity are looked at as the coverage area for the proposed wireless network. The region is endowed with an unsurpassed mix of eco-attractions and is a potential tourist destination. The prospective economic and social growth for the area is immense.

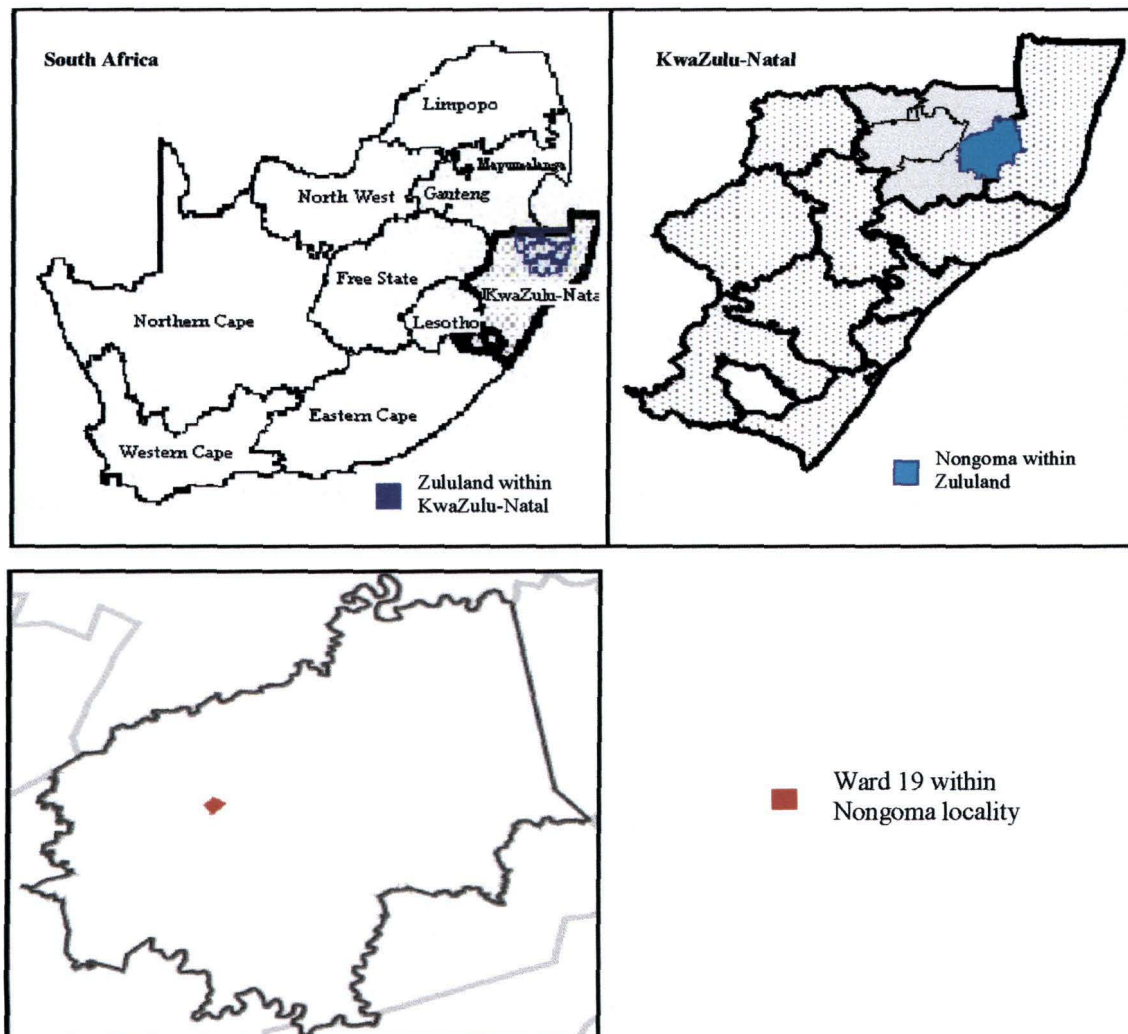


Figure 2.1 - Map of Region

The entire region has an under-developed telecommunications infrastructure, with majority of the communities having no access to a telephone network. Approximately 98% of Nongoma's residents live in rural settlements, which comprise of rural villages and rural scattered areas. Infrastructure in the region is minimal. According to the Nongoma Local Municipality Integrated Development Report (IDP) [3], only 6.25% of the population between the ages of 15-64 is economically active, thus the creation of employment opportunities for recently matriculated people in Nongoma is a high priority for the development of the local district. Thus the proposed telecentre solution holds the promise of providing a means for these communities to be able to participate in the global economy.

2.2 Telecentre Profiling

The network will comprise of four telecentres, a simple telecentre, a hospital telecentre and two clinic telecentres. Figure 2.2 below show the placement of the telecentre solution within Ward 19 of Nongoma. A fixed WiMAX, line of sight model is considered.

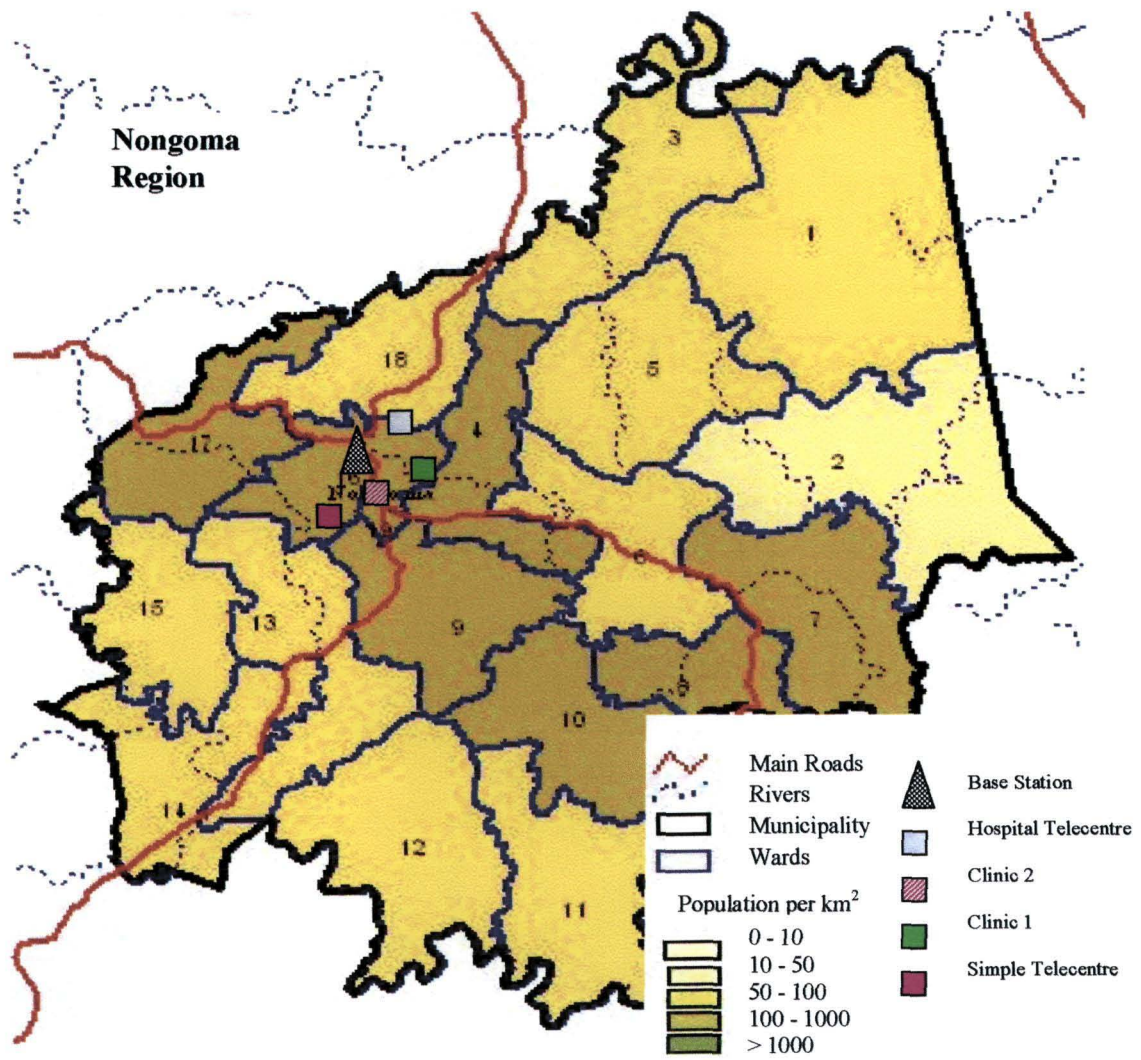


Figure 2.2 – Placement of Telecentre Solution in Ward 19, Nongoma

The hospital telecentre and clinic 2 are situated approximately 5 km from the base station. Clinic 1 is 10 km away from the BS, and 5km away from the hospital and clinic 2. The simple telecentre is approximately 15 km away from the BS.

All of the telecentres include Internet, VoIP and data applications. Internet applications comprise of two types:

- a) non-real time
- b) real time

The applications offered at the telecentres, which include email, web-browsing and telephony, act as tools for accelerating development and improving rural welfare.

The possible subscribers at the telecentres were calculated using equation 2.1 below [44]. Thus the possible subscribers at the simple telecentre are 2552 people. The hospital telecentre has 6380 possible subscribers and the possible subscribers at each of the clinic telecentres is 638 people.

$$S = P\alpha\beta \quad (2.1)$$

where:

S is the number of people that are possible subscribers;

P is the total population of Ward 19 and areas around its perimeter. This is estimated to be approximately 6.72% of the population of Nongoma;

α is the fraction of the total population that lives in and around Ward 19 with access to the telecentre or within walking distance of the telecentre. It is estimated to be 20% for the simple telecentre and 50% for the hospital telecentre (since there is only one hospital in Nongoma). The fraction of the population with access to the clinic telecentres is estimated to be 5%;

β is the fraction of the population of Ward 19 locality that are possible subscribers. This is approximately 80% of the population, excluding the very young children and very old people [3].

A literature review of rural telecentres in [5], [8] and [10] has given insight into the trends and the kinds of content and applications predominantly utilized by telecentres in developing countries such as South Africa. It also indicates that there is an overdue demand for communications in these areas, and the need for affordable communications is increasing. VoIP and e-mail are thus considered to be key applications. In addition a telemedicine service is offered at the hospital and clinic telecentres. This application holds the hope of providing revolutionary and innovative specialty care services to this developing rural community.

Based on these applications, a bandwidth estimate was developed. It includes current and growth values for the future. A time period of 2006-2014 was considered.

2.2.1 Data Bandwidth Requirements

The typical applications offered at each telecentre and their required bandwidth volumes were estimated over the eight-year period. The bandwidth measurements include uploads and downloads. The initial file size values for email and documents were selected from a telecentre case study in Australia [4], where typical file sizes for these applications were established. Values were given for typical minimum, average and maximum file sizes. For the South African case, the minimum file sizes were chosen. These initial measurements for 2006 are considered as a good starting point for an emerging rural market. The initial page size for web-browsing is obtained from a study on wireless networking in the developing world in [4].

The annual increases in volumes for each of the applications, and hence total bandwidth required, are dependent on a number of interacting factors. These include technological, social as well as economic factors [11]. Technologically, the complexity of web pages is increasing with additional graphics being included. Thus, a greater volume of data is being downloaded. Socially, as people begin to realize the value of the technology as well as its ease of use, they will utilize the telecentres more often. Thus bandwidth requirements increase as user needs change with time. As the economic position of the region improves, a faster growth in the number of subscribers will be seen.

An indication of how Internet user needs have changed in South Africa can be seen in the growth trend of the number of Internet users in South Africa [12]. The trend shows an initial slow uptake on the "arrival" of the technology. The subsequent years show a small "boom", with the following years pointing toward a maturity, showing smaller increases in number of users. Taking into account the technological, social and economic factors, it is assumed that a similar pattern will be followed in terms of bandwidth and data traffic increase for this network.

Table 2-2 below shows an example calculation of data transfer rates or bandwidth required per user, for the data applications in 2006:

Application/Content	Average Transaction Size (kB)	Data Transfer Rate (kbps)
Documents/Government Forms	50	40
Formatted Email Text	2	1.6
Web-browsing	62.5	50

Table 2-1: Data Transfer Rate Required for Average Sized Applications

The tolerance of users to download times or delays has been inferred from [13] as approximately 10 seconds. Therefore assuming that no data compression is used, a data transfer rate of 40kbps would be required to download a 50kB file in 10 seconds. The data transfer rates for the email and web-browsing applications was obtained similarly.

The average transaction or file size for each of the data applications is assumed to increase at the rates shown in Table 2-3 below. The growth values differ for the file sizes of the different applications. The email file sizes increase at approximately half the rate as the documents and web-browsing applications. These percentage growth values follow the trend of Internet user needs in South Africa. The file sizes of the documents exceed that of the web-browsing in 2012, for this case study. This is due to the fact that more government forms will be online in the years to come. Also, in the clinic and hospital telecentres, the transfer of medical documents, such as x-rays will increase with the years.

Year	% Growth	Documents (kB)	% Growth	Web-browsing (kB)	% Growth	E-mail (kB)
2006		50		62.5		2.00
2007	10	55	10	68.75	5	2.10
2008	30	71.5	30	89.375	15	2.415
2009	50	107.25	45	129.59	25	3.019
2010	75	187.69	65	213.83	35	4.075
2011	80	337.84	55	331.44	33	5.399
2012	72	581.08	53	507.10	25	6.749
2013	65	958.78	50	760.65	20	8.099
2014	55	1486.1	45	1103	18	9.557

Table 2-2: Average Transaction Size for Data Applications over the period 2006 – 2014

Thus, the data transfer rates for each of the applications were obtained. This is illustrated in the table 2-4 below:

Year	Documents (kbps)	E-mail (kbps)	Web-browsing (kbps)
2006	40	1.600	50
2007	44	1.680	55
2008	57.2	1.932	71.5
2009	85.8	2.415	103.68
2010	150.15	3.260	171.06
2011	270.27	4.319	265.15
2012	464.86	5.399	405.68
2013	767.03	6.479	608.52
2014	1189	7.64	882.35

Table2-3: Non-real time Application Data Transfer Rate

The following rule of thumb formula [4] was used to determine the bandwidth requirements for non-real time data:

$$\text{Bandwidth} = \text{Number of Users} * \text{Data Transfer Rate} * \text{Oversubscription Ratio} \quad (2.2)$$

where:

Number of Users is the number of PCs at a telecentre.

Data Transfer Rate is the bandwidth required. The maximum data transfer rate is used to support all of the non-real time applications, i.e. from 2006 to 2010, the data transfer rate of the web-browsing application was used. From 2011 to 2014, the data transfer rate of the documents was used.

Oversubscription Ratio is the overbooking factor for bandwidth allocation. (Assumes that not everyone is online at the same time, therefore 'peak' bandwidth has not been allocated to every PC) Typical ratio used is 0.4, given that number of users > 1[14].

The data application bandwidth requirements for each of the telecentres can be seen in the following graph:

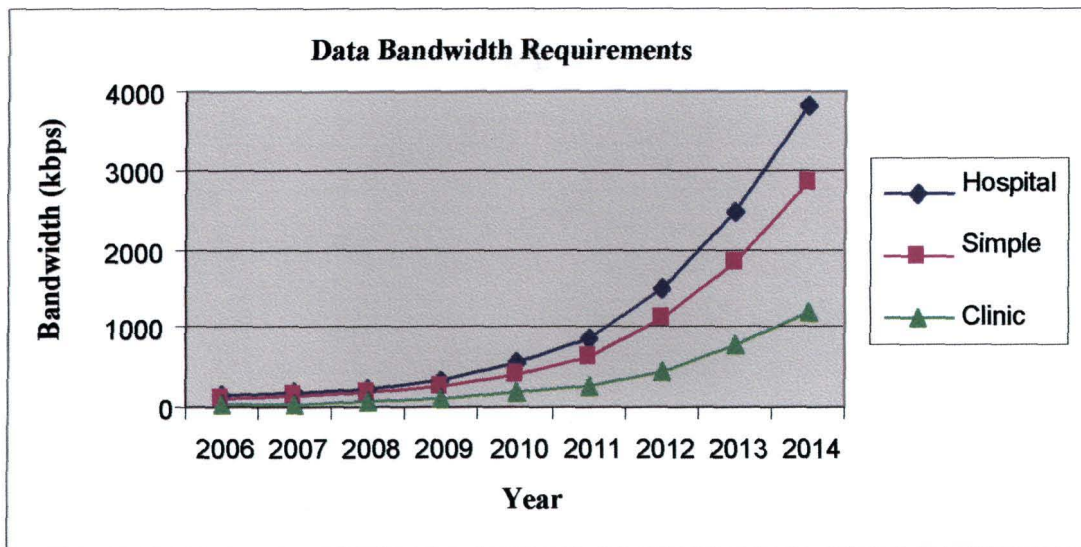


Figure 2.3 - Data Bandwidth Requirements of each Telecentre

Bandwidth (kbps)			
Year	Hospital	Simple	Clinic
2006	140	120	50
2007	176	132	55
2008	228.8	171.6	71.5
2009	331.76	248.82	103.68
2010	547.40	410.55	171.064
2011	864.86	648.65	270.27
2012	1487.6	1115.68	464.86
2013	2454.5	1840.86	767.03
2014	3804.5	2853.3	1188.9

Table 2-4: Source data: Data Bandwidth Requirements of each Telecentre

The values obtained can be considered conservative, when compared to [11], which draws a general conclusion that Internet traffic doubles every year.

2.2.2 Telemedicine and VoIP Bandwidth Requirements

Literature indicates that the data transfer rate or bandwidth required to meet medical diagnostic needs for telemedicine is 1.5Mbps [4]. It is assumed that the telemedicine data transfer rates

increase as shown in table 2-5. The bandwidth requirements increase over time to accommodate future complex diagnosis where the transmission of critical images is needed. Telemedicine also assumes no oversubscription; therefore the bandwidth required is equal to the data transfer rate.

Year	% Growth	Telemedicine (kbps)
2006		1500
2007	5	1575
2008	10	1733
2009	16	2009
2010	20	2415
2011	26	3039
2012	26	3829
2013	30	4977
2014	15	6471

Table 2-5: Telemedicine Bandwidth Requirements

The bandwidth requirements for VoIP were also dimensioned over the eight-year period.

The type of codec used to carry voice over the IP network plays a key role in determining the amount of bandwidth used. The different codec's have different call qualities and different bandwidths.

The codec chosen for the Nongoma solution is the G.729 codec. This codec is generally considered as the most cost-effective codec to provide the toll call quality voice calls at low bandwidth utilization [7]. Thus this codec is well suited to the rural Nongoma scenario.

Voice service generally has a lower oversubscription ratio than data and an oversubscription ratio of 0.17 is used in this research. This is approximately 1.5 times smaller than the ratio used for data.

By observing demand for public switched telephone network (PSTN) telephone access as well as taking into account social, technological and economic factors, an estimate of the growth for VoIP bandwidth requirements was obtained. The following equation was used to determine VoIP bandwidth requirements in bits/sec [15]:

$$\text{VoIP (bits/sec)} = \text{EPCHV} \quad (2.3)$$

where:

E is the predicted telecentre traffic in Erlangs.

P is the coding rate for PCM coded voice (64kbps).

C is a factor for the codec compression in bits/sec. (The 8kbps G.729 was used). A value of 0.125 was used.

H is a factor for the compressed real time transport protocol (CRTP) header compression and layer 2 headers. A value of 3 was used. This compresses the headers by approximately 66%

V is voice activity detection. A value of 0.6 was used.

The predicted telecentre traffic in Erlangs, **E**, was obtained as follows:

Taking the case of the simple telecentre for 2006; it is assumed that the average mean holding time for a call is 3 minutes. The number of possible subscribers (i.e. 2552 people) has been oversubscribed by a factor of 0.17, to obtain an active population of 425 people. It is assumed that each subscriber makes on average, 10 calls/month. Assuming an eight hour day for the simple telecentre, the traffic is 0.9Erlangs. The predicted telecentre traffic was obtained similarly over the eight year period for the hospital and clinic telecentres. It is assumed that these telecentres operate on twelve hour days.

Average traffic values are used in the planning, over a period of eight years in order to assess performance based on demand.

The calls/month for each telecentre increase at the rate shown in Table 2-6. The average mean holding time is assumed to remain constant for the time period specified.

Bandwidth (kbps)			
Year	Hospital	Simple	Clinic
2006	1656.9	132.8	1550.8
2007	1770.1	146.41	1631.0
2008	1984.3	188.89	1805.2
2009	2369.7	270.09	2114.8
2010	2993.2	436.28	2584.4
2011	3944.6	679.53	3311.0
2012	5363.5	1151.2	4295.9
2013	7486.6	1882.1	5747.1
2014	9590.8	2900.3	6915.9

Table 2-8: Source data: Estimated Total Bandwidth Requirements for each Telecentre

2.2.4 Infrastructure at Telecentres

The initial deployment for infrastructure at the telecentres is illustrated in the table below. The proposed numbers for the infrastructure at the telecentres were chosen based on a survey of telecentres around the world. An Australian study [4], found that 75% of 387 telecentres surveyed have four or more PCs, serving people within 30 minutes walking distance of the telecentre. Telecentre deployments in India [5] have at least five computers serving approximately 1000 families within a 3 km radius of the telecentres. A study closer to home [6], in which community telecentres in Africa were investigated, showed an example of a model business plan for a telecentre in its first year. It shows that five computers and six telephones are sufficient to serve approximately 20000 people that live within 30 minutes walking distance of the telecentre.

The number of PCs and VoIP phones for the South African case was chosen based on these studies, and the number of possible subscribers at each of the telecentres. (The calculation for the number of possible subscribers is described below). The Hospital has eight PCs, because it is the only hospital in the region, and it also hosts a telemedicine application. The VoIP phones were chosen such that people in the region don't have to queue to use the service. The VoIP requirements are discussed in more detail in section 2.2.2.

Telecentre Type	Number of PC's	Number of VoIP Phones
Simple	6	4
Hospital	8	6
Mobile Clinic	2	1

Table 2-9: Infrastructure at Telecentres

The network infrastructure at the telecentres has been oversubscribed, in order to accommodate all possible subscribers.

2.2.5 Summary

The predicted traffic measurements for the Nongoma solution are based on prior research in the various areas. The initial take-up for the data requirements is primarily based on minimum bandwidth measurements from an elaborate study on telecentres [4]. The growth is based on Internet user needs in South Africa, as well as the technological, social and economic issues [12].

The number of VoIP calls is estimated from the literature [45]. Its growth pattern follows PSTN growth trends in South Africa based on ITU estimates [12], as well as the technological, social and economic issues mentioned.

The telemedicine bandwidth requirements are also obtained from the literature [4].

The predicted file sizes for the FTP, email and web-browsing application will be used in the simulation investigations in Chapter 4. The traffic load for the different years in the simulation scenarios will be increased by increasing the file sizes of these applications. A profile is setup in the simulation scenarios for the VoIP application where the predicted call rates from Table 2-7 are implemented. For the telemedicine application, the video frame sizes are increased according to the data rates in Table 2-6. Thus the simulation scenarios are carried out using the Nongoma traffic profile over the eight year period.

3 OVERVIEW OF IEEE 802.11 AND 802.16 STANDARDS

3.1 IEEE 802.11 Standard

3.1.1 Introduction

The first 802.11 standard was ratified by IEEE in 1997 which specified data rates of 1Mbps and 2Mbps. Subsequent to the first ratification, there have been a number of amendments to the standard, most notably the 802.11b, 802.11a and 802.11g standards, specifying data rates of up to 11Mbps for 802.11b and 54Mbps for 802.11a and g.

Like any 802 protocol, the IEEE 802.11 standard has been designed to comprise of Physical and Medium Access Control (MAC) layers. The standard has also been designed for extensibility, and there have been many efforts to extend its functionality, by means of QoS extensions (802.11e), security extensions (802.11i) as well as mesh extensions (802.11s) to the base standard. There have also been efforts to increase the data rates (802.11n) for a faster standard, as well as enhancements to extend the applicability of 802.11 (802.11d, 802.11h and 802.11j) to other frequencies, such as 4.9GHz in Japan. Future improvements also include support for faster handoffs (802.11r), thus enhancing application performance for mobile devices.

The IEEE 802.11 standard doesn't specify compliance, interoperability or compatibility; consequently the WiFi alliance was founded. It is a trade association that provides a test suite for the 802.11 standard, thus providing compatibility and interoperability between different products.

3.1.2 Topologies

An 802.11 wireless LAN defines the Basic Service Set (BSS) as the standard building block. A BSS is defined as a group of two or more nodes. One topology is an Independent Basic Service Set (IBSS) or an Ad-Hoc Network. In this type of network, nodes communicate directly with each other. Another topology is the Infrastructure Basic Service Set. This type of network comprises of a BSS and an access point. Nodes now only communicate with the access point and not directly with each other. The access point may also provide connection to a distribution system, which is the means by which an access point communicates with another access point. The distribution system is considered as the backbone of the WLAN, and it can be either a wired or wireless system.

Communication among access points to forward traffic from one BSS to another via a distribution system is referred to as an Extended Service Set (ESS). This is illustrated in the figure below:

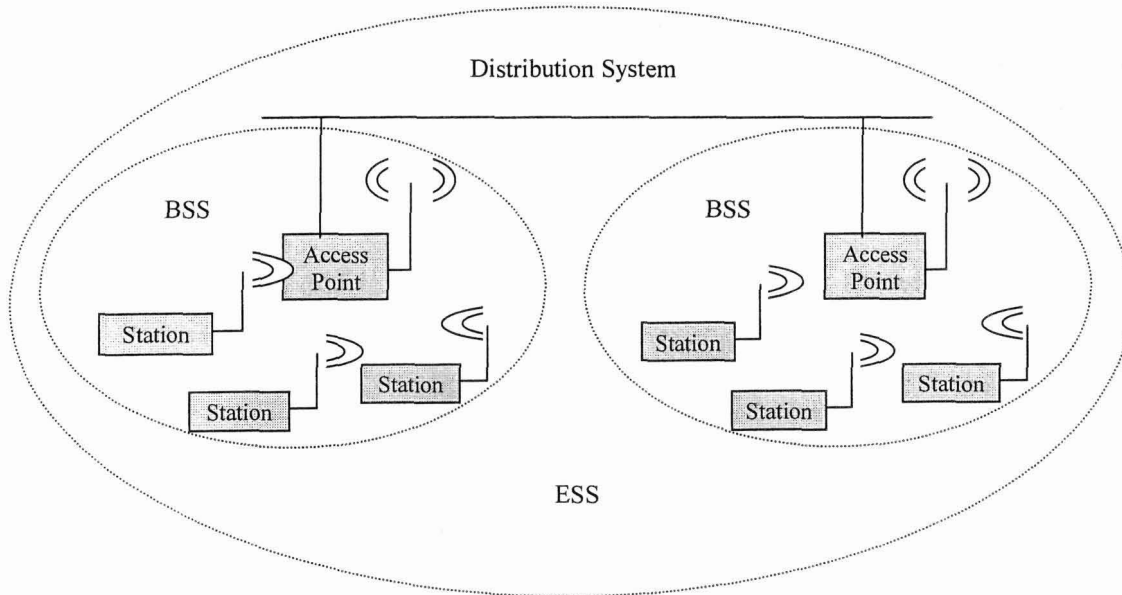


Figure 3.1 – IEEE 802.11 Extended Service Set (ESS)

3.1.3 Physical Layers

The physical layers that were originally designed included direct-sequence spread spectrum (DSSS), frequency-hopping spread spectrum (FHSS) and Infrared pulse modulation. The most recent addition to the physical layer modes is orthogonal frequency division multiplex (OFDM). The sophisticated OFDM technique was made part of the 802.11a and g specification allowing data rates to increase to 54Mbps.

The figure below illustrates the scope of the standard:

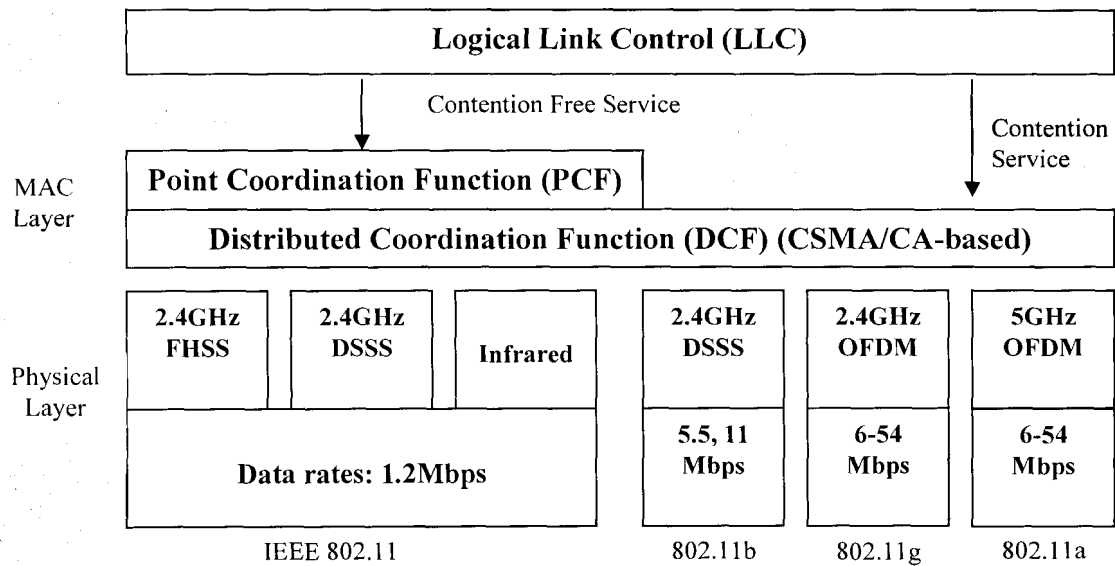


Figure 3.2 - IEEE 802.11 Standards

3.1.4 Medium Access Control Layer

One common MAC layer has been defined for all 802.11 physical layers. Two types of access methods are defined by the IEEE 802.11 standard; the distributed coordination function (DCF) or carrier sense multiple access with collision avoidance (CSMA/CA) and the point coordination function (PCF).

CSMA/CA is designed to detect and avoid collisions with other transmitting devices. In this type of network, a transmitting device senses the medium to determine whether it is in use. Transmission only occurs if the channel is idle for a given time interval, i.e. the distributed inter-frame space (DIFS). If the channel is in use or active, transmission is deferred until the current transmission is complete. The node then uses the Backoff Exponential Algorithm defined by the standard. A time interval (backoff window) is randomly selected and this timer is decremented every time it senses the channel to be idle. The node is allowed to transmit when the backoff timer reaches zero. The transmission process is complete when the receiving node issues an acknowledgment (ACK) frame to the transmitting node, to confirm that the data packet was received intact.

The point coordination function mode is optional in the IEEE 802.11 MAC, and may be used to implement time-sensitive traffic such as voice and video. In PCF mode, the access point issues a

polling request to each node for data transmission. Thus every node is given a turn to transmit in a predetermined manner and a maximum latency is guaranteed, so PCF is a contention-free protocol.

A problem specific to WLANS is the “hidden node” issue, which occurs when nodes are hidden or unable to “hear” each other (e.g. on opposite sides of an access point). A collision occurs when one node incorrectly accesses the channel, thinking it is clear, when in fact, the other node is transmitting. This problem is addressed by a virtual carrier sense mechanism defined by the standard. A transmitting node will send a request-to-send (RTS) packet to the receiver. The receiver will respond with clear-to-send (CTS) packet if the channel is idle. All nodes within “hearing” distance of the transmitting or receiving node detect the RTS/CTS exchange and will defer transmission until the end of the ongoing transmission.

Other MAC layer functionalities include, support for fragmentation of data packets; i.e. breaking up large packets into smaller packets. (Fragmentation occurs in the event of congestion and interference). Support for dynamic rate shifting in which the data rates are automatically adjusted depending on the noise conditions of the wireless channel. There is also support for dynamic frequency selection (DFS) and transmit power control, which allows a node to optimally pick the right radio channel and select the power setting.

3.2 IEEE 802.16 Standard

3.2.1 Introduction

Broadband wireless access (BWA) is generally regarded as the best means of providing cost-effective, integrated voice, video and data services. The work entailed in IEEE Project 802.16 forms part of the research activity in the BWA area. 802.16 is working group number 16 of IEEE Project 802. The IEEE 802.16 standards specify the air interface for a wireless metropolitan area network (WMAN), supporting multimedia services. Other BWA research activities include work by the European Telecommunications Standards Institute (ETSI). IEEE 802.16 standardization has been done in collaboration with the ETSI High Performance Metropolitan Area Network (HiperMAN) standards.

Thus the WiMAX Forum was developed. The Forum assures the interoperability and conformance of broadband wireless access equipment based on the IEEE 802.16 and ETSI HiperMAN standards. The WiMAX Forum has identified certain operational frequency bands for the standards. These are

the licensed 3.3-3.8GHz, 2.5-2.69GHz and 10-66GHz bands as well as the unlicensed 5.25-5.85GHz bands.

The IEEE 802.16 standard details the physical and medium access control (MAC) layers of the OSI model. An important feature of the 802.16 standard is that it defines a common MAC layer that supports three PHY layer specifications; a single carrier option and two OFDM based options. This section aims to distil the important features of the standard.

3.2.2 Overview of IEEE 802.16 Family of Standards

- The "Air Interface for Fixed Broadband Wireless Access Systems" was approved in December 2001, known as IEEE 802.16-2001. The standard operated in the licensed 10-66GHz range, addressing LOS connections. This initial version of the standard supported a point to multipoint network topology.
- The next significant version was the IEEE 802.16a standard, approved in January 2003, and designed to operate in the 2-11GHz frequency range for licensed and licensed exempt bands. It included support for NLOS operation. Additional physical layer specifications were included. A notable modification was the addition of an optional mesh mode.
- IEEE 802.16d standard combines the 802.16 and 802.16a standards. It also specifies system profiles that are intended to simplify system setup. The IEEE 802.16-2004 standard replaces 802.16, 802.16a and 802.16d.
- The IEEE 802.16e standard operates in the licensed 2-6GHz frequency band, addressing both fixed and mobile operations. It supports vehicular mobility of up to 100km/h.
- IEEE 802.16.20 is a proposed Mobile Broadband Wireless Access Standard. Unlike 802.16e, which supports local/regional mobility, handoff and roaming, 802.16.20 supports global mobility, hand-off, and roaming. It can sustain vehicular mobility of up to 250km/h and operates in the licensed 3.5GHz band.

3.2.3 Architecture

The standard specifies two modes of operation, point to multipoint (PMP) and mesh mode. In a PMP topology, traffic only occurs between the base station (BS) and the subscriber station (SS).

Transmission on the uplink (from SS to BS) is time division multiple access (TDMA) based and transmission on the downlink (from BS to SS) is time division multiplex (TDM) based.

The mesh mode topology allows traffic to be routed through the subscriber stations. Thus the SSs communicate directly with each other. Each subscriber station can act as a router to relay packets for its neighbours. Mesh topology can provide robust communications, as there are many options for routing traffic. Several WiMAX connections can be set up at low cost. These advantages are achieved at the cost of complexity, as mesh mode entails more difficulty.

3.2.4 Physical Layer

The focus of this research is on the MAC layer, but since the Physical and MAC layers depend on each other, some understanding of the operation of the Physical layer is imperative. The Physical layer performance is reliant on technologies such as orthogonal frequency division multiplex (OFDM), time division duplex (TDD), frequency division duplex (FDD) and quadrature amplitude modulation (QAM). A brief review of these technologies follows.

3.2.4.1 OFDM

OFDM is not a new technology; it dates back to the 1960's, where it has been used successfully in digital subscriber line (DSL) modems and cable modems. OFDM is based on frequency division multiplexing (FDM), which uses multiple frequencies to simultaneously transmit multiple signals in parallel [16]. The difference is that OFDM uses orthogonal frequencies to space the carrier signals much closer together, until they are actually overlapping. The frequencies are orthogonal, thus allowing overlapping without interference. The effect is increased throughput and efficiency. The figures below show how this digital encoding and modulation scheme achieves higher bandwidth efficiency even in adverse channel conditions:

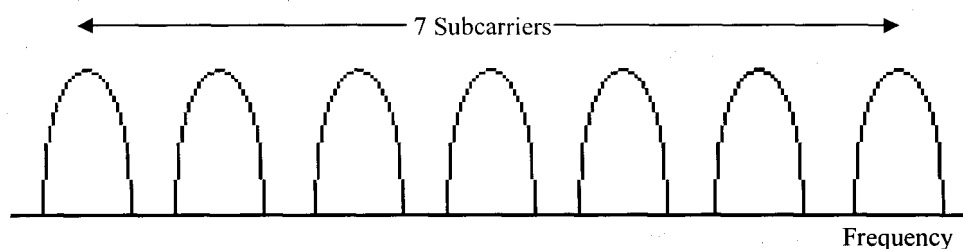


Figure 3.3 – FDM with Seven Subcarriers

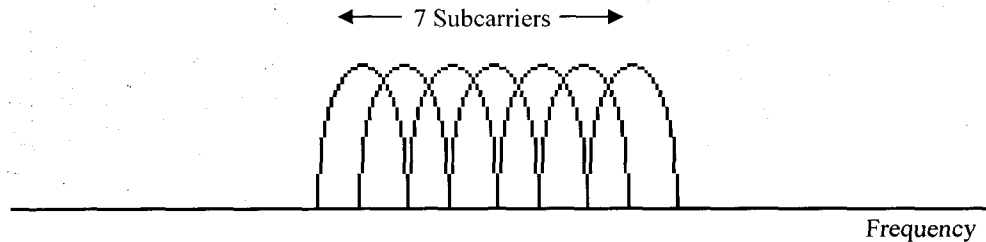


Figure 3.4 – OFDM with Seven Subcarriers

OFDM technology can be implemented with 256 sub-carriers, or by using orthogonal frequency division multiple access (OFDMA) with 2048 sub-carriers. Not all sub-carriers are used to carry data in a WiMAX system. There are null sub-carriers to the right and left of an OFDM channel. These are guard carriers which mitigate interference with adjacent channels. Some of the sub-carriers are also pilot carriers, used for synchronization. The sub-carriers can be divided into subchannels. A subchannel is a subset of carriers from the set of total available carriers [17]. The subchannels separate the data into coherent streams on the downlink, and are used for multiple access on the uplink.

3.2.4.2 TDD and FDD

There is support for both time division duplex (TDD) and frequency division duplex (FDD) operation in WiMAX systems. TDD allows the system to transmit and receive on the same channel but at different times, thus affording a more flexible system. The TDD solution has the advantage of being able to dynamically allocate bandwidth depending on traffic requirements. This technique is employed in WiMAX systems utilizing the license-exempt bands.

FDD allows transmit and receive functions to occur simultaneously on different frequencies. This means that the downlink (DL) and uplink (UL) subframes occur simultaneously. This legacy duplexing technique does not require the overhead of a guard time and has been designed for symmetrical traffic, such as voice. It is used in licensed solutions.

3.2.4.3 Modulation

IEEE 802.16 supports four types of modulation schemes; binary phase shift keying (BPSK), quadrature phase shift keying (QPSK), 16 QAM and 64 QAM. Each of these modulation schemes uses mandatory forward error correction techniques, i.e. Reed Solomon and convolutional coding.

Thus the following modulation rates are obtained:

Modulation	Coding Rate
BPSK	$\frac{1}{2}$
BPSK	$\frac{3}{4}$
QPSK	$\frac{1}{2}$
QPSK	$\frac{3}{4}$
16 QAM	$\frac{1}{2}$
16 QAM	$\frac{3}{4}$
64 QAM	$\frac{2}{3}$
64 QAM	$\frac{3}{4}$

Table 3-1: Modulation Coding Rates

The modulation rates are closely linked to the data throughput of a system, where:

$\frac{1}{2}$ means that 1 out of 2 bits are used for data, while the rest are used for error correction.

$\frac{3}{4}$ means that 3 out of 4 bits are used for data, while the rest are used for error correction.

IEEE 802.16 systems are capable of choosing the modulation type based on channel conditions. Subscriber stations that are further away from the base station use more robust modulation schemes such as BPSK so as to maintain the quality of the connection, whereas subscriber stations that are close to the base station can increase their throughput by using more efficient schemes such as QAM. Thus, the use of adaptive modulation allows WiMAX systems to achieve optimal efficiency. The diagram below illustrates this concept.

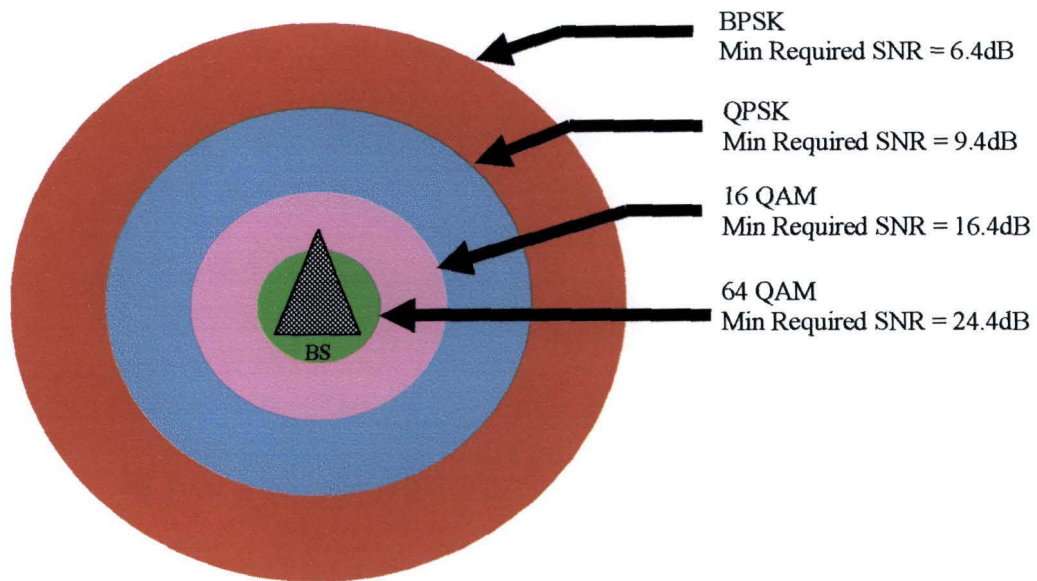


Figure 3.5 – Adaptive PHY Modulation

3.2.4.4 Framing

Transmission over the air-link divides the time into frames. There are UL and DL subframes. The diagram below shows the structure of these frames using FDD duplexing.

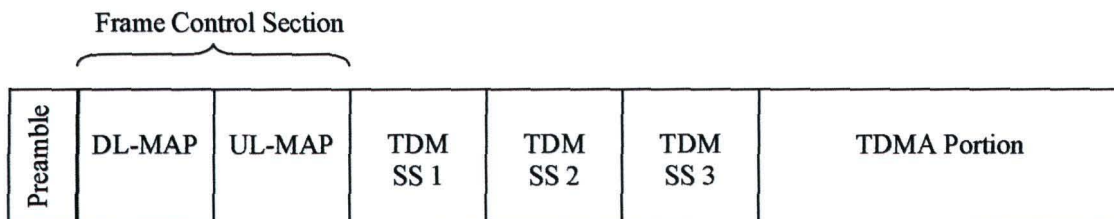


Figure 3.6 – Downlink Frame Structure

The downlink subframe consists of a DL preamble, used for synchronization. This is followed by the frame control section, where the DL-MAP and UL-MAP are found. The DL-MAP informs all SSs when to listen for transmissions destined for them in the current frame [18]. The UL-MAP informs the SSs of their transmission opportunities for a specified time in the future.

The frame control section is followed by a TDM portion. This is where the data (MAC protocol data units (PDUs)) is transmitted to a SS using a specific burst profile that is identified by the downlink interval usage code (DIUC). The burst profile refers to the physical layer (PHY)

parameters such as modulation and forward error correction (FEC). The TDMA portion of the DL subframe is optional, to allow better support for half-duplex SS's.

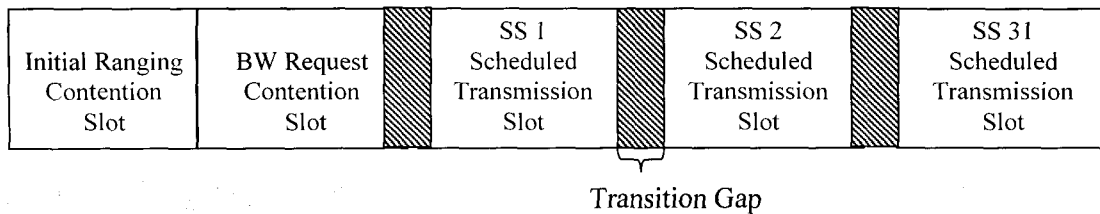


Figure 3.7 – UL Frame Structure

The UL subframe begins with an initial ranging and a bandwidth request contention slot. These are comprised of minislots, and contending SSs use the truncated exponential backoff algorithm to decide which minislot to transmit in. The rest of the UL subframe comprises of transmissions from different SSs. These transmissions are separated by SS transition gaps, which the BS uses to resynchronize to different SS transmissions.

3.2.5 Medium Access Control (MAC) Layer

The MAC layer interprets data between the upper data link layer and the lower physical layer. The scarcity and high cost of resources in the air-interface link, results in the need for the MAC layer to efficiently manage these resources. The 802.16 MAC is a scheduling MAC, with an access scheme that is contention-less based. The scheduling nature enables high bandwidth utilization, as opposed to contention based access schemes, which suffer from the problem of low bandwidth efficiency and high collision rates. The diagram below shows the layering of the standard, in particular, the MAC sublayers.

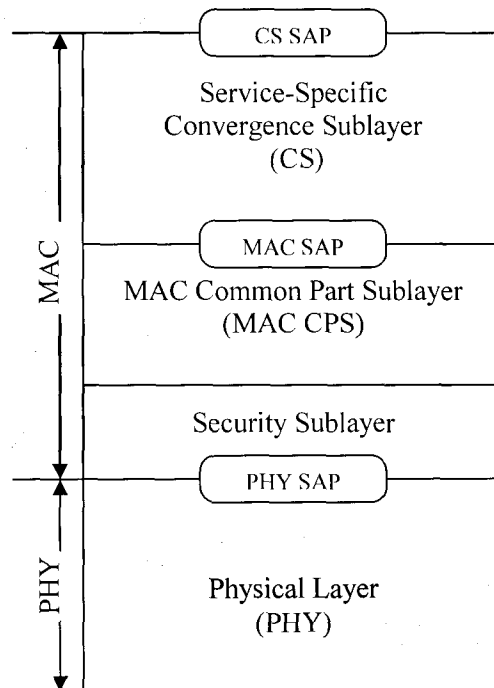


Figure 3.8 - Layering of IEEE 802.16 Standard

3.2.5.1 MAC Sublayers

- There is a service access point (SAP) that interfaces each sublayer. The Service Specific Convergence Sublayer (CS) transforms or maps the data received from the higher layer into MAC service data units (SDUs). The transformation or mapping process also includes classification of the packets, which are then assigned to a given connection, with a given set of QoS parameters. The connections are identified by a connection ID (CID).
- The MAC Common Part Sublayer (CPS) is the heart of the MAC. It receives the MAC SDUs through the MAC service access point, and is responsible for transforming the SDU into a MAC PDU, by adding the headers, and optional cyclic redundancy check (CRC), for delivery of the packet through the air-interface. It provides the MAC layer's core functions, which include system access and connection management, bandwidth allocation and scheduling of data, as well as applying QoS to connections. The subsequent sections will provide more detail on these functions.

- The MAC Privacy Sublayer (PS) layer provides authentication, secure key exchange, and data encryption. [19] Thus subscribers are provided with privacy across the network. The MAC header indicates whether encryption has been included or not.

3.2.5.2 IEEE 802.16 MAC Connections

The IEEE 802.16 MAC is connection-oriented, where the IEEE 802.16 standard defines a connection to be a unidirectional flow of data, established on the downlink or uplink [19]. All traffic is carried on connections, even for flows supported by connectionless protocols, such as IP. Each connection that is established is classified by a 16 bit connection identifier (CID).

There are signalling or management connections which carry only management messages and data or transport connections which carry data traffic. When a SS begins its network entry process, the BS allocates to it, two mandatory and one optional management connections. Each of these connections has associated QoS parameters, i.e. they carry messages of different lengths and time constraints. They are as follows:

- The Basic Connection is used for short MAC messages, constrained by time.
- The Primary Management Connection transmits messages that are not as time critical, for example connection setup messages.
- The Secondary Management Connection is optional and used to deliver messages using standard based protocols, such as, Dynamic Host Configuration Protocol (DHCP), Simple Network Management Protocol (SNMP) and Trivial File Transfer Protocol (TFTP).

The transport connections (not to be confused with Transport layer connections found in the Open Systems Interconnection (OSI) model) are instituted after the management connections have been established. These connections are set up according to the scheduling services supported and associated QoS parameters.

3.2.5.3 Bandwidth Allocation

The IEEE 802.16 standard uses a request/grant mechanism to control BW allocation. The BS responds to the SSs request for BW by allocating the appropriate number of transmission time-slots on UL-MAP messages. The SS receives the BW grants by decoding the UL-MAP messages. The SS can accept one aggregated BW grant for all connections, referred to as a grant per SS (GPSS), or

it can accept the BW grants separately for each connection, referred to as a grant per connection (GPC).

The former method is more complicated as the SS manages the bandwidth (BW) allocated to each connection. If a situation arises where a connection requires more BW than anticipated, the SS can steal BW from a lower QoS connection. Thereafter, the SS can request more BW to meet its new requirements. In addition to requesting bandwidth for each individual connection, the GPC class of SS's has to request additional bandwidth for any unanticipated radio link control requirements, thus making it less efficient than GPSS systems.

3.2.5.4 Bandwidth Requests and Polling

SSs request bandwidth by using a bandwidth request header or by a piggyback request. Piggyback requests refer to requests that are sent with a PDU. Bandwidth request MAC PDUs consist of only the bandwidth request header and no payload. The BS allocates bandwidth request opportunities or polls the SSs, so that they maybe able to send a bandwidth request. The polling is done in one of three forms;

- a unicast poll, where SSs are polled individually
- a multicast poll, where a group of SSs are polled
- a broadcast poll, where all SSs are polled

Multicast and broadcast polling only occur if there is insufficient bandwidth to individually poll each SS. Only the SSs that need bandwidth reply. This type of bandwidth request mechanism can incur collisions, so the contention resolution algorithm is applied to determine the timeslot in which to transmit the bandwidth request.

3.2.5.6 Scheduling Services

There are four uplink scheduling mechanisms defined by the standard; unsolicited grant service (UGS), real-time polling service (rtPS), non-real-time polling service (nrtPS) and best effort (BE). Each connection is associated with one of the service types. These service types are further defined below:

- UGS is designed to support real-time data streams, such as VoIP, that generate fixed sized data packets periodically. The BS allocates fixed-sized data grants periodically. So applications

using this service don't request bandwidth for each connection. This eliminates additional overhead and latency.

- rtPS is designed to support applications, such as MPEG video, that generate variable sized data packets periodically. Connections associated with this service type use the periodic unicast polls sent by the BS, to request bandwidth, based on current real-time needs and packet sizes.
- nrtPS supports data streams with variable sized data packets that have no specific delay requirement, such as FTP. Bandwidth requests for connections associated with this service type use unicast or broadcast polls. The unicast polls though, for this service type are not necessarily periodic.
- BE service is designed for Internet type traffic; traffic with no minimum service level. Broadcast polls are used to request bandwidth.

There are various QoS parameters associated with the scheduling services. The most pertinent being the minimum reserved traffic rate (MRTR), which is defined as the minimum rate in bps, reserved for a flow.

3.2.6 IEEE 802.16 QoS

Figure 3.6 describes the QoS mechanism used in the IEEE 802.16 standard. It is further described below:

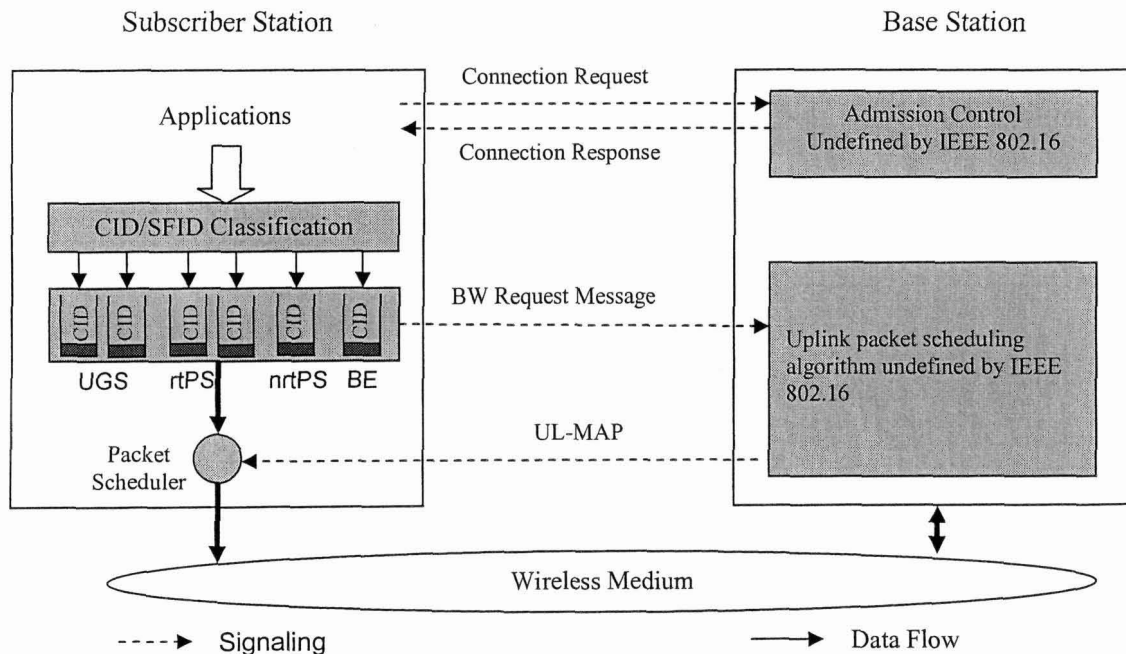


Figure 3.9 – IEEE 802.16 QoS Mechanism [20]

- At the SS, when a data packet (SDU) arrives from the higher layer, the classification process is the first process that it undergoes. This is where it is assigned a classification ID (CID) and service flow ID (SFID). The SDU is encapsulated into a MPDU and directed to the appropriate data queue.
- A bandwidth request is sent on arrival of the packet at the data queue. This only occurs if automatic repeat request (ARQ) is not enabled.
- The BS receives the bandwidth request and places it in a queue in the “MAP generator” module. The BS then issues a grant for the bandwidth request. The bandwidth grant is comprised of information elements (IE) and is encapsulated in an UL-MAP.
- The SS receives the UL-MAP and decodes it, to extract the information elements. This information is then passed onto the scheduler within the SS (either GPC or GPSS). The scheduler directs the IE’s to the appropriate connection in the data queue.
- The SS uses the bandwidth grant to send the MPDU that is in the data queue. It is received by the BS, converted to an SDU, and forwarded to the MAC layer.

The schedulers at the BS and SS may use any queuing discipline that is able to provide fair queuing, for the different traffic types. These mechanisms work together to ensure that there is traffic prioritization and that the data is transmitted within its QoS parameters.

The reason for QoS over WiMAX, is to ensure that the high priority traffic such as voice and video can be sustained in this wireless medium, and that it can be sustained well. One of the main factors that contribute to voice and video quality degradation is delay. The International Telecommunication Union (ITU) suggests an end-to-end delay of approximately 100ms for interactive video [23], and 350ms for VoIP [24]. This research uses a delay budget of approximately 100ms for video and 150ms for voice.

One approach to measure the performance of the voice connections is by using a numerical measure of the perceived quality of the voice connections, i.e. the mean opinion score (MOS). Determining the voice quality entails using the ITU's and the Telecomms Industry Association (TIA) approved E-model. The E-model defines the R-value as a measure of voice quality. The R-value is a transmission factor that comprises of a base quality value, and impairment factors due to echoes, distortion and transmission equipment.

The R-value can be mapped to an equivalent MOS value. The table below shows how the E-model and MOS scales compare with regard to user perceived quality.

E – Model R - value	MOS	User Satisfaction
90	4.3	Very satisfied
80	4.0	Satisfied
70	3.6	Some users dissatisfied
60	3.1	Many users dissatisfied
50	2.6	Nearly all users dissatisfied
0	1.0	Not recommended

Table 3-2: Comparison of E-model and MOS Rating Systems [25]

The above-mentioned QoS mechanisms are further investigated in Chapter 4, and the voice delays obtained are mapped to a MOS.

3.2.7 ARQ and FEC

ARQ is an optional mechanism that is employed in the MAC layer to make data transmission reliable in an unreliable medium, i.e. the wireless air-link. When a connection is established, it is determined whether ARQ will be used, and the ARQ parameters are specified. ARQ is only used for data applications, because the mechanisms it employs can lead to variable delays that are intolerable for real-time applications. There are generally three ARQ mechanisms that are employed for wireless transmissions. These include; Stop and Wait, Feedback (Go Back-N) and Selective

Repeat. Each technique has its advantages and disadvantages. The feedback and selective algorithms are based on the sliding window mechanism, and are generally the techniques implemented in the IEEE 802.16 standard's MAC layer.

The feedback (Go back-N) mechanism works by retransmitting all PDUs, starting from the first MPDU that was lost. The PDUs that are to be retransmitted are stored in a retransmission window. As acknowledgements are received for the PDUs, the window is shifted. The disadvantage of the feedback mechanism is that it is very inefficient in terms of bandwidth. Some PDUs that were received correctly may be retransmitted unnecessarily.

The selective repeat technique only retransmits the lost MPDU. As a result it is more bandwidth efficient.

ARQ system parameters include block size and window size, where a certain number of equal sized blocks comprise a MSDU. The IEEE 802.16 standard defines the window size to be the maximum number of unacknowledged blocks at any given time.

Forward error correction (FEC) is also an error correction technique that is used to make data transmission reliable in an unreliable link, such as the air interface used in WiMAX technology. The FEC techniques specified by IEEE 802.16 standard are a Reed Solomon outer code and a convolutional inner code. FEC works by adding redundant bits to the data, to help recover the incorrect bits. The methods used by FEC can meet the stringent delay requirements of real-time applications, and has thus been suggested to be used for these applications, but FEC has its disadvantages as well. When channel error rate is high long FEC codes are necessary, thus the decoding error rate increases quickly. Thus a hybrid-FEC scheme has been developed that overcomes the disadvantages of ARQ and FEC. This error correction mechanism is suitable to use for real-time applications.

3.3 Recommended Solution

Based on the features of the technologies, recommendations are suggested below and are several of these investigated through simulation. These results are documented in subsequent chapters.

3.3.1 WLAN Access

The topology recommended for WLAN access at the telecentres is the Infrastructure Basic Service Set. This allows one telecentre to connect to the other via the distribution system, i.e. the WiMAX backhaul. The highest data rate of 54Mbps is recommended to transport the voice and data services mentioned in Chapter 2.

A key issue in wireless networking is interference, which can lead to decreased throughput and increased latency. The IEEE 802.11a specification minimizes interference because it operates in a less crowded spectrum offering 23 non-overlapping channels, as compared to 3 non-overlapping channels offered in the b and g specification.

Although PCF has been designed to support multimedia traffic, it has limitations which lead to poor QoS, thus DCF is recommended, to maximise channel resources.

3.3.2 WiMAX Backhaul

The licensed, fixed LOS model is recommended for WiMAX backhaul to optimise transmission of voice and data services. Voice traffic is a priority, thus the FDD duplexing technique is recommended. Use of adaptive modulation is also recommended to achieve optimal efficiency in terms of delay and throughput.

Although the GPSS method of bandwidth allocation is more complicated, it is more efficient and scalable. This is desirable for a backhaul solution. In the Nongoma scenario, GPSS allows for 'QoS' on the WLAN access side.

It is recommended that the various scheduling mechanisms be used as defined by the standard, for the different applications. This is to maintain QoS for the data and multimedia traffic.

If the ARQ mechanism is implemented, the selective repeat technique is recommended as it is the most bandwidth efficient.

3.4 Summary

This Chapter presented an overview of the IEEE 802.11 and IEEE 802.16 standards. The physical and medium access control layers for both the standards were examined. The IEEE 802.16 QoS mechanism was emphasised. In addition, a recommended solution for implementation of the

WLAN access and WiMAX backhaul was presented. These recommendations are further investigated in Chapter 4.

4 SIMULATION RESULTS

4.1 Introduction

The focus of this research is to investigate the performance of an IP based network, using user specific parameters. This task involves identifying suitable performance metrics as quality indicators, as well as studying the performance of the IEEE 802.16 MAC whilst supporting the predicted traffic mix for the Nongoma region.

The implementation of a WiFi/WiMAX network is a complex task, involving the configuration of many parameters. A simulation tool was chosen to undertake this study. This chapter discusses the simulation environment used and presents the simulation results and analysis.

The source data for all the results that have been plotted are tabulated in Appendix A.

4.2 Assumptions

Several assumptions were made for the simulation analysis. The model is for a fixed wireless PMP network. The OPNET simulation model assumes LOS between nodes, as physical layer losses and physical effects such as fading, interference and ranging are not included in the model. OPNET does however include a PDU dropping probability to emulate the PHY losses of the wireless channel. This parameter is taken into account in one of the simulation scenarios considered.

All of the simulation scenarios are such that, each of the WiMAX SSs are located at various distances from the BS, as described in Chapter 2. According to the IEEE 802.16 standard, the closer a SS is to the BS, the more efficient the modulation scheme. In order to make the scenario as realistic as possible, the SSs closest to the BS, i.e. the Hospital and Clinic 2, use the 16 QAM modulation scheme, while the Simple telecentre and Clinic 1 use the QPSK modulation scheme.

Regarding the network traffic, the simulations focus on using the predicted traffic for the region, as presented in chapter 2. The BS is connected to a server that supports all of the application types throughout the simulations.

4.3 Simulation Environment

OPNET Modeler 11.5A was the simulation tool used to implement the IEEE 802.11 and IEEE 802.16 standards. The TCP/IP protocol stack is implemented in the OPNET simulation framework. The WiMAX model implemented in OPNET does not model physical layer losses of the air

interface explicitly, but instead includes a PDU dropping probability that can be used to model PHY layer losses. Since the focus of this research is on MAC layer performance, it was considered an appropriate simulation model to use.

The modelling environment of OPNET enables the design and implementation of networks that are similar to real network systems. There are three hierarchical modelling domains in OPNET. At the highest level is the network domain that describes the topology of the system. It consists of nodes and links. The internal architecture of these nodes and links are described within the node domain. The node models consist of modules such as sources, sinks and processors. The operation of the processors are defined in the in process domain, by means of finite state machines (FSM) and high-level language. Thus the simulator is an event-driven simulator.

The request/grant mechanism is modelled such that every bandwidth request is allocated a transmission interval individually throughout the simulation. The scheduling that is necessary for this kind of data transmission is illustrated in Figure 4.1 below:

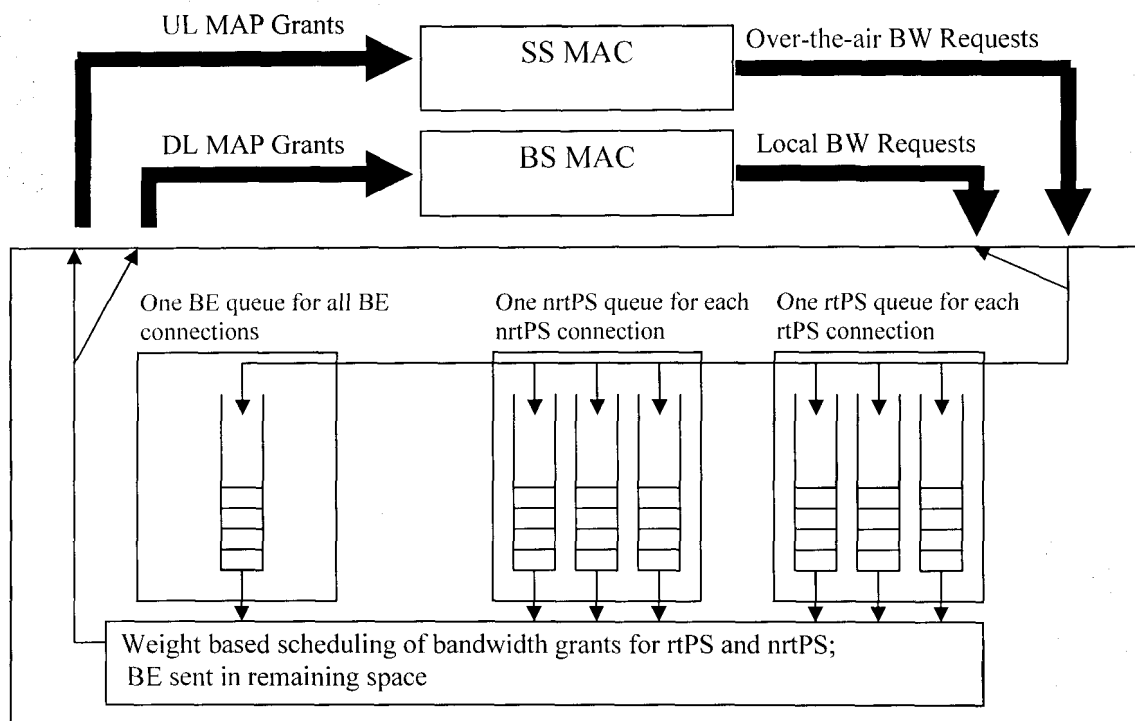


Figure 4.1 – Data Transmission Scheduling Mechanism

For the scheduling on the BS side, there is a separate bandwidth request queue for each rtPS and nrtPS connection. The bandwidth request queues for these scheduling services implement the

modified deficit round robin (MDRR) queuing discipline. It assigns a weight to each queue that is proportional to the minimum reserved rate of the corresponding connection. The BE bandwidth requests have a single queue for all connections. The grant system only processes bandwidth requests from this queue, when there are no bandwidth requests to serve among the rtPS and nrtPS connections. So one can see that the BE service uses the idea of a high latency queue. The UGS grants bypasses the scheduler in order to allocate bandwidth grants according to QoS requirements. The SS uses a GPSS grant scheduler. It duplicates the BS scheduler decisions.

4.4 Performance Metrics

The following metrics were identified as appropriate metrics to evaluate the performance of the network:

4.4.1 WiMAX Connection Delay

The uplink (UL) WiMAX connection delay is defined as the average time taken to send a packet from the WiMAX MAC of the SS, to the higher layer at the BS. SS here refers to the node in the network that has a dual WiFi/WiMAX capability. It serves as an access point on one side, and is considered as a WiMAX SS on the other side.

The downlink (DL) WiMAX connection delay represents the average time taken for a packet to travel from the WiMAX MAC at the BS to the higher layer at the SS. The definition of SS has the same meaning as above. This study looks at the WiMAX connection delays for each of the service flows used. This metric is measured in seconds and a smaller value is considered better.

4.4.2 End-to-End Delay

End-to-end delay is used as performance measure for voice and videoconferencing applications. It is defined as the average time taken to send an application packet to a destination node application layer. This metric is also measured in seconds, and a smaller value is considered better.

4.4.3 Download Response Time

The download response time is chosen as the metric to evaluate the email, FTP, and web-browsing application performance. For the case of the email application, it is defined as the average time that has elapsed between sending a request for emails and receiving emails from the server in the network. The signalling and connection setup delays are included in this statistic.

OPNET Modeler defines the download response time for the FTP application to be the average time that has elapsed between sending a request and receiving the response packet. It is measured from the time a request is sent to the server, to the time a response packet is received by the workstation in the WLAN/telecentre.

The page download response time denotes the average time required to download a page for the web-browsing application.

4.4.4 Packet Loss

Packet loss is described as the ratio of the number of packets lost to the total number of packets transmitted. It is expressed in percentage form.

4.5 Related Work

Previous performance evaluation work on IEEE 802.16 MAC includes work done by Ekland et al [21]. Here two specific scenarios were considered; a residential scenario where only a BE type of service was provided by the BS, and a Small Medium Enterprise (SME) scenario, where three types of services were investigated. A WiMAX simulator was used to investigate the above-mentioned scenarios. The study assumed an ideal channel in a fixed wireless PMP network. The load was increased by increasing the number of SSs. The average WiMAX connection delay results were presented. The results show the service differentiation in terms of delay between the data and multimedia traffic [21].

Work done by Cicconetti et al [22], is also a performance evaluation work on the IEEE 802.16 MAC. A PMP, FDD system was considered for the OFDM air-interface, although no packet corruption due to wireless channel impairment was considered. The scenarios investigated are for full-duplex SS's, and consider the BE, nrtPS and rtPS scheduling services. All scenarios assume that each connection carries a combined traffic load from a variable number of data sources. The number of stations carrying the aggregate traffic from the data sources is also variable. The results are presented in terms of average WiMAX connection delay and throughput. The system performance is found to depend on factors such as frame duration and bandwidth request mechanisms.

The scenarios investigated in this study use the work presented Ekland et al [21] and Cicconetti et al [22], as benchmarks. The research for the Nongoma region investigates the performance of the different services based on the scheduling methods used, as described in section 4.3 above. A

comparison is made between the system performance under ideal and non-ideal channel conditions. Ideal channel conditions imply no packet losses or corruption due to wireless channel impairments, such as fading. The system performance is also investigated when ARQ is enabled.

4.6 Simulation Scenarios

The figure below represents the network structure used for the simulation scenarios. The network comprises of one BS and four WLANs, thus four WiMAX SSs. They represent the four telecentres. Application requests are made to a server that is connected to the BS, via a digital signal level 3 (DS3) link. This link data rate is approximately 45Mbps. Generally DS3 links are used to support the high volume traffic requirements of Universities, large corporations or call centres.

The distances of the telecentres from the BS have been set to correspond to those mentioned in Chapter 2.

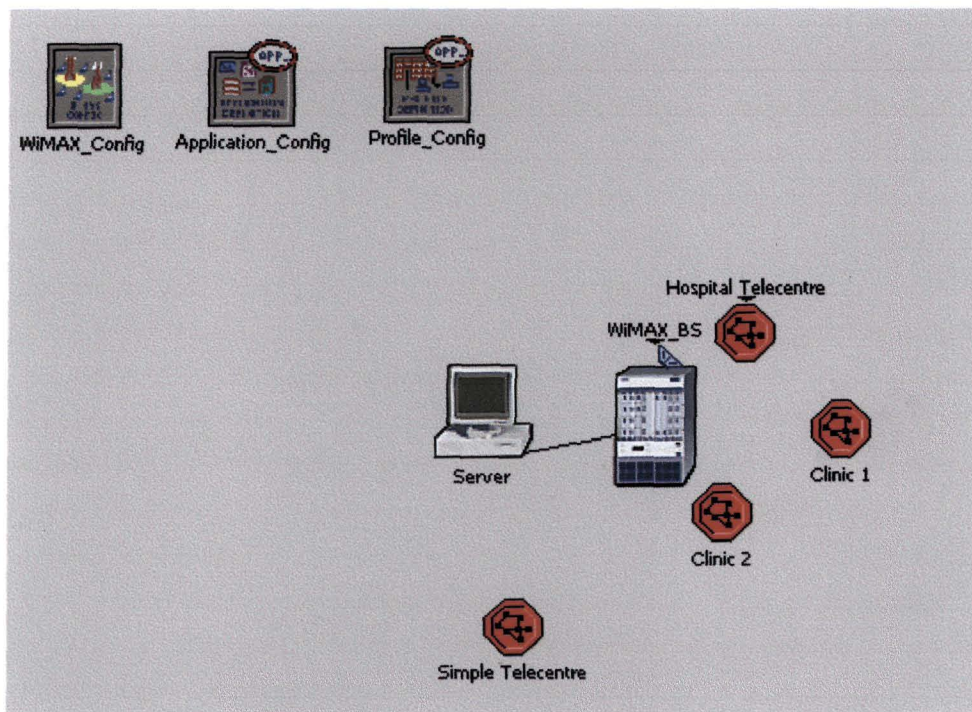


Figure 4.2 – OPNET Simulation Scenario for Nongoma Case Study

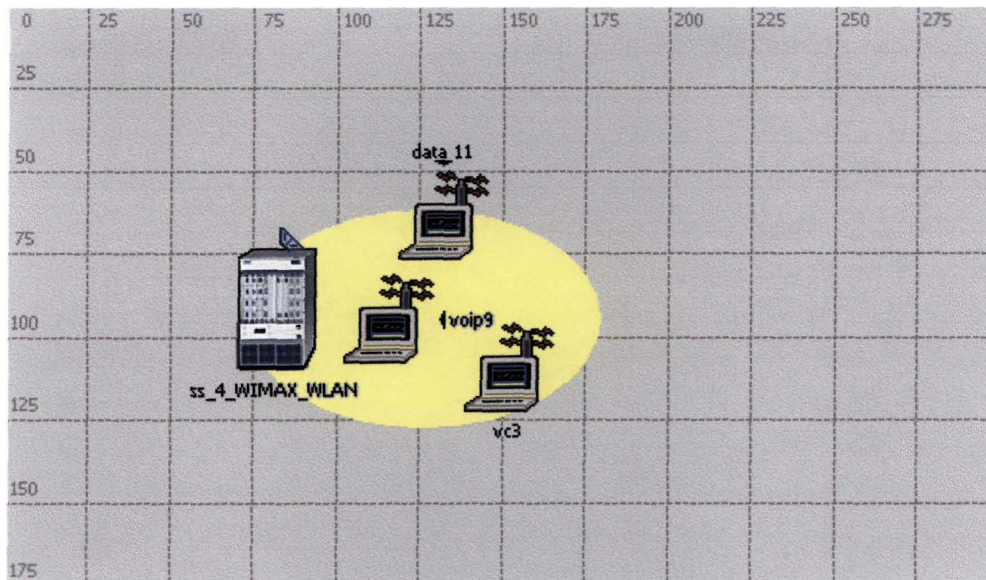


Figure 4.3 – OPNET Simulation Scenario within Telecentre

Figure 4.3 shows the simulation setup within one of the clinic telecentres. The other LANs have been setup similarly. The WLANs within each telecentre have been confined to 300m x 300m in the simulation setup.

The applications supported by the workstations in the wireless LANs generate traffic patterns that correspond to the different service types specified by the IEEE 802.16 standard. Each connection to the BS carries aggregate traffic from the data sources (workstations) in the wireless LANs (telecentres). The system parameters used in the simulation analysis is shown in the Table 4-1 below:

WiMAX Simulation Parameters	Value
Frequency Band	3.5GHz
Maximum Channel Data Rate	40Mbps
Frame Duration	10 μ s
Modulation	QPSK $\frac{3}{4}$ and 16 QAM $\frac{3}{4}$
Duplex Mode	FDD
Minimum Reserved Traffic Rate	
Best Effort	0bps
Non-Real Time Polling Service	83kbps
Unsolicited Grant Service	1Mbps
Real-Time Polling Service	7Mbps

WiFi Simulation Parameters	Value
Channel Data Rate	54Mbps
(Effective Channel Data Rate)	27Mbps
Protocol	802.11a
MAC Protocol Mode	DCF
Traffic Model	
<p>Email (UL and DL) File size variable according to Table 2-3 Interarrival time 240s</p> <p>FTP (DL) File size variable according to Table 2-3 Interarrival time 240s</p> <p>Web-browsing (UL and DL) Object sizes variable according to Table 2-3 VoIP as constant bit rate (CBR) traffic (UL and DL) G.729 codec Holding time 3 minutes Increasing number of calls for each year as per Table 2-6 implemented in the voice traffic profile of each VoIP phone</p> <p>Video conferencing (UL and DL) Varied frame size parameter to meet predicted traffic rate as per Table 2-5</p>	

Table 4-1: System Parameters

The hospital and clinic telecentres host five applications, namely: videoconferencing, FTP, email, web-browsing and VoIP. The simple telecentre hosts all of the applications mentioned, excluding the videoconferencing application. Each of the telecentres establishes an uplink and downlink connection for the applications supported.

4.7 Performance Analysis

In each of the simulation scenarios the load is increased by increasing the files sizes for the data applications, the call rates for the voice applications and the frame sizes for the videoconferencing applications, according to predictions in chapter 2. The interarrival times remain constant. All of the scenarios investigated in the simulations, consider the performance of the different service flows in terms of UL and DL WiMAX connection delays for the total network load. The total network load is the sum of the total load on the uplink and the total load on the downlink, where the total load on the UL is the sum of the load from all the WiMAX SSs/telecentres, and the total load on the DL is the load transmitted to all of the WiMAX SSs/telecentres.

4.7.1 Ideal Channel Conditions: Case 1

This scenario consists of the server and BS providing Internet, voice and videoconferencing access to the SSs. The performance is considered under ideal channel conditions. Two cases are considered, which differ in terms of the scheduling service used for the FTP application. Figures 4.4 and 4.5 below illustrate the average UL and DL WiMAX connection delays for each type of traffic, as the network load is increased. In this scenario the FTP, email and web-browsing applications use the BE scheduling service. VoIP and videoconferencing applications use the UGS and rtPS scheduling services, respectively.

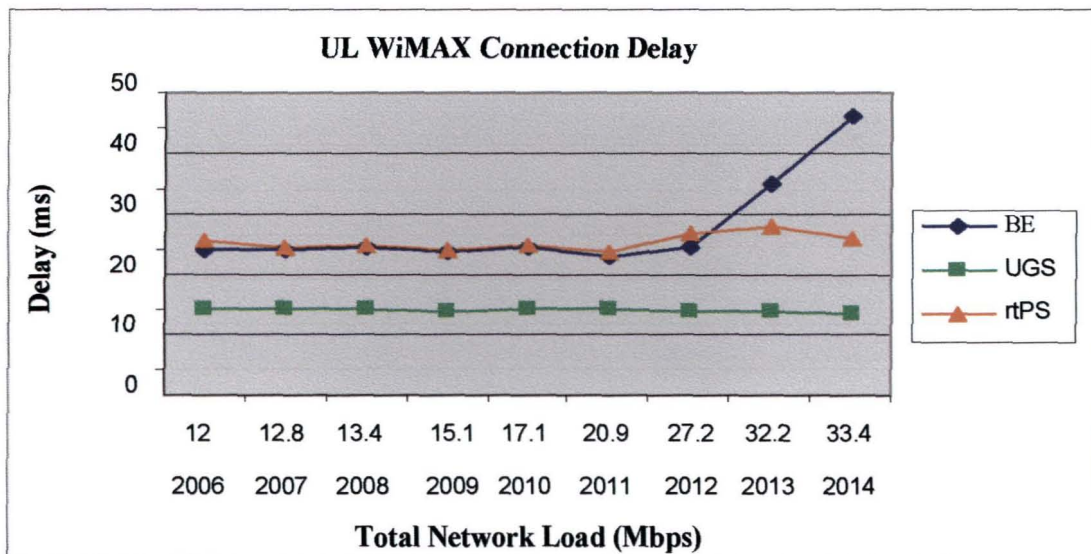


Figure 4.4 – Ideal Channel Case 1: Average UL Delay vs. Total Network Load for each year

The average delay of the UGS connections is almost constant as the network load increases for the uplink curves. This is because, the UGS is granted bandwidth every 10ms. Thus its delay is approximately bounded to this value, and doesn't depend on the increasing load.

Figure 4.4 above, also indicates that the network is experiencing congestion at around 2012, where the total network load obtained by simulation is 27.2Mbps. Relating this to Chapter 2 traffic predictions, the telemedicine bandwidth requirement is around 3.8Mbps in 2012 (taken from Table 2-6). Therefore the combined rtPS load for the Hospital and Clinic telecentres is 11.4Mbps on the UL and 11.4Mbps on the DL. Thus the network load for the rtPS is around 23Mbps. It is approximately 84% of the total network load. The network load is obtained similarly for the UGS service, and is approximately 2% of the total network load. The data is obtained from Table 2-8 in Chapter 2. The remaining 14% of the total network load comprises of data traffic. (Values obtained from Table 2-5 in Chapter 2).

The network utilization is around 68%, in 2012, when congestion sets in and a sharp increase in the average delay of the BE traffic is seen, despite the high amount of rtPS traffic on the network. (The utilization is obtained by the ratio of total network load at 2012 to the WiMAX maximum channel data rate, i.e. 40Mbps). BE traffic experiences sharp increases in delay, because applications using this scheduling service have to compete for bandwidth requests. Therefore, the average time needed to request bandwidth increases with increasing network load. The rtPS that is granted unicast polls to request bandwidth reserved to it, every 1.5ms, and therefore is able to maintain smaller increases in average delay.

The polling interval is calculated in OPNET as 8 times the estimated packet/second rate for the rtPS. The packet/second rate is the ratio of the reserved rate in bps to the average SDU size. The reserved rate is 7Mbps, and the average SDU size is fixed at 1280 bytes for rtPS. This works out such that the packets/second rate for rtPS is 5468packets/second, or 1 packet per 182.86 μ s. Thus, a unicast poll rate of 1.5ms is obtained.

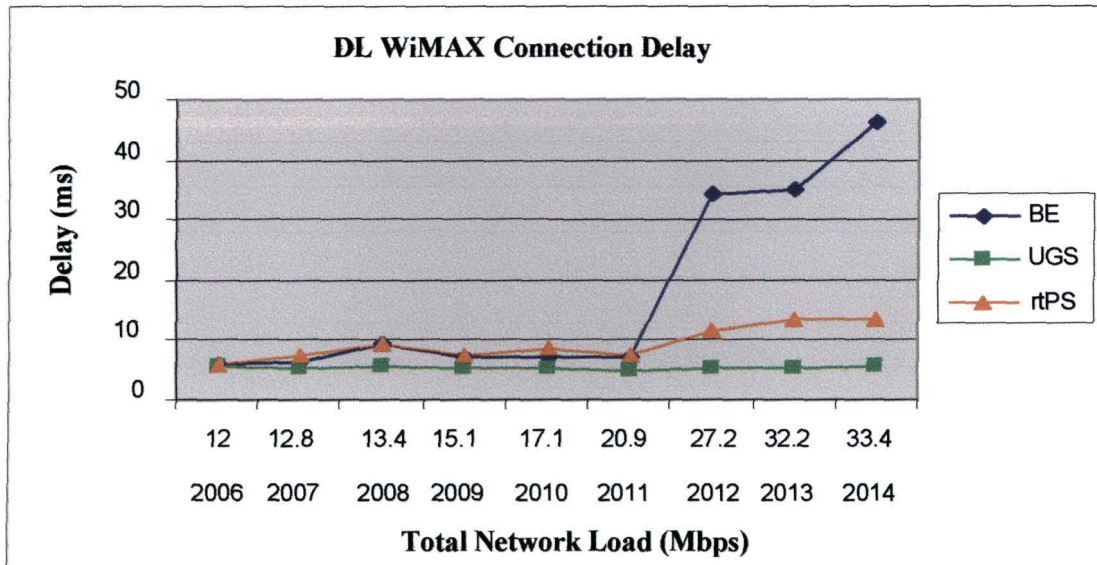


Figure 4.5 - Ideal Channel Case 1: Average DL Delay vs. Total Network Load for each year

Figure 4.5 shows that the average delay for all service types increases with increasing network load. When the system is not congested, one can barely see the service differentiation between the real and non-real-time traffic, for the downlink curves. This implies that the data queues at BS are not full, and that packets do not have to wait to be transmitted. There is a sharp increase in the average delay of the data traffic between 2011 and 2012 (this implies a congested network). Meanwhile, the delay of the multimedia traffic or videoconferencing traffic increases slightly, despite the high traffic rates. This is because the BE scheduling service doesn't guarantee any bandwidth for its applications, whereas, the rtPS on the other hand has bandwidth reserved for its applications.

Also the average delays of the UL connections are higher than the DL connections. This is because SSSs have to first request the bandwidth required, then wait for the bandwidth grant, before being able to transmit any data. Although the UGS connection doesn't request bandwidth explicitly, it still has to wait for the bandwidth grant, and therefore spends a little longer in the queue for the UL transmission, as opposed to the downlink transmission, where as soon as a packet is queued, it is eligible for transmission.

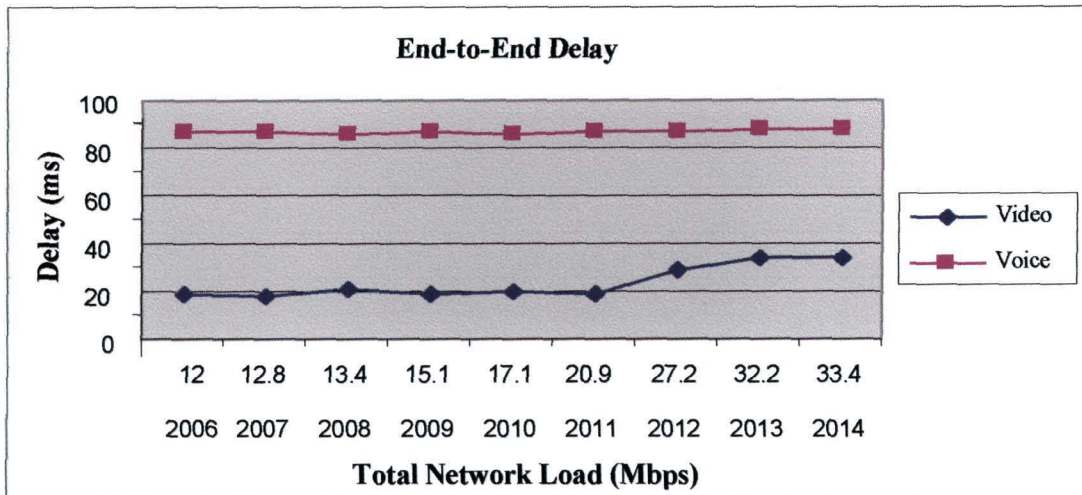


Figure 4.6 - Ideal Channel Case 1: Average End-to-End Delay vs. Total Network Load for each year

Figure 4.6 illustrates end to end delays of videoconferencing and VoIP applications. It can be seen that the videoconferencing end-to-end delay does begin to degrade at the onset of congestion, but it is still within the end-to-end delay bound of 100ms as suggested by the ITU for interactive video [23]. It should be noted that the network utilization is at approximately 84% in 2014. Voice end-to-end delay is maintained at a steady rate, also within its suggested ITU delay bound of 350ms [24]. The voice end-to-end delay is also within the delay bound used in this research, i.e. 150ms.

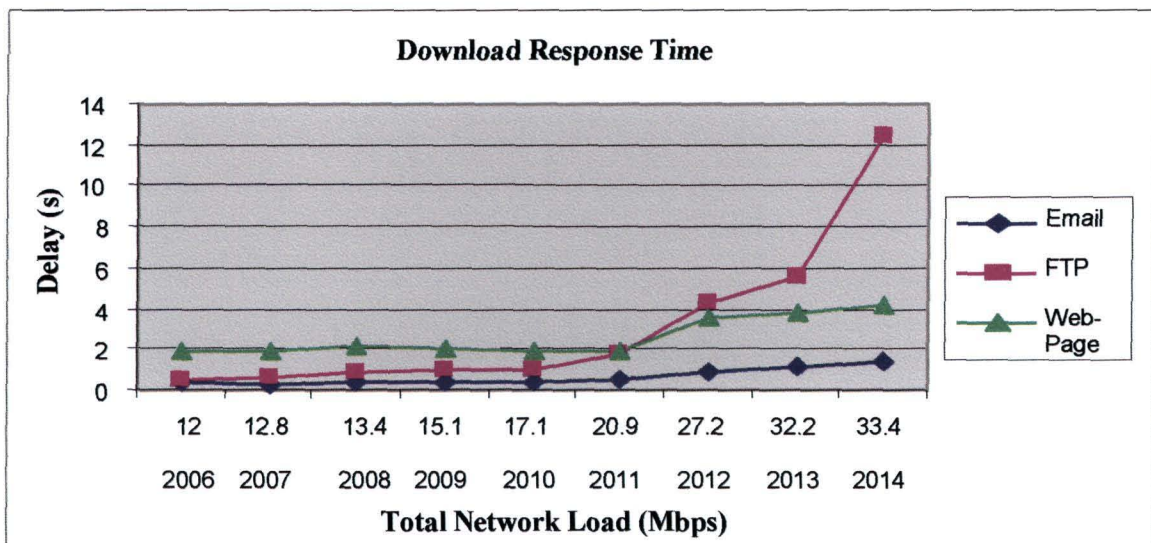


Figure 4.7 - Ideal Channel Case 1: Average Download Response Time vs. Total Network Load for each year

The download response times of the FTP, email and web-browsing is shown in Figure 4.7 above. The download response time times can be considered high when compared to WiMAX connection delays. This is because application packets are segmented at a lower layer, i.e. the transport control protocol and Internet protocol layers (TCP/IP). Thus the application request can consist of multiple packets, one or more of which can be transmitted through the network asynchronously. Also if packets are duplicated or lost in the network, which generally happens during congestion, it will cause TCP to retransmit those packets. Therefore, the download response times are much higher. One can also notice that the FTP download response time surpasses the webpage download response times from 2011. This follows from the data traffic predictions in Chapter 2, where the FTP file sizes are higher than the webpage sizes from 2011 to 2014.

The results show that it takes approximately 4 seconds, to download a 1.027MB webpage and approximately 2 seconds to access a 10kB email in 2014. (These file sizes are obtained from Table 2-3 in Chapter 2). An interesting point to note that was mentioned in Chapter 2 and used in the calculations is that, in general, the average tolerance of users to download times is approximately 10 seconds [13]. So these download response times are acceptable within this guideline until 2014, for webpage downloads and email traffic. FTP downloads take slightly longer time in 2014, where a 657kB document takes approximately 12 seconds to download, but this download response time can still be considered acceptable within the guideline.

4.7.2 Ideal Channel Conditions: Case 2

This scenario is also investigated under ideal channel conditions. In this case, though, the FTP application uses the nrtPS scheduling service. The nrtPS scheduling service is generally used for traffic types that have no stringent delay requirements, such as FTP. The email and web download traffic use the BE scheduling service, with VoIP in the unsolicited grant service (UGS) class. The results are presented below:

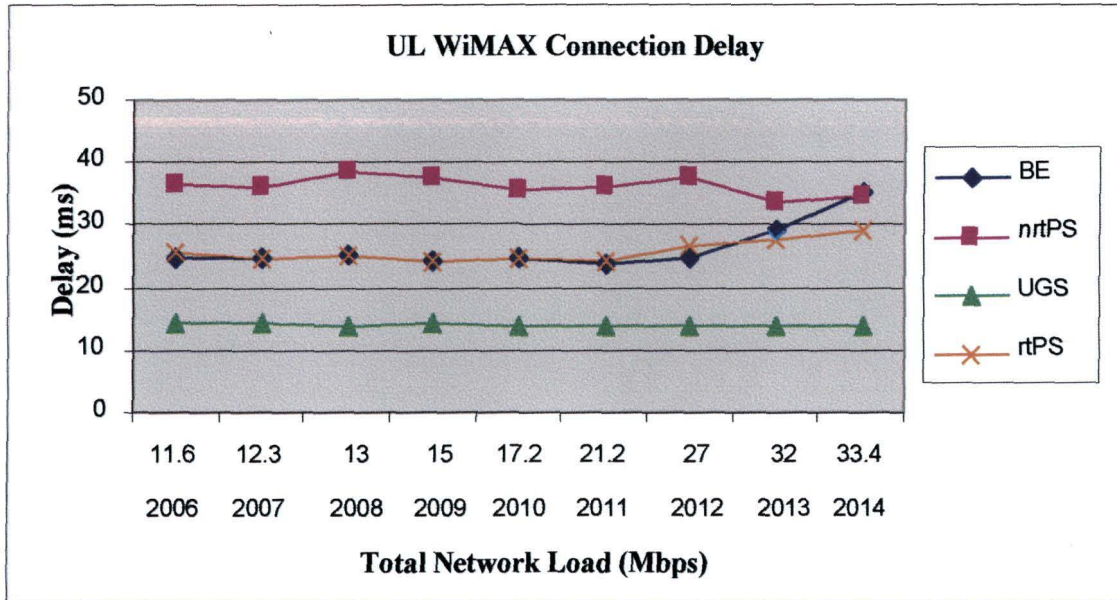


Figure 4.8 - Ideal Channel Case 2: Average UL Delay vs. Total Network Load for each year

Figure 4.8 above shows that in a lightly loaded system, (i.e. in the years preceding 2012), the nrtPS does not offer better service than the BE class, in terms of delay on the UL. This is not an intuitive result, and the conclusion that can be drawn from this result is that it takes longer for the nrtPS connections to request bandwidth than it does for the BE connections, in a lightly loaded system. It means that the time taken for unicast polls to reach the nrtPS connections is longer than the time taken to request bandwidth using contention.

This non-intuitive result corresponds to results obtained by Cicconetti et al [22], where a bandwidth request analysis was carried out. The study investigated the relative effectiveness of the BE and nrtPS bandwidth request mechanisms with data traffic. It was found that in a lightly loaded system, the nrtPS scheduling service did not improve the average delay performance of uplink connections.

Non-real time polling service (nrtPS) connections are provided with unicast polls approximately every 15ms for a fixed SDU size of 1280 bytes. This polling rate was obtained using the same method as for the rtPS in the previous simulation scenario. The unicast polls provided to the nrtPS connections take 10 times longer, than the unicast polls provided to the rtPS connections.

When the system starts to get congested (in year 2012 and beyond), BE delays are seen to rise faster than the nrtPs and the rtPS. It can be noted that the increase in delay, though, is not as sharp as in the previous case. This is because there is less traffic that is requesting bandwidth on a contention basis now, in the BE class.

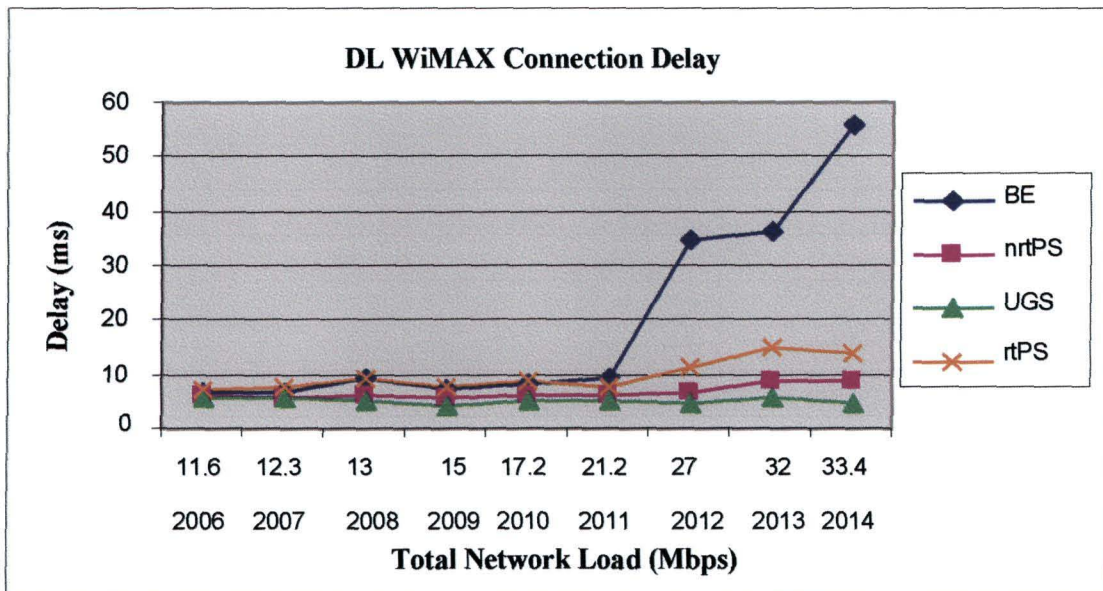


Figure 4.9 - Ideal Channel Case 2: Average DL Delay vs. Total Network Load for each year

For the case of the DL delays, it can be seen that the videoconferencing delay is almost always higher than the nrtPS. This is because, whether the system is lightly loaded or not, the amount of traffic using the rtPS is always higher. The amount of traffic using the rtPS is always more than 80% of the total network load. Thus, more bandwidth is reserved to this scheduling type. It can also be noted that the BE delay increases more sharply than the DL BE delay in Case 1. This is because, the BE traffic in the data queue at the BS, has to wait for the rtPS queues, and now also for the nrtPS traffic queues to be empty before its data packets can be transmitted on the downlink.

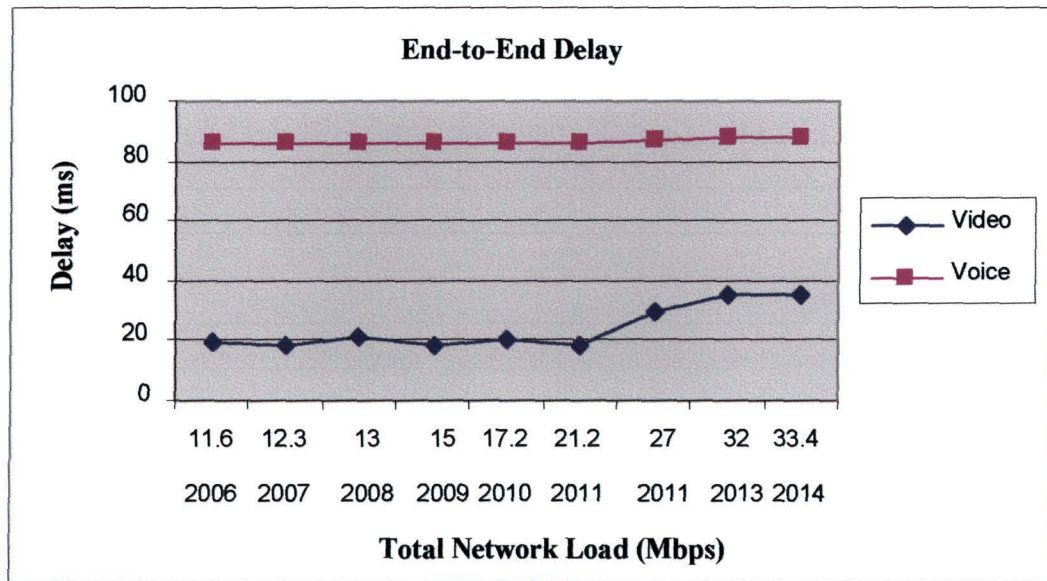


Figure 4.10 - Ideal Channel Case 2: Average End-to-End Delay vs. Total Network Load for each year

The end-to-end voice and video delays in this case are not affected by the FTP traffic using the nrtPS scheduling service. This is because the main delay component is from the transport layer; in this case, the user datagram protocol (UDP) is used as the transport layer protocol. The traffic offered at this layer is the same as before.

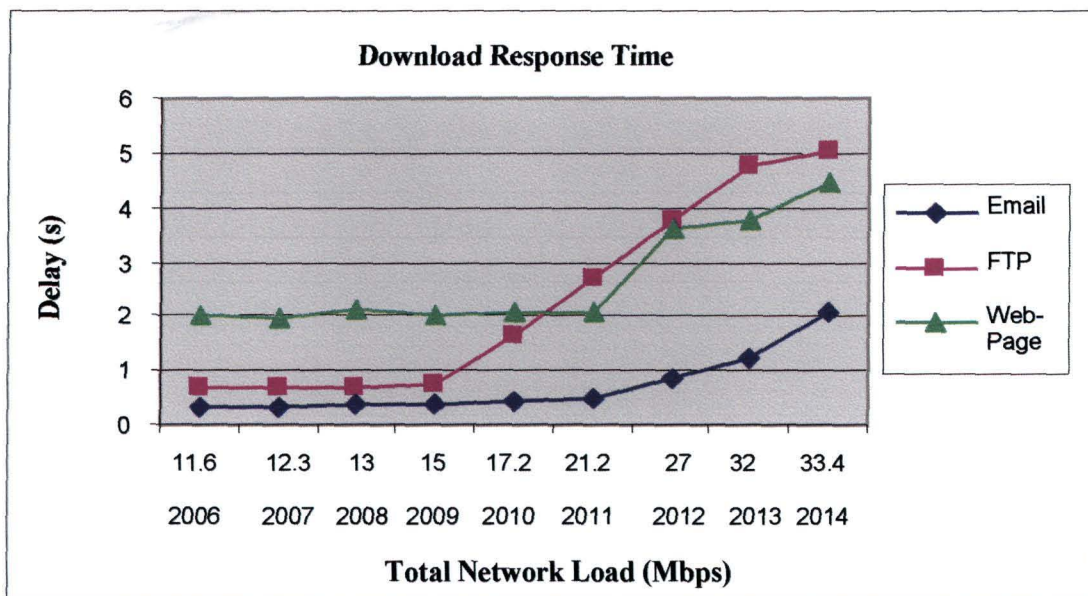


Figure 4.11 - Ideal Channel Case 2: Average Download Response Time vs. Total Network Load for each year

There is a significant improvement in the FTP download response time. Previously it increased to approximately 12 seconds in 2014, now the highest delay is around 5 seconds. In the Case 1 of the ideal channel where FTP was using the BE scheduling type, TCP began retransmitting packets when the network began to get congested, thus increasing the delay. Because the nrtPS scheduling service has some bandwidth guaranteed to it, it suffers less during congestion, and thus fewer packets are retransmitted using TCP.

A conclusion can thus be drawn, that using the nrtPS service flow for FTP traffic, does improve the overall system performance, despite the fact that the performance of the uplink connections did not improve in a lightly loaded system. The real benefit is seen in the download response time for the FTP traffic, where the delay is less than half that, as in the case where the FTP traffic used the BE service flow. Since the goal of the network performance is obtaining good end-to-end QoS, a recommendation can be made, that in a practical system implementation, it is better to use the nrtPS service flow for delay tolerant traffic such as FTP. Therefore, the rest of the investigations that follow use four service flows for the different traffic types. The service flows are UGS, rtPS, BE and nrtPS.

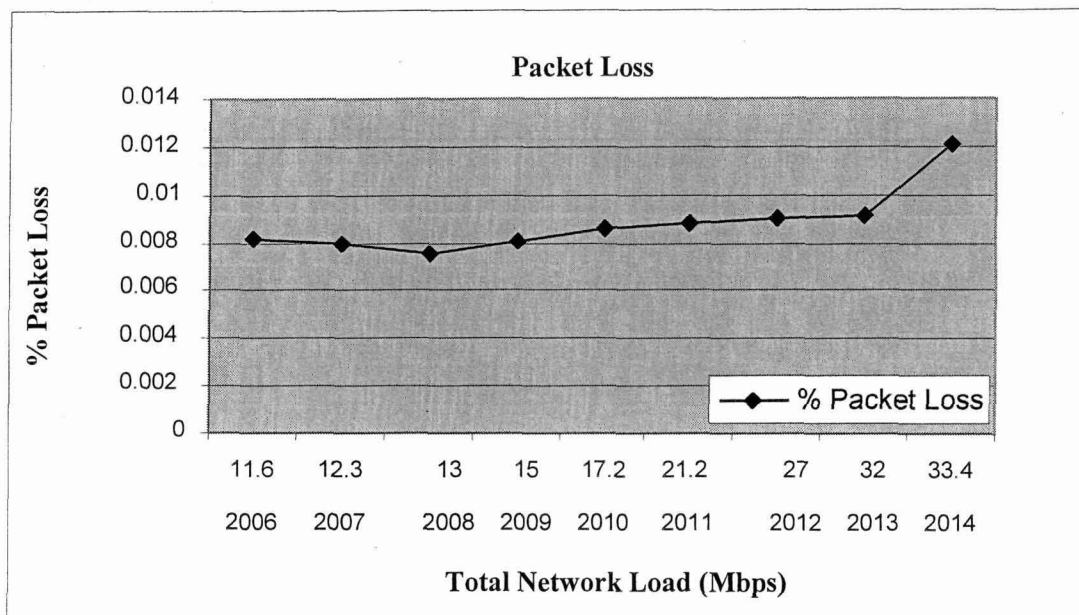


Figure 4.12 – Ideal Channel Case 2: Percentage Packet Loss vs. Total Network Load for each year

The packet loss details for this case is presented, with a view to compare it with the non-ideal channel scenario in section 4.7.3, and the scenario using ARQ in section 4.7.4. The results show

that the use of an ideal channel affords the system a packet loss ratio of less than 1%, even in 2014 when the network utilization is 84%.

4.7.3 Non-Ideal Channel Conditions

A non-ideal channel is considered, because an essential characteristic of wireless channels is that the signal strength varies randomly over time. This is caused by physical layer losses such as channel fading [41]. Methods such as ARQ are considered to mitigate these effects.

This research uses a PDU dropping probability to emulate the PHY losses of the wireless channel. A PDU dropping probability of 5% was chosen. Work done by Niyato D and Hossain E, [26] provide a mapping of the PDU dropping probability to signal-to-noise ratio (SNR). Thus, a PDU dropping probability of 5% corresponds to a SNR of approximately 25dB, in a wireless channel that is undergoing fading. Generally, the guideline for deployment of outdoor access points is that transmissions lower than 10dB is discarded [27]. This is equivalent to a PDU dropping probability of approximately 8% according to [26].

Experimental studies [29], [30] of wireless links indicate that typical SNR for good link quality is considered to be above 25dB. QPSK $\frac{3}{4}$ requires SNR of 11.2dB and has a fade margin of approximately 13.8dB. This is in line with industry experts that suggest that a fade margin in excess of 10dB be used [42]. The fade performance for 16 QAM $\frac{3}{4}$ is slightly worse.

According to [42], if the link distance increases, then the fade margin will also have to increase to provide reliable link quality. QPSK $\frac{3}{4}$ is used for the Simple telecentre and Clinic 1 which are placed a further two to three times the distance from the telecentres using 16 QAM $\frac{3}{4}$. Studies show that when the distance doubles, the power required to sustain the signal level is four times more [43].

Thus, emulating a wireless channel with a 25dB SNR, which corresponds to a 5% packet loss, is a fair choice for non-ideal channel conditions and was the basis of the simulations.

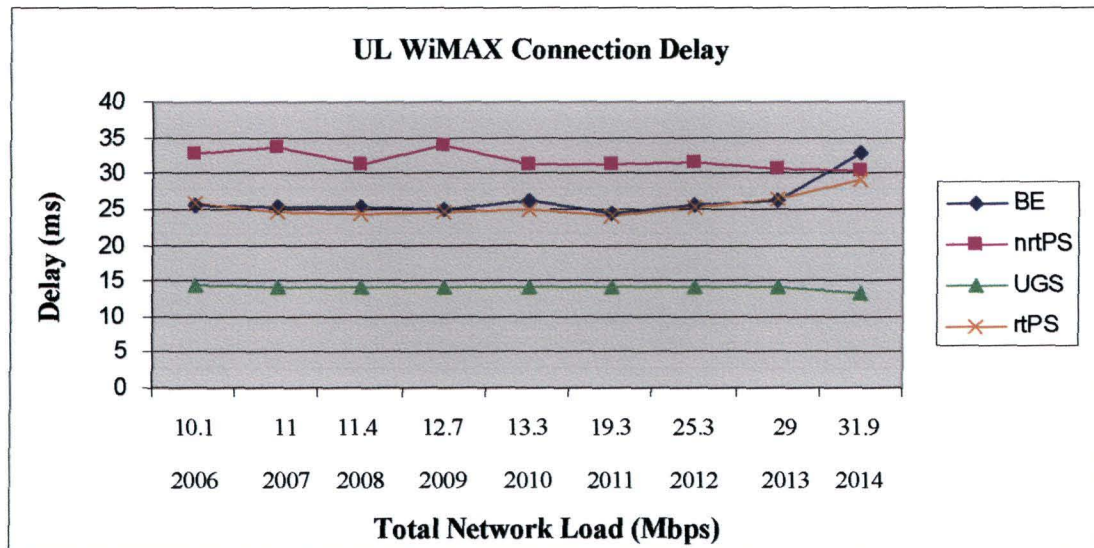


Figure 4.13 – Non-Ideal Channel: Comparison between average UL WiMAX Connection Delays of each service flow for each year

Figure 4.13 above illustrates the UL delay curves for the different service types, for the case of the non-ideal channel. A comparison between Figure 4.13 and the UL curves for the ideal channel case, i.e. Figure 4.8, shows that the average of the total network load is approximately 11% lower. So, the 5 % probability of a PDU being dropped on UL and DL caused the average total number of packets transmitted over all the years to decrease by approximately 11%. Therefore the total network load is not the same as for the ideal channel Case 2. The delay values are slightly smaller. This can be attributed to some of the traffic being lost.

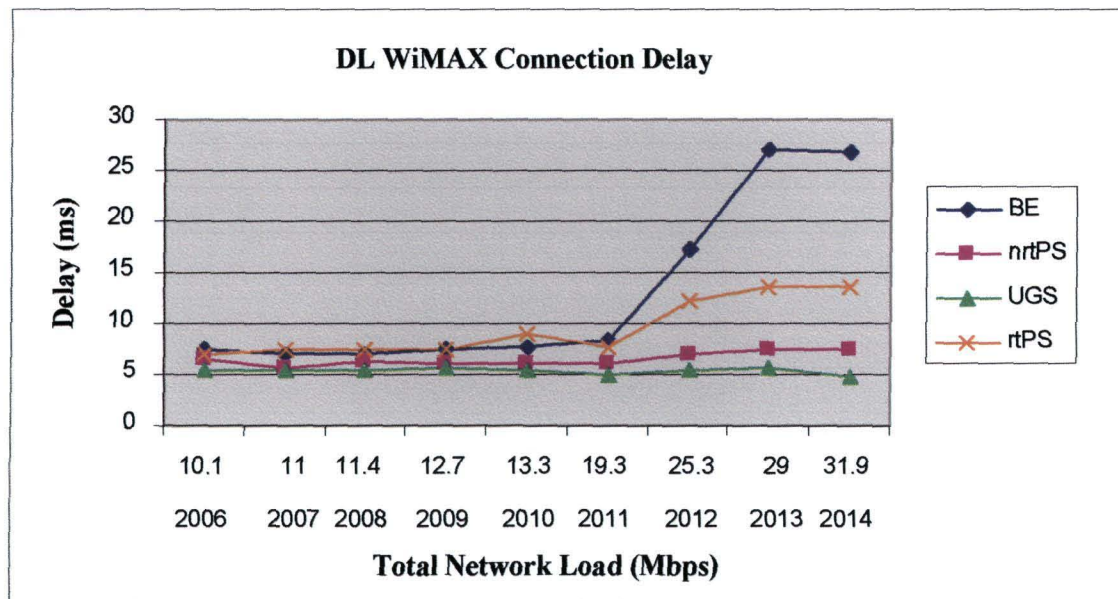


Figure 4.14 - Non-Ideal Channel: Comparison between average DL WiMAX Connection Delays of each service flow for each year

The same argument can be applied for the DL curves. The queues at the BS and SS are smaller because some of the traffic is lost, therefore the slightly smaller delays.

The conclusion that can be drawn from Figure 4.13 and Figure 4.14 is that introducing fair amount of packet loss into the wireless channel does not have a significant impact on delays at the MAC layer for the different traffic types.

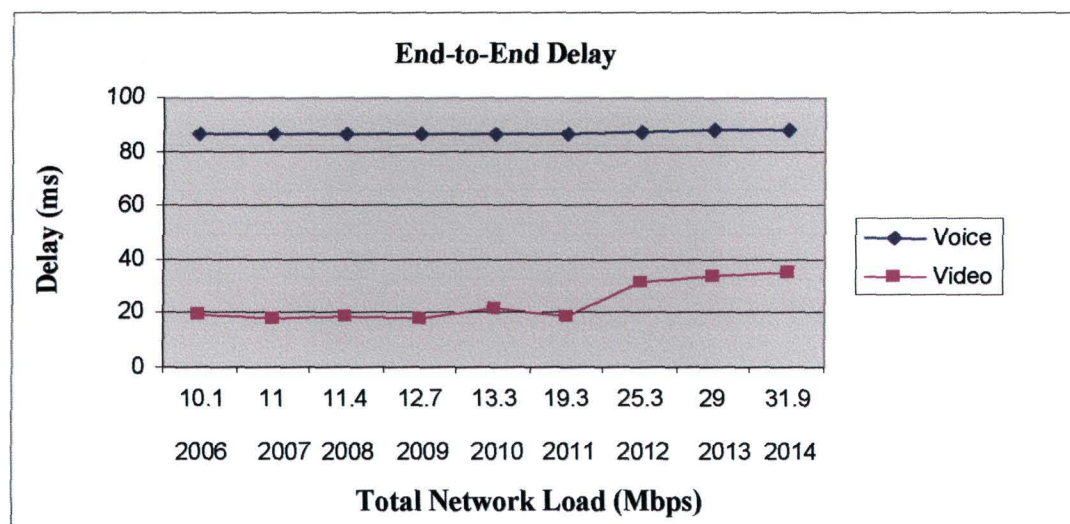


Figure 4.15 - Non-Ideal Channel: Average End-to-End Delay of real-time traffic on network vs. Total Network Load for each year

The voice and video end-to-end delay results are very similar to Figure 4.9 in the ideal channel Case 2, based on the 5% PDU dropping probability on UL and DL.

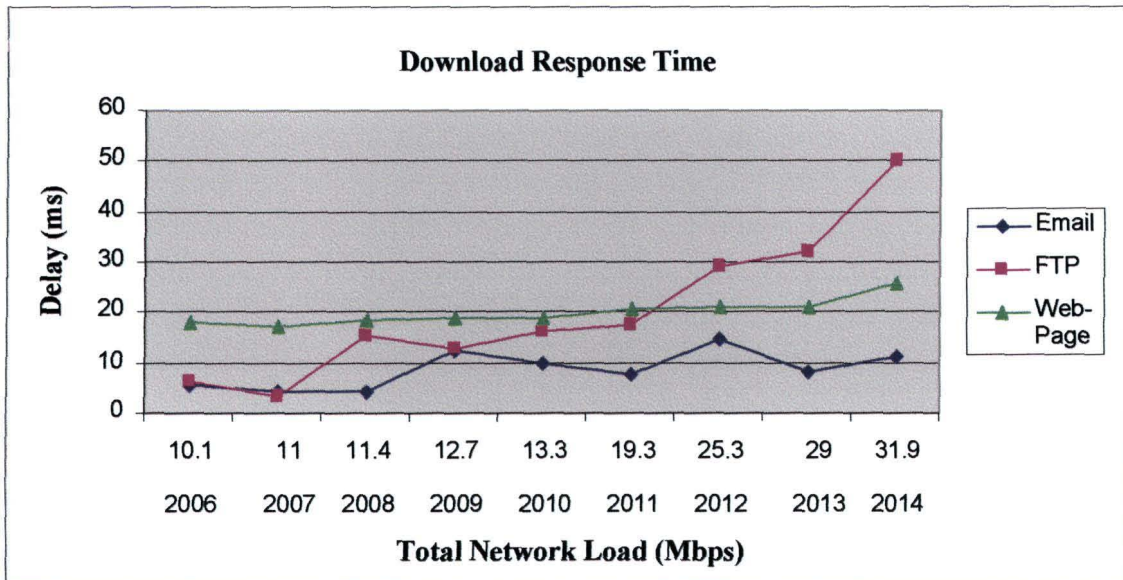


Figure 4.16 - Non-Ideal Channel: Download Response Time of data traffic on network vs. Total Network Load

Figure 4.16, shows a significant increase, in the delay for the download response times as compared to Figure 4.11 of the ideal channel Case 2. It is approximately 10 times more, in 2014. Figure 4.11 also shows a steady increase in the delay, as compared to Figure 4.16, where there are some delay variations. This is because data traffic arrival rate is unpredictable, this is even more so when there are packet losses due to channel impairments, and the variations in the delay can be attributed to the instability of the wireless channel.

This performance degradation can be explained by the fact that the email, FTP and Webpage traffic use TCP as the transport layer protocol. TCP which was originally designed for reliable wired mediums assume that any packet loss experienced is the result of congestion on the link. TCP uses the Go-Back-N protocol, where packets need to be acknowledged when they are received. When an acknowledgement is not received before a timeout interval, it is retransmitted. The time-out interval is calculated based on the end-to-end delay. So when packets are lost due to wireless channel impairments, such as the fading that was emulated in this scenario, acknowledgements for the packets are not received, and as a result there are frequent retransmissions. Figure 4.16 below shows and compares the number of TCP retransmissions over simulation time for the ideal channel Case 2 and the non-ideal channel in 2006.

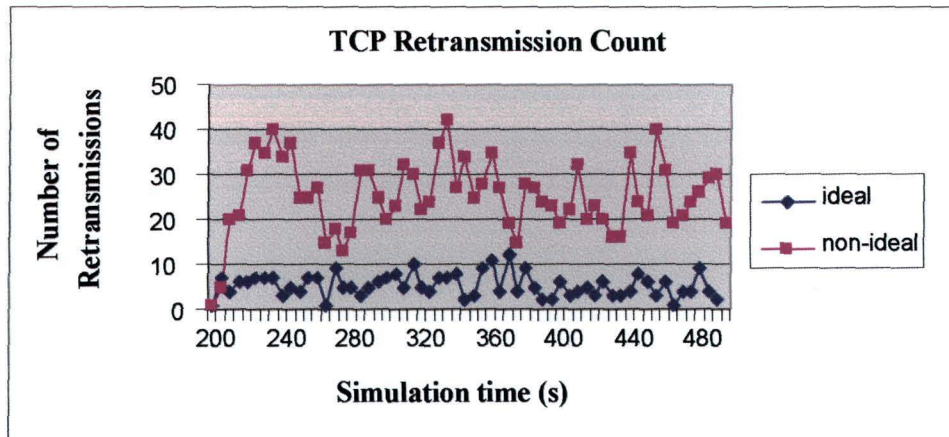


Figure 4.17 - Number of TCP retransmissions over simulation time for ideal channel Case 2 and non-ideal channel in 2006

The average number of TCP retransmissions for the non-ideal case is approximately 25. This is five times higher than the number of TCP retransmissions for the ideal channel Case 2. So, one can see the frequent number of retransmissions for the non-ideal channel case. When this happens, TCP reacts by using its congestion control mechanism to improve the situation. The congestion control mechanism works by reducing the transmission (congestion) window size, and thus the transmission data rate. As the data rate is reduced, the increase in delay is considerable. This increase in delay is displayed in the above curves of Figure 4.16.

It is not unusual to see this much degradation. Previous studies [31], on TCP performance in wireless mediums have shown that FTP download response time for 100kB files is approximately 45 seconds at a packet error rate of 5%.

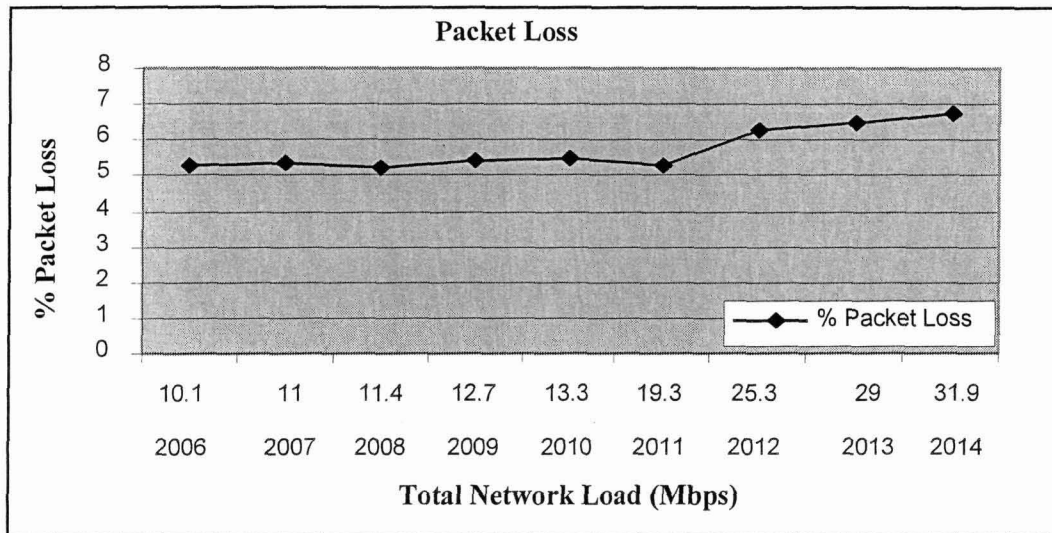


Figure 4.18 – Non-Ideal Channel: Percentage Packet Loss vs. Total Network Load for each year

The percentage of packets lost in the system remains fairly constant at 5% under low load conditions, and increases at the onset of congestion. It is much higher than for Case 2 of the ideal channel.

4.7.4 System Performance with ARQ

This scenario investigates the system performance in a non-ideal channel, same as before, but with the ARQ enabled on the traffic using TCP as transport layer protocol. These correspond to the traffic using the BE and nrtPS scheduling services. The ARQ block size has been set to 256 bytes, and the window size is four times larger for the FTP traffic which has a fixed SDU size of 1280bytes. The maximum ARQ window size as specified by IEEE 802.16 is 1024bytes. Thus the FTP traffic is fragmented. Since the BE traffic has smaller packet sizes (fixed SDU size of 320bytes), its block and window size was set to 64 bytes and 960 bytes respectively. The selective repeat ARQ mechanism is implemented in OPNET Modeler. The results are presented below.

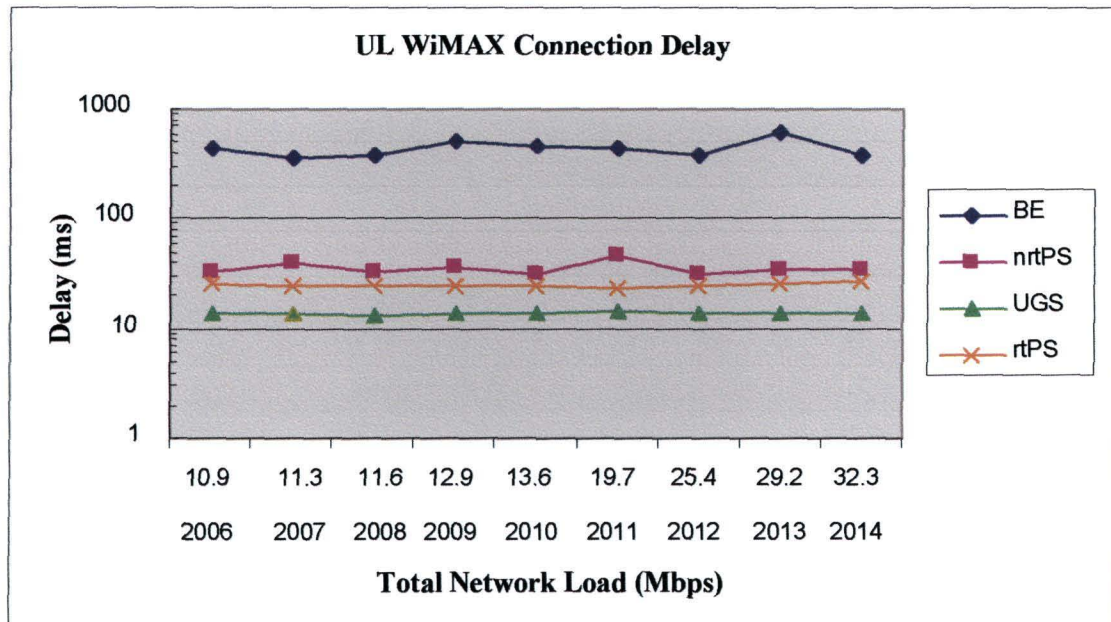


Figure 4.19 – ARQ: Comparison between average UL Delays of different service flows for Total Network Load of each year

Figure 4.19, shows significant impact on the BE traffic. These delays are much higher, by approximately a factor of 16, than the previous UL WiMAX connection delays. This is because the ARQ mechanism reacts to lost packets by retransmitting them. As a result these packets spend a large amount of time in the retransmission window until an acknowledgement has been received. This large delay is seen in the UL curves of Figure 4.19. The BE ARQ retransmission window can fit a maximum of three 320bytes SDUs.

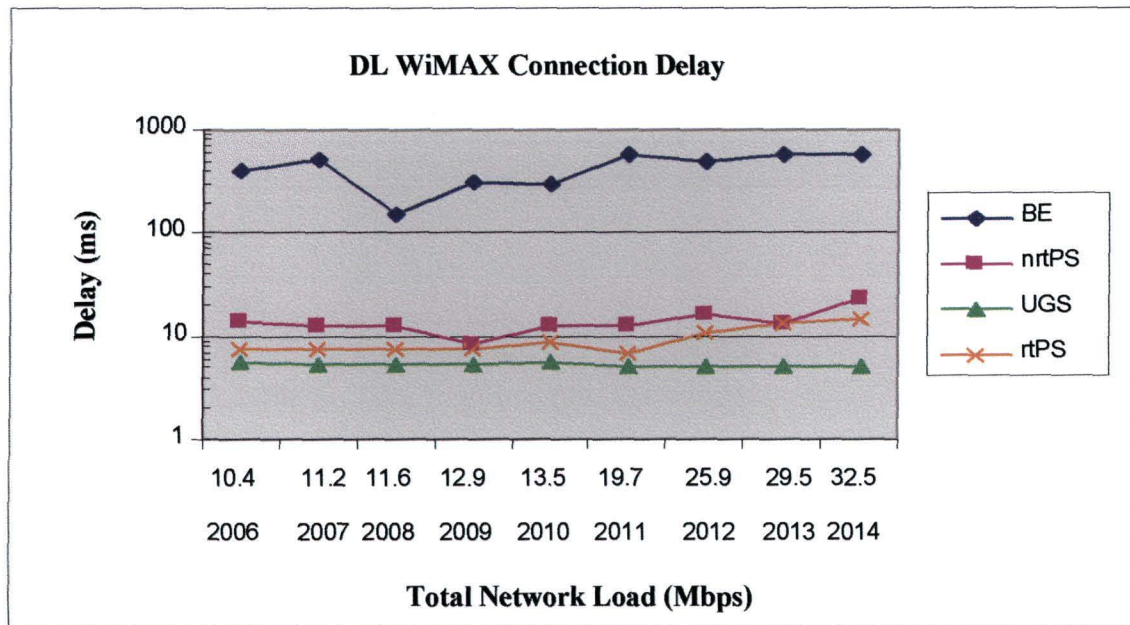


Figure 4.20 - ARQ: Comparison between average DL Delays of different service flows for Total Network Load of each year

The downlink curves display a similar behaviour, and thus the same argument applies, since ARQ is implemented on the UL and DL. The multimedia delays are unaffected. This is because ARQ is not enabled on this type of traffic.

The conclusion that is drawn from Figure 4.19 and Figure 4.20 is that the implementation of ARQ on the MAC layer increases the queuing delay, especially for the BE traffic. The delay tolerant BE traffic has no bandwidth reserved to it, and thus spends the most time in the queues. Figure 4.19 and 4.20 also show that the total network load has improved by about 1.5% as compared to the total network load in the non-ideal channel case.

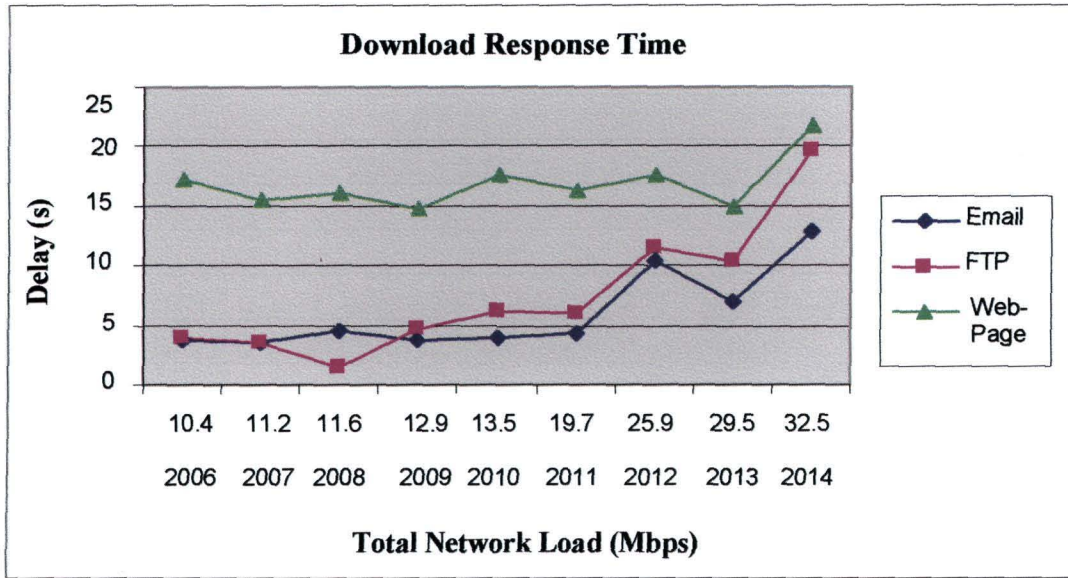


Figure 4.21 - ARQ: Download Response Time of data traffic vs. Total Network Load for each year

ARQ is an error correction method that makes the wireless link appear more reliable to TCP. This means that TCP retransmissions are less frequent, as seen in Figure 4.22 below.

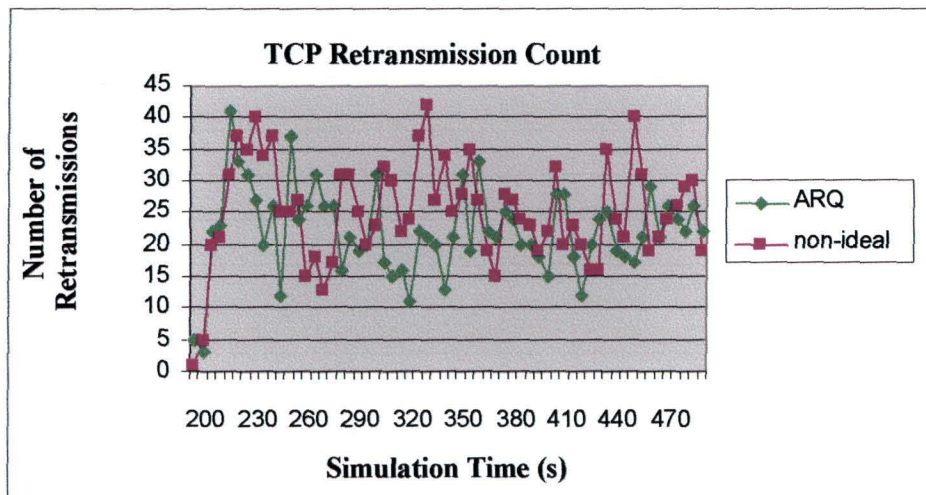


Figure 4.22 - Number of TCP retransmissions over simulation time for non-ideal channel and non-ideal channel with ARQ enabled in 2006

Figure 4.22 shows that the TCP retransmissions over time in 2006, have reduced by approximately 14% from the non-ideal channel case. Thus, TCP doesn't initiate its congestion control mechanism unnecessarily. This results in considerably lower download response times, since a large part of the download response time comprises of TCP delay, as seen in Figure 4.21 above. So even though the

packets spend a longer time in the queues at the MAC layer, the overall delay performance is improved. It can be considered acceptable within the 10 second user tolerance time for a download.

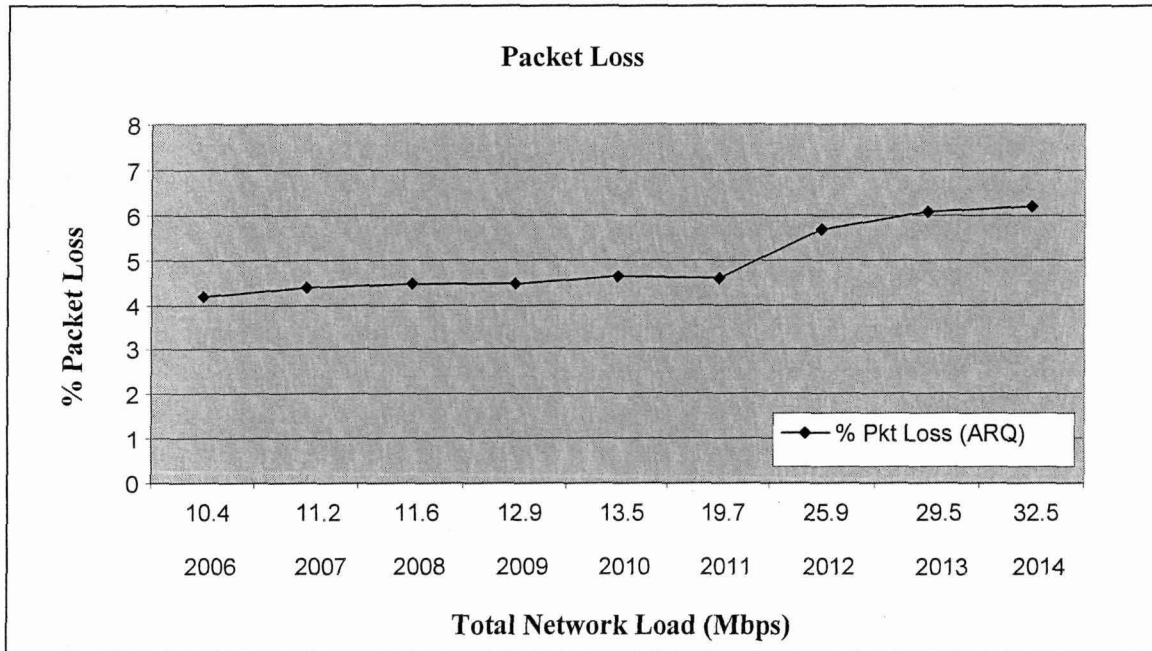


Figure 4.23 – ARQ: Total percentage Packet Loss vs. Total Network Load for each year

Figure 4.23 shows that by using ARQ, the packet loss ratio is decreased by approximately 1.5% as compared to Figure 4.18 of the non-ideal channel case. The total network load has improved by about 2%. (It accounts for packet loss on the UL and DL). This packet loss is still around 4 % higher than the ideal channel Case 2. This can be explained by the fact that there is a large amount of videoconferencing traffic on the network, which is also prone to packet loss. ARQ has not been enabled for this type of traffic, because it is a real-time application. Improving the packet loss ratio statistics means that some other type of error correction method is needed for the higher priority traffic. There have been investigations into using hybrid FEC-ARQ schemes for this type of traffic in next generation high-throughput wireless networks [32]. There is a fair amount of complexity involved in its implementation, and it has thus not been considered in this research.

4.8 Adaptive Modulation

Adaptive modulation is included in the IEEE 802.16 standard. The WiMAX BS and SS are able to negotiate the most appropriate modulation scheme, based on channel conditions and in particular SNR. Figure 4.24 below shows the theoretical maximum data rate achievable. It also shows the

maximum theoretical coverage achievable under LOS conditions for QPSK, 16 QAM, and 64 QAM, all with a coding rate of $\frac{3}{4}$. Since the OPNET simulation model, does not simulate the physical layer thus, only the theoretical data rate and coverage results are presented. The calculations for the maximum theoretical data rate and coverage use equations derived from a study done on coverage prediction and performance evaluation of IEEE 802.16 WMANs in [33], and are documented in Appendix B. The calculations assume that the sum of the transmit power and transmit antenna gain is 35dBm. This gives the total system effective isotropic radiated power. The receiver antenna gain is assumed to be 15dBi. This is accordance with ETSI regulations for the 3.5GHz band [34].

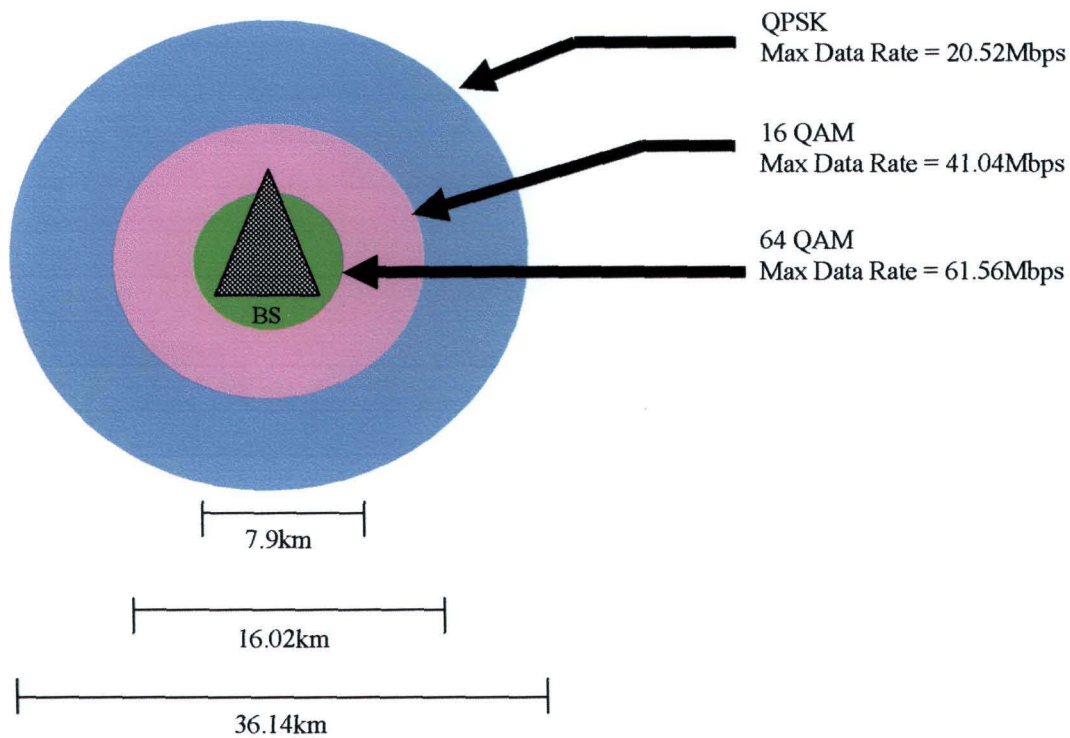


Figure 4.24 - Adaptive Modulation showing maximum theoretical data rate and coverage diameter

The 16 QAM modulation scheme was used in the simulation scenarios for the WiMAX SSs/telecentres closest to the BS and the maximum channel data rate achieved by simulation corresponds with the theoretical maximum data rate for 16 QAM $\frac{3}{4}$, for corresponding transmit power and gain.

The use of adaptive modulation is an important strategy for the Nongoma scenario. Using adaptive modulation allows the system to achieve a high data rate for the Hospital and Clinic SS 2 closest to

the BS, but it also allows the system to optimize its coverage and be able to include the Simple telecentre and Clinic 1 in the WiMAX footprint, since these telecentres are approximately 15km and 10km away respectively and thus out of the coverage diameter using 16 QAM modulation. In this research the study of the actual quantitative benefits of adaptive modulation is left as additional work.

4.9 Summary

The results for the simulation experiments have been presented. The investigations found that for each case the voice and video end-to-end delays were within their suggested ITU limits. In the simulations VoIP phones were set up at both the calling and receiving stations. In the event that the receiving phone lies in another network the additional delay needs to be considered so that the total end-to-end delay does not exceed 150 ms. VoIP is in particular, a key application for this case study, and thus ensuring that the stringent delay requirements were met was imperative. The TIA voice quality recommendation [25] includes a graph that plots R-value against delay. This graph is shown below:

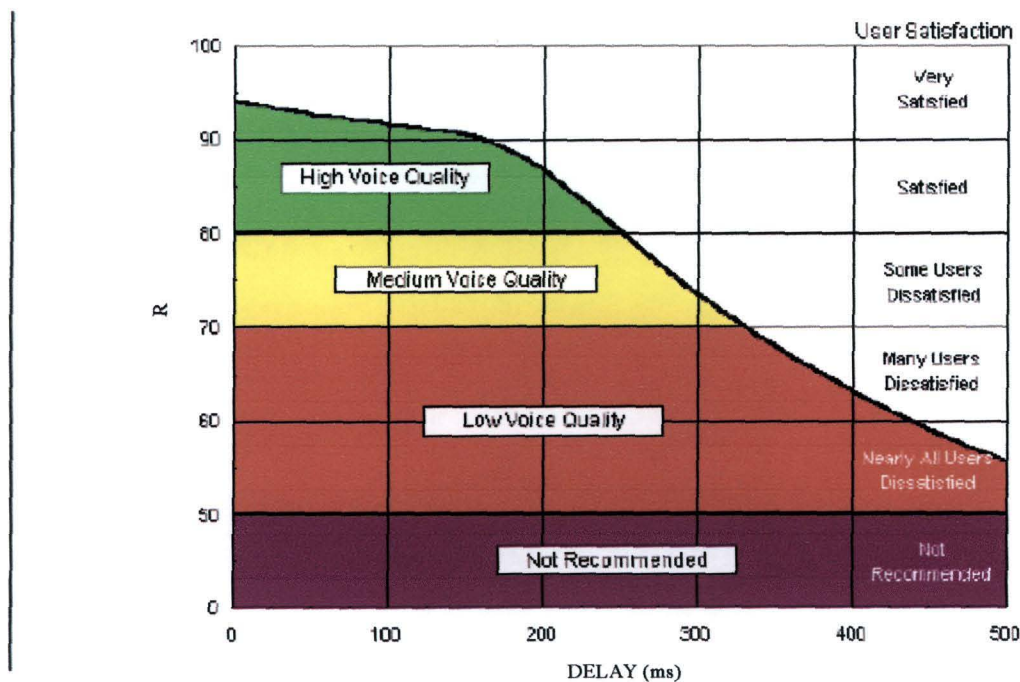


Figure 4.25 - E-model VoIP Classification [25]

The voice end-to-end delays were between 80 and 90ms, corresponding to an R-value above 90, and a MOS of 4.3, leaving users with high quality voice service.

The UL WiMAX Connection delays for the ideal channel Case 1 showed that the MAC delays are dependent on the BW request mechanisms used. This in turn relates to the service flow used. This conclusion corresponds with the conclusion drawn by Cicconetti et al in [22]. The DL WiMAX connection delays were found to increase much more sharply than the multimedia traffic in an overloaded system. This conclusion is in line with the conclusion drawn by Ekland et al in their investigations in [21].

The trend for the UL and DL WiMAX connection delays in ideal channel Case 2, are the same as the ideal channel Case 1. In, addition it was found that the download response times for FTP, web-browsing and email traffic improved by 50%, when the nrtPS was used for FTP traffic.

The non-ideal scenario found that introducing a 5% PDU dropping probability on the UL and DL for all the traffic types caused the total network load to decrease by approximately 11%. The most significant impact was seen on the download response times of the data traffic, which increased 10 times over.

The ARQ investigation caused the MAC queuing delay of the BE traffic to increase by a factor of 16 under low load conditions, and the delay remained fairly constant even when the system began to get congested. The overall delay performance of the data traffic, i.e. the download response times decreased by 50% as compared to the non-ideal channel case, but the packet loss was still approximately 4% higher than the ideal Case 2. This is attributed to the large amount of multimedia traffic on the network that is subject to packet loss, but don't have ARQ enabled.

Section 4.8 confirmed the maximum channel data rate used in the simulation scenarios. It also presented the trade-off between coverage and bandwidth. The Nongoma scenario is a typical case where one would like to maximize coverage at the expense of bandwidth. The Clinic 1 and Simple telecentre combined bandwidth requirements in 2014 are approximately 10Mbps (obtained from Table 2-9, Chapter 2), so the maximum data rate for QPSK of 20Mbps is able to support these requirements.

5 MESH MODE ANALYSIS

5.1 Introduction

Mesh mode is an optional topology that is supported in the IEEE 802.16 standard. This topology allows each node to directly communicate with neighbouring nodes within their coverage radius, as opposed to PMP mode where communication only occurs through the base station. Applied to the Nongoma scenario, it means that each telecentre will be able to communicate with any other telecentre within its coverage limit. This kind of communication between the telecentres can provide a cost effective, robust WiMAX backhaul network. It also means that the coverage of the network can be extended, so that more people can benefit from the backhaul network.

This chapter presents a brief technical overview of mesh mode in the IEEE 802.16 standard. It also includes a high level cost analysis of the benefits of a WiFi/WiMAX mesh network in the entire Nongoma region.

In addition, this chapter investigates the case of extending the reach of the WiMAX backhaul network in the Nongoma region. Results are obtained in terms of reach extension and maximum efficiency. Maximum efficiency is defined as the ratio between the net throughput on the MAC layer and the maximum throughput on the physical layer. The two scheduling methods of mesh mode namely, centralized scheduling (CS) and coordinated distributed scheduling (DS) are analyzed and compared.

5.2 Technical Overview: Mesh Mode

5.2.1 Centralized Scheduling

IEEE 802.16 mesh mode defines three scheduling mechanisms for data transmission, namely centralized scheduling, coordinated distributed scheduling and uncoordinated distributed scheduling. In centralized scheduling mode, traffic is scheduled using a request-grant-confirm handshake. The BS is termed the mesh BS (MBS). The MBS determines how the resources are shared among the mesh SSs. As a result, the scheduling process is simplified. This simplified scheduling procedure, however does come at the cost of long connection setup delays, and in general, centralized scheduling is not suitable for low traffic needs.

5.2.2 Distributed Scheduling

In the distributed scheduling method, each node in the mesh network schedules its traffic by using a request-grant-confirm, three-way handshaking scheme. Each node transmits its resource availability information as well as any requests it has to its one-hop neighbours. One of the destination nodes in the network will grant the request if it can. This grant message is also sent to all neighbouring nodes in the network. The source node receives the grant and re-transmits the grant back to the destination node, as a confirmation. This confirmation message is sent to all neighbouring nodes as well. The difference between the coordinated and uncoordinated distributed scheduling methods is that, in coordinated distributed scheduling, the scheduling messages are sent in a collision-free manner, as opposed to the uncoordinated case where collisions may occur.

The result of every node having the scheduling information of its neighbouring nodes is that the distributed scheduling mechanism is more scalable and efficient in terms of connection setup delays and data transmission. Again, this advantage does come at a cost. This time it's the complexity of scheduling mechanism.

5.2.3 Mesh Mode Frame Structure

The centralized and distributed scheduling mechanisms use a TDD frame structure. The frame structure used in mesh mode is illustrated in Figure 5.1:

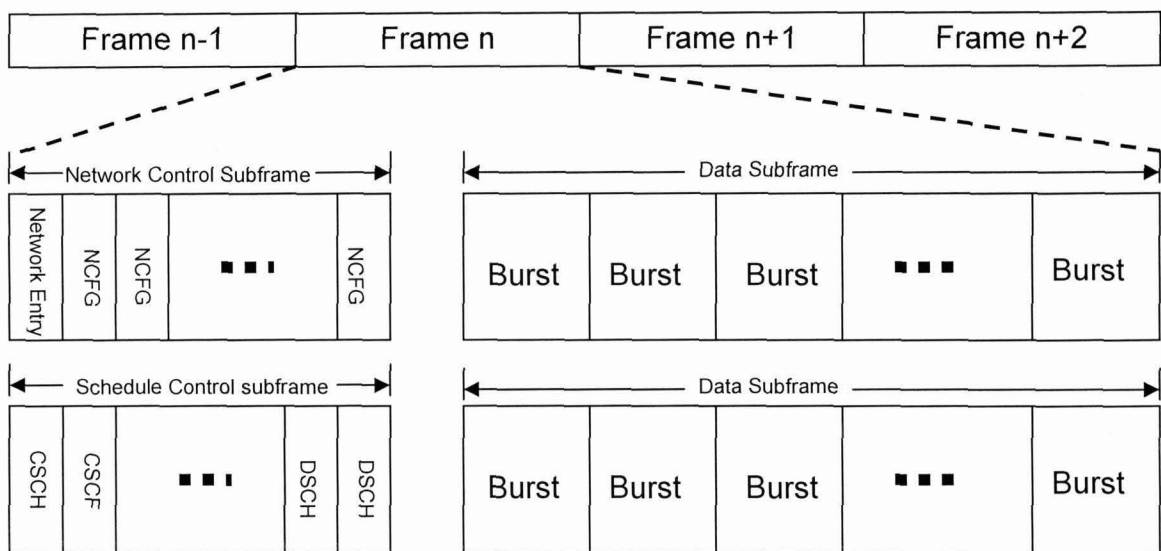


Figure 5.1 - Mesh Mode Frame Structure

It is comprised of a control subframe and a data subframe. The control subframe is used to transmit signalling or management messages, while the data subframe, as the name indicates, transmits data packets. There are two types of control subframes, namely the network control subframe and the schedule control subframe. The network control subframe is periodically transmitted and contains network information to aid new nodes entering the mesh network.

The net throughput on the MAC layer depends on the schedule control subframe and the data subframe. Therefore, these subframes are discussed in more detail below.

Both types of control subframes are divided into sixteen transmission opportunities, which means that the length of the control subframe, `MSH_CTRL_LEN` can have a value between 0 and 15. A transmission opportunity equals seven OFDM symbols. The data subframe is time division multiplexed (TDM) and comprises of variable length physical bursts.

5.2.3.1 Schedule Control Subframe

The schedule control subframe is where the resource allocation occurs. It coordinates the data transfer in the system. It consists of centralized scheduling messages and distributed scheduling messages. The number of distributed scheduling messages is given by network parameter `MSH_DSCH_NUM`. The number of centralized scheduling messages (MSH-CSCH) and centralized configuration messages (MSH-CSCF) in the control subframe is given by length of the control subframe less the length number of distributed scheduling messages, i.e. `MSH_CTRL_LEN - MSH_DSCH_NUM`.

In centralized scheduling (CS), MSH-CSCH messages are used in the three-way handshake, whilst in distributed scheduling (DS), MSH-DSCH messages are used to set up a connection in the request-grant-confirm process. Both CS and DS modes can be deployed at the same time in mesh networks.

5.2.3.2 Data Subframe

The data subframe uses variable length PHY bursts to transmit user data. The composition of the data subframe is shown in Figure 5.2 below.

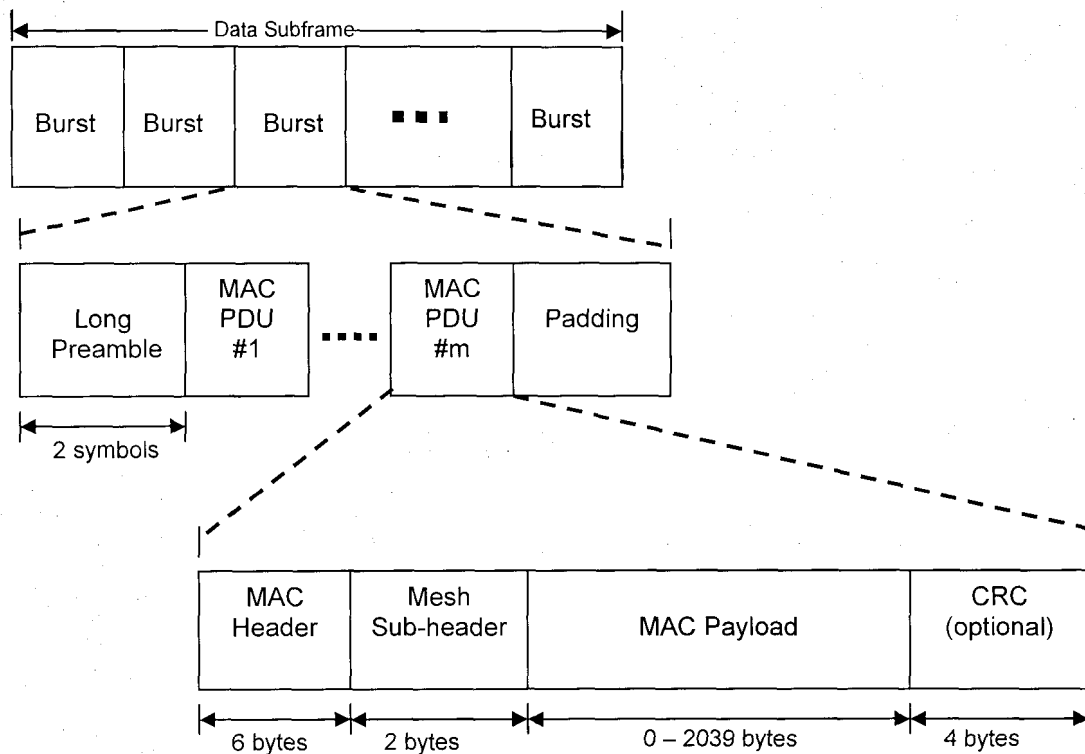


Figure 5.2 - Mesh Mode Data Subframe

The physical bursts that make up the data subframe begin with a long preamble, L_{LP} , of 2 OFDM symbol lengths. The MAC PDUs follow, where the maximum length of a MAC PDU is 2051 bytes.

5.3 Nongoma Solution using Mesh vs. Non-mesh

One of the benefits of mesh networks is its cost-effectiveness. This section compares the cost benefits of using a mesh option to cover the entire Nongoma region, as opposed to a non-mesh case. It is important to note that the strategy adopted in this part of the work is to provide the entire Nongoma region with telecommunication services rather than just Ward 19 as considered earlier. The chosen mesh option is centralized scheduling, because it is easier to manage and more predictable than distributed scheduling. Thus it is more suited to meet the higher traffic demands of the entire Nongoma region. The cost-benefit for first year expenses is considered, since the expenses in the first year of a wireless deployment are the highest based on capital cost.

Figure 5.3 below, shows the entire Nongoma region, and its different wards with the population densities. The area of Nongoma is approximately 2184km². To provide complete wireless coverage for the people of Nongoma, using WiFi for end user access and WiMAX technology for backhaul, it would require approximately six base stations. This figure is obtained by assuming that each BS has

coverage radius of approximately 11km resulting in a coverage area of approximately 380km². Thus to cover an area of 2184km², six BSs are required. It is assumed that the maximum channel data rate of each BS is 40Mbps. In the previous chapter, this data rate was used to support the traffic requirements until 2014. The number of WiFi hotspots that each BS can support is based on the maximum capacity of 40Mbps. The BSs can be strategically placed, depending on the population density, as shown in the example below:

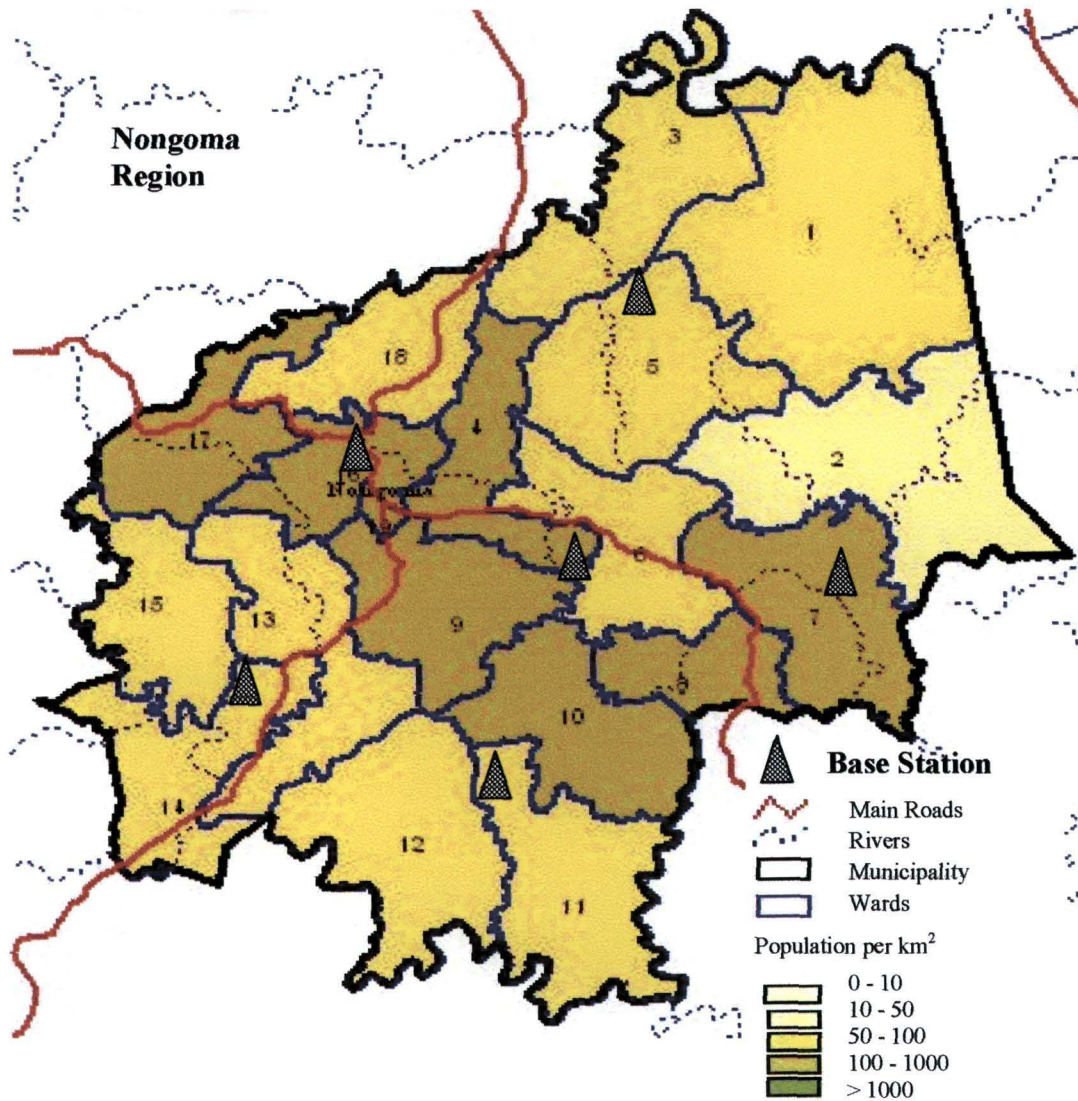


Figure 5.3 - Example of complete wireless coverage for Nongoma region

5.3.1 Non-mesh Case

The non-mesh case implies that each of the six BSs would each require six dedicated lines. The dedicated lines are used to connect to the nearest point where there is routing and switching equipment that allows the BSs to connect to the Internet. This nearest point is termed the point of presence (POP). This case assumes that the nearest POP is 100km away for all BSs. The capacity of each dedicated line is assumed to equal the BS maximum channel capacity.

5.3.2 Mesh Case

The case for the mesh scenario assumes that centralized scheduling is used, thus there is one main BS that is at the centre of the other BSs. The other BSs gain access to the POP via this main BS, since this is the only BS that is connected to the POP via a dedicated wired line. This is how the mesh option is able to provide a cost benefit over the non-mesh option. The main mesh BS, though, needs increased radio capacity as compared to the other BSs. The sectorization technique is used to increase the capacity of the main BS so that it can carry the aggregate traffic from the other BSs as well as users in its own coverage area.

The radio capacity expansion technique of sectorization follows that used by Gunasekaran and Harmantizs [38], where a study was conducted to compare the economics of deploying a WiFi/WiMAX wireless system using non-mesh and mesh topology.

The Nongoma scenario assumes a channel size of 20MHz. Spectral efficiency, which is defined as the number of bits/sec transmitted per Hz of bandwidth, is assumed to be 3 bits/sec/Hz for this study. The spectral efficiency is related to the modulation scheme used and according to a study on the scalability of fixed broadband wireless access network deployment [39], a spectral efficiency of 3 bits/sec/Hz is obtained using 16QAM. Thus, the capacity of each sector for the main BS is 60Mbps. A capacity of 240Mbps is needed, (i.e. 6 BSs * 40Mbps). The required capacity is obtained by using two 180° sectors with four channels. This concept is illustrated in Figure 5.4 below.

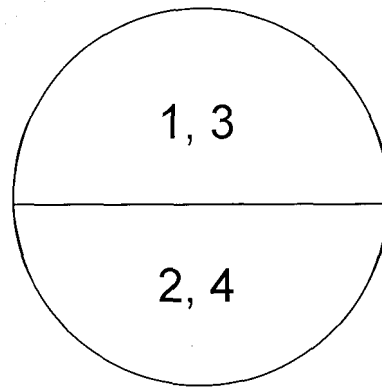


Figure 5.4 – Main BS coverage area with two sectors and four channels

For this mesh case, it is assumed that the backhaul from the 5 WiMAX BSs to the main BS does not require dedicated bandwidth, and thus a dedicated line is not necessary. The assumption is that bandwidth is allocated dynamically on demand. The wired backhaul from the main BS to the POP needs a dedicated line with the same capacity as the main BS. Two such lines are needed for this case. It is also assumed that the cost of the main BS is twice that of the other base stations to account for the sectorization.

5.3.3 Mesh vs. Non-mesh Summary

The mesh option simplifies deployment in terms of wired connections. In addition to a cost savings, the operational costs for future years are reduced. The performance of a mesh network will be lower than the non-mesh based option, and the mesh deployment will be based on traffic demands of the region.

5.4 Nongoma Scenario: Mesh Benefit of Coverage Extension

A small amount of meshing in an 802.16 network can result in a large improvement in coverage, up to several kilometres. A typical case for coverage extension in the Nongoma area is presented below.

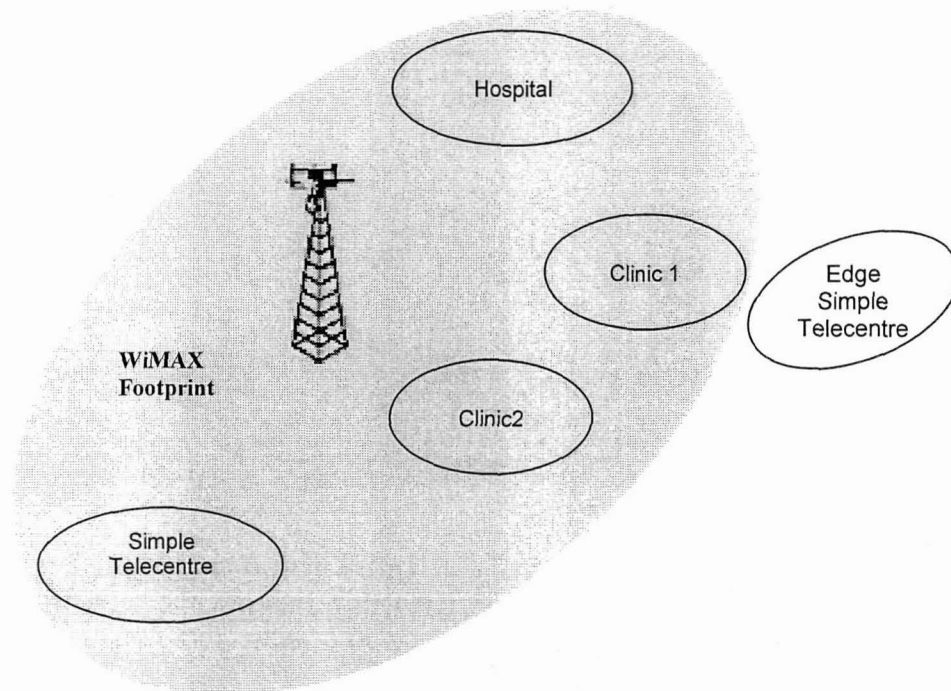


Figure 5.5 – Coverage Extension WiMAX Backhaul

Figure 5.5 above, shows a simple telecentre on the edge of the WiMAX network. This telecentre can still benefit from the backhaul network if 802.16 mesh mode is used.

Work done by S Redana and M Lott [40], provides an analytical evaluation of the mesh mode MAC layer, in terms of efficiency. Both CS and DS scheduling were considered. Section 5.2 above outlined that in centralized scheduling the MSH-CSCH messages are used in the three-way handshake process to set up a connection between nodes. The MSH-CSCH messages relay requests, grants and confirmations. The efficiency for CS mode depends on the number nodes that transmit the MSH-CSCH messages. In [40], the efficiency in CS mode is calculated only for the case where the grant MSH-CSCH messages are sent. In other words the overhead of the request and confirmation MSH-CSCH messages are not considered.

The overhead of the MSH-CSCF messages used in CS mode also affects the efficiency. In [40] a solution is presented for the centralized scheduling where the overhead of the MSH-CSCF message is taken into account for every frame transmitted, as well as for the case where the overhead of the MSH-CSCF message was not considered at all.

In distributed scheduling, the MSH-DSCH messages are used for relaying requests, grants and confirmations of the three-way handshake process. Similar to CS mode, the efficiency for DS mode depends on the number of nodes that transmit the MSH-DSCH messages. The efficiency for DS mode was also obtained by only considering the overhead of the grant MSH-DSCH message.

This research aims to use the equations derived in [40], to determine the net throughput on the MAC layer for CS and DS modes. The efficiency calculation will also take into account the overhead of the request and confirmation MSH-CSCH and MSH-DSCH messages.

Also, a more realistic case is considered for the CS mode, where the MSH-CSCF message is sent in only half of the frames. In addition a variable payload size is considered in this work.

5.5 Efficiency Analysis

The definitions in this section follow the definitions in [40]. The maximum efficiency is defined by the equation 5.1, below, where m is the PHY mode index that refers to the modulation and coding rate used.

$$\max[\eta]^m = \max_{k \in [13; 2051]} \left\{ \frac{\Theta_{Net/MAC}^{k,m}}{\Theta_{PHY/symbol}} \right\} \quad (5.1)$$

where:

k is the MAC PDU length in bytes

$\Theta_{Net/MAC}^{k,m}$ is defined as the maximum throughput on the MAC layer for a particular value of m and payload size k .

$\Theta_{PHY/symbol}$ is the maximum throughput/OFDM symbol

Six possible types of modulation and coding rates that could be used are shown in Table 5-1 below. The bytes per symbol (BpSm) parameter refer to the uncoded block size.

m	Modulation	Bytes per Symbol (BpSm)	Coding Rate
1	QPSK	24	1/2
2	QPSK	36	3/4
3	16QAM	48	1/2
4	16QAM	72	3/4
5	64QAM	96	2/3
6	64QAM	108	3/4

Table 5-1: Channel Coding per modulation

$$\Theta_{PHY / symbol} = \frac{BpSm * 8}{T_{symbol}} \quad (5.2)$$

where:

T_{symbol} is OFDM symbol duration

$$\Theta_{Net / MAC}^{k,m} = \frac{(N_{MACPDU}^{k,m}) * (L_{payload}^k) * (8)}{T_{FRAME}} \quad (5.3)$$

where:

T_{FRAME} is the frame duration

N_{MACPDU} is the number of MAC PDUs in a frame. Since a frame is comprised of the control subframe and data subframe, it means that the number of PDUs in a frame is dependent on the overhead of these subframes. Following [40] N_{MACPDU} is defined as:

$$N_{MACPDU} = \left\{ \left(\left(\frac{L_{DS}}{L_{minislot}} \right) - (n_c * L_{LP}) \right) * \frac{L_{minislot}}{k / BpSm} \right\} \quad (5.4)$$

where:

L_{LP} is length of the long preamble which is equal to 2 OFDM symbols

n_c is the number of active connections

$L_{minislot}$ is the resource allocation size defined in OFDM symbols as:

$$L_{minislot} = \frac{L_{DS}}{256} \quad (5.5)$$

L_{DS} is the number of transmission opportunities in the data subframe, given by

$$L_{DS} = N_{symbols} - L_{CS} \quad (5.6)$$

$N_{symbols}$ is the number of OFDM symbols in a frame, obtained by the frame duration over the symbol duration.

L_{CS} is the duration of the control subframe expressed as number of symbols:

$$L_{CS} = 7 * MSH_CTRL_LEN \quad (5.7)$$

The MSH_CTRL_LEN is dependent on the MSH-CSCH, MSH-CSCF and MSH-DSCH messages and are derived separately for centralized and distributed scheduling.

5.5.1 Centralized scheduling

The scenario for centralized scheduling is depicted below. Resource allocation for each of the telecentres is done by the base station. Using centralized scheduling, the coverage of the WiMAX backhaul network is extended to include the edge simple telecentre. Clinic 1 could be for instance the intermediate node that relays communication messages between the MBS and edge telecentre. Thus a 2-hop scenario is considered.

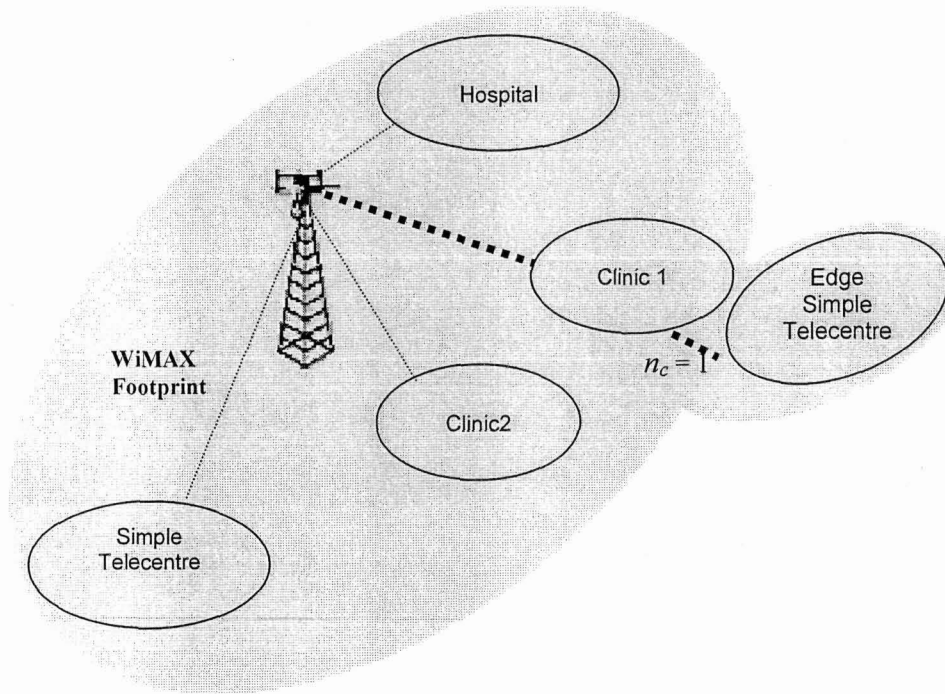


Figure 5.6 – Centralized Scheduling as means for Range Extension

The control subframe overhead is described by the following equations:

$$MSH_CTRL_LEN = \left(\frac{(L_{MSH-CSCF} + L_{MSH-CSCH}) * (\#SS)}{7} \right) + n_c \tag{5.8}$$

where:

MSH_CTRL_LEN is defined as the number of available OFDM symbols for transmission in the control subframe

$L_{MSH-CSCF}$ and $L_{MSH-CSCH}$ are lengths of the MSH-CSCF and MSH-CSCH messages respectively.

$\#SS$ is the number of SSs that transmit the MSH-CSCH message.

This research assumes that all nodes involved in the communication transmit the message. For the above scenario #SS would be 3 nodes, i.e. the edge Simple telecentre, the Clinic 1 telecentre, and the MBS. This is to account for the request, grant and confirmation messages.

To ensure that the MSH-CSCF message is only transmitted in half of all the frames:

$$MSH_CTRL_LEN = \left(\frac{(L_{MSH-CSCF} + L_{MSH-CSCH}) * (\#SS)}{7} \right) + \text{ceil} \left(\frac{n_c}{2} \right) \quad (5.9)$$

where:

$$L_{MSH-CSCF} = 7 * \left[\frac{OH_{MSH-CSCF}}{4 * BpS1 - OH_{MACPDU}} \right] \quad (5.10)$$

$$L_{MSH-CSCH} = 7 * \left[\frac{OH_{MSH-CSCH}}{4 * BpS1 - OH_{MACPDU}} \right] \quad (5.11)$$

where:

$OH_{MACPDU} = 12$ bytes to account for the MAC header, mesh subheader and CRC

$$OH_{MSH-CSCF} = 3 + \left(\frac{N_{CH}}{2} \right) + (2 * N_{flow}) + (3 * N_{NODE}) \quad (5.12)$$

$$OH_{MSH-CSCH} = 4 + N_{flow} \quad (5.13)$$

Here N_{NODE} is defined as the number of SS's that are directly controlled by the MBS. So for the above scenario, this is the Simple telecentre, the two Clinics and the Hospital, and thus, $N_{NODE} = 4$.

$$N_{flow} = \sum_{i=1}^{N_{NODE}} N_{CHILD}^i \quad (5.14)$$

where:

N_{CHILD}^i is the number of neighbours with a hop distance that is one higher than that of SS^i from the MBS. Applied to the above scenario, there are four SSs under direct control of the MBS (i.e. the Hospital, Clinic 1, Clinic 2 and the Simple telecentre), but only has a neighbour with a hop distance that is one higher than that of the SS. This SS is Clinic 1, thus $N_{CHILD}^1 = 1$.

5.5.2 Distributed Scheduling

The distributed scheduling scenario is illustrated below. It shows the edge Simple telecentre, communicating with the Hospital, via Clinic 1. This is possible with distributed scheduling. The

MBS is not involved in the transfer of application data between these nodes. This is also a 2-hop scenario.

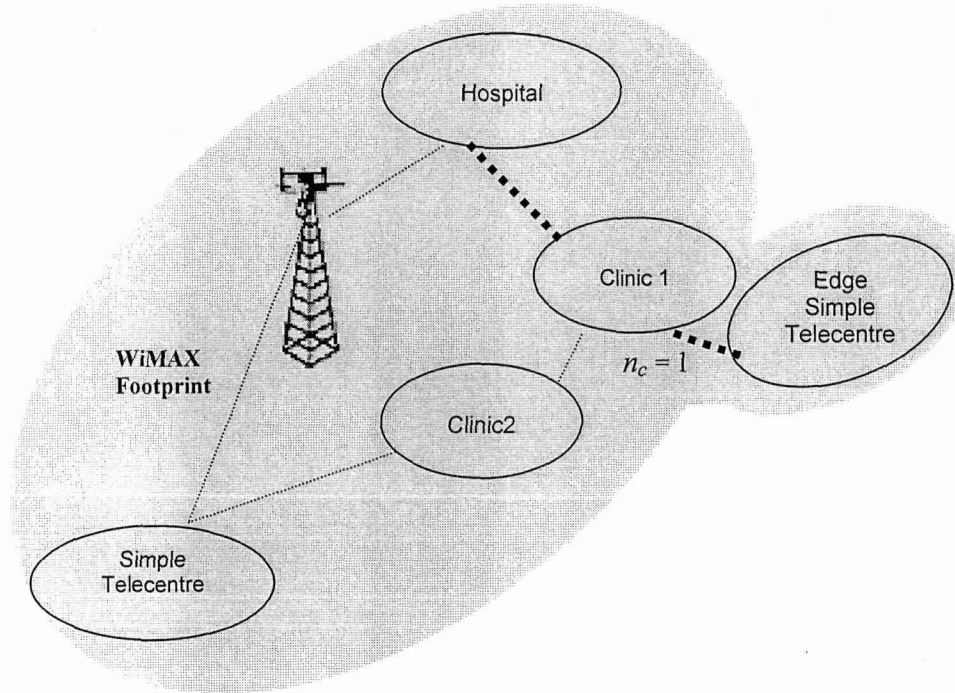


Figure 5.7 – Distributed Scheduling as a means for Range Extension

Again in this section the definitions follow [40]. The MSH_CTRL_LEN is defined as:

$$MSH_CTRL_LEN = \frac{(L_{MSH-DSCH}) * (\#SS)}{7} \quad (5.15)$$

where:

$\#SS$ is the number of nodes that transmit the MSH-DSCH messages. Like in the centralized scheduling case, it would be all the nodes involved in the communication, i.e. 3 nodes for the above scenario

$L_{MSH-DSCH}$ is the length of MSH-DSCH message:

$$L_{MSH-DSCH} = 7 * \left[\frac{OH_{MSH-DSCH}}{4 * BpS1 - OH_{MACPDU}} \right] \quad (5.16)$$

where:

$OH_{MSH-DSCH}$ is the overhead for the MSH-DSCH message expressed as number of OFDM symbols:

$$OH_{MSH-DSCH} = 6 + (3 * N_{SCHED}) + (2 * N_{REQUEST}) + (4 * N_{AVAILABILITY}) + (5 * N_{GRANT}) \quad (5.17)$$

here:

N_{SCHED} is the number of neighbours that also receive the DS message

$N_{REQUEST}$ is the number of requests in the message

$N_{AVAILABILITY}$ is the number of availabilities reported in the MSH-DSCH message

N_{GRANT} is the number of grants in the message

In [40], the efficiency was considered for the number of grants reported. This research, considers the requests, availabilities and grants, as well as other neighbours. It assumes that the number of requests equals the number of availabilities equals the number of grants.

$$N_{REQUEST} = N_{AVAILABILITY} = N_{GRANT} = n_c + (n - 1) \quad (5.18)$$

where:

n is the number of hops

So for the above scenario, the edge Simple telecentre makes a request to the Hospital telecentre via Clinic 1, and broadcasts this request and availabilities to the neighbour nodes as well, i.e. Clinic 2, the Simple telecentre and the MBS. This is in accordance with the DS regulations. The hospital grants the request, and again sends it to all the nodes in the network, not only the nodes involved in the communication. The edge Simple telecentre confirms the request, by again broadcasting it to all nodes.

5.6 Results

The physical parameters that were used are listed in Table 5-2 below. The source data for all graphs can be found in Appendix B.

N_{FFT}	256
Bandwidth	20MHz
Frame Duration (T_{frame})	10ms
Modulation	QPSK
Coding Rate	$\frac{3}{4}$
T_{symbol}	13.7 μ s
N_{symbol}	729

Table 5-2: PHY Parameters

This work was analytically evaluated and simulations were not included.

Figure 5.8 below shows the maximum efficiency as a function of the payload size, for a 2-hop scenario. The variable payload size considers the different types of applications that could be used, for instance email application files sizes use a smaller payload size, whereas the file sizes of FTP applications are much larger and therefore use a larger payload size such as 1292 bytes or 2051 bytes. The number of active connections, n_c , is limited to 1 for this result. It means that there is only one node on the edge (edge Simple telecentre), trying to benefit from the WiMAX backhaul network via the Clinic 1 telecentre.

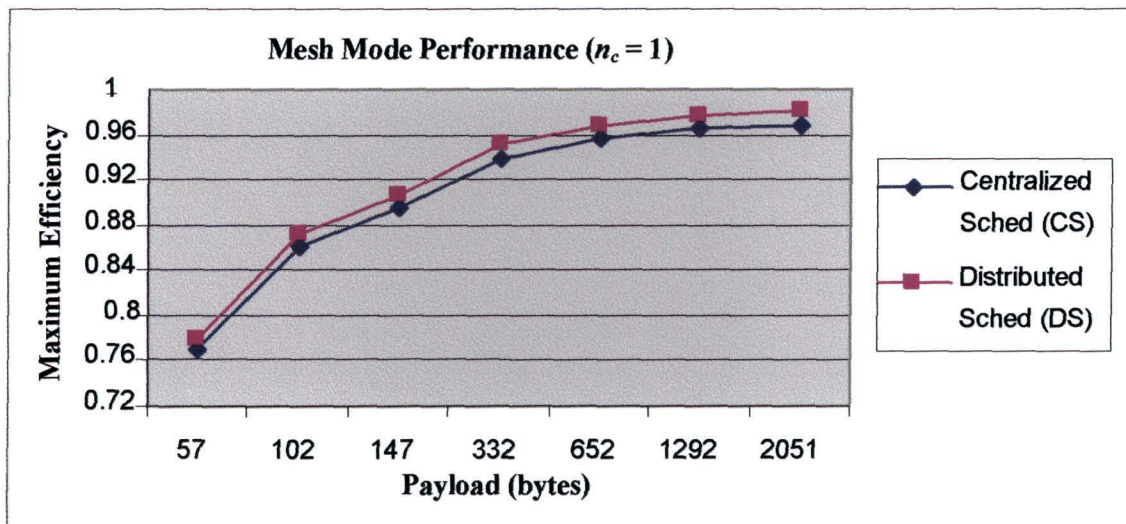


Figure 5.8 – Maximum Efficiency vs. Payload for $n_c = 1$

For the 2-hop scenario, it can be seen that the distributed scheduling slightly outperforms the centralized scheduling, for any payload size. This can be attributed to the mesh CS mode taking into account the overhead introduced by the MSH-CSCF message. One can also see that efficiency is at its peak when maximum payload size is used. This means that there is very little overhead when the payload is large.

The Nongoma scenario for CS and DS modes assumes that the maximum value for $n_c = 6$. This means that at any time there is a maximum of six edge nodes that will be able to benefit from the WiMAX backhaul network via a single SS within the WiMAX footprint. These edge nodes are not necessarily other telecentres and could be individual subscribers, possibly even a business, and since Nongoma is a rural area, it assumed that there will not be many individual subscribers. Thus a low maximum value for n_c is assumed. The performance for $n_c = 6$ is shown in Figure 5.9 below.

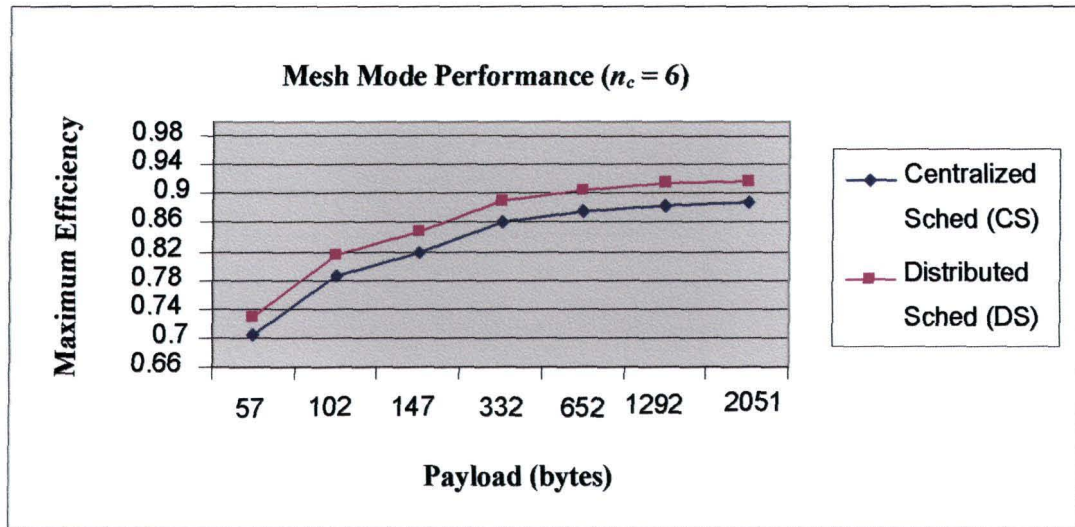


Figure 5.9 - Maximum Efficiency vs. Payload for $n_c = 6$

Comparing Figure 5.8 and Figure 5.9, one can see that by increasing the number of edge nodes, i.e. n_c , results in a decrease in the efficiency for both CS and DS modes. The efficiency for CS mode is decreased by approximately 8% and for DS mode the decrease is approximately 6%. The maximum efficiency of DS mode decreases slower than CS mode when the number of connections increases from $n_c = 1$ to $n_c = 6$. This is because in centralized scheduling the MSH_CTRL_LEN overhead is dependent on the number of nodes under direct control of the MBS, in the network. The two-hop scenario considered in this research for $n_c = 1$ and $n_c = 6$ has four nodes under direct control of the MBS, i.e. the Hospital, Clinic 1, Clinic 2 and the Simple telecentre. The overhead calculation for the MSH-CSCF message, given by equation 5.12 above, takes into account the number of SSS under direct control of the MBS, i.e. N_{NODE} . Thus the overhead for the MSH-CSCF message is still very high, even though the MSH-CSCF message is only sent in half of the frames.

5.7 Summary

This chapter presented a high level cost analysis of a WiFi/WiMAX network for the entire geographic area of Nongoma. A non-mesh and centralized mesh topology was considered. It was shown that the centralized mesh topology reduced the cost of backhaul for the network, by aggregating the traffic and using the wired backhaul more efficiently. Reducing the cost of backhaul for the Nongoma region, will in effect make the wireless service offered more affordable.

This chapter also included an efficiency analysis of the two scheduling methods used in IEEE 802.16 mesh mode, namely centralized and distributed scheduling. This was based on work done by

S Redana and M Lott in [40]. Results obtained show that distributed scheduling outperforms centralized scheduling for coverage extension in the Nongoma scenario.

6 CONCLUSION

6.1 Problem Outline

In the research conducted, a telecentre system based on wireless technology was specified as a telecommunications solution for a typical South African rural region, namely Nongoma, so that the concept of rural connectivity may become a reality. The ultimate goal of rural connectivity is that the telecommunications infrastructure in rural regions is developed such that people living in these areas are able to easily access telephones, as well as access the Internet, and thus enable them to improve their quality of life and make a meaningful contribution to the economy.

Telecommunications infrastructure is needed in order for rural connectivity to become a reality. Thus, broadband wireless technology, such as WiFi and WiMAX were identified as an affordable and equitable means to provide the infrastructure boost in the Nongoma region. The number of possible rural subscribers in Ward 19 and surrounding areas of Nongoma were estimated. A traffic estimate was also determined for data, Internet voice and video applications over an eight year period namely, 2006 -2014. The network growth was found to be exponential for all of the supported services. This growth was based on technological, social as well as economic factors.

Providing reliable high quality data and multimedia communications over a large scale network such as Ward 19 and surrounding areas in Nongoma is a complex task. The goal of this research was to investigate the performance of the WiMAX backhaul for the Ward 19 telecentre system in Nongoma. This was done using the estimated traffic profile for the region. The performance of the network was assessed under five simulation scenarios for a fixed wireless PMP network. The simulation investigations were conducted for an ideal channel case1 and case 2, which assumed ideal physical layer conditions, a non-ideal channel and performance with ARQ enabled and results were obtained in terms of delay and IP packet loss.

6.2 Research Findings

The voice and video end-to-end delays for all simulation scenarios were found to be within their suggested ITU limits and delay budget used in this research, i.e.100ms for video and 150ms for voice. Therefore it can be said that this type of traffic can be sustained on a WiMAX network.

Investigations conducted for the ideal WiMAX channel showed the service differentiation, between data (using the BE scheduling service) and multimedia traffic (using the UGS and rtPS scheduling

services). This is because the scheduling done at the BS uses fair scheduling algorithms. In this case the MDRR scheduling algorithm was used. It was also found that using the nrtPS resulted in a 50% improvement in the download response times of the data traffic. Thus, it is recommended to that this scheduling service be used for data traffic.

When a 5% PDU dropping probably was introduced into the system on the UL and DL to emulate PHY losses such as fading, the total network load decreased by approximately 11%. The effect of the transport protocol delay, i.e. TCP delay was also seen in this investigation. TCP delay significantly impacts the end-to-end QoS in terms of delay, because the download response times for the data traffic increased by 10 times. This indicated that some type of error correction mechanism needed to be included such as ARQ, so as to make the wireless channel appear more reliable to the transport layer.

This lead to the next investigation, where the use of ARQ was considered. Using ARQ caused the download response time to decrease by 50%, but the packet loss was still approximately 4%. This can be explained by the large amount of multimedia traffic on the network, also subject to the packet loss but don't have ARQ enabled. Thus, as a recommendation hybrid-FEC schemes could be used to improve the performance of multimedia traffic.

A WiMAX mesh based solution was proposed to cover the entire Nongoma region. The cost benefit of WiMAX mesh solutions for rural coverage has been investigated in [38]. In this work several strategies that enhance cost effectiveness have been investigated. For example, the CS and DS mesh modes allow the coverage of the BS to improve. The investigation found that using DS mode was more effective than CS mode to extend the coverage for the Nongoma scenario. Mesh mode also allows alternate paths in the communication system.

The work conducted in this research can be used as a basis to provide a telecommunications solution for other rural regions in South Africa. The results from the investigations carried out show that wireless technologies, namely WiFi and WiMAX can provide an efficient and cost-effective solution for rural subscribers.

6.2 Future Work

Further research can be conducted in the use of hybrid-FEC schemes to improve the performance of multimedia traffic on a WiMAX network, in particular the Nongoma telecentre system, with its user traffic profile. An investigation can be conducted to determine the packet loss improvement, when using this error correction method in a non-ideal channel.

The investigation on adaptive modulation, showed the trade-off between coverage and capacity. This is an important issue for the Nongoma case. The Hospital and Clinic 2 telecentre closest to the BS can obtain the maximum channel data rate possible by using a higher order modulation, such as the 16 QAM used in this scenario, while the Simple telecentre and Clinic 1 that are further away use a more robust modulation scheme such as QPSK, so that they are still covered by the WiMAX footprint, but at a lower data rate. Thus, the adaptive modulation technique allows for a balance between coverage and capacity. The quantitative analysis for this is left as future work.

The mesh based solution presented a cost-effective coverage for the Nongoma scenario, but further work can be conducted in this area, where economic modelling can be used to determine actual cost savings based on the traffic estimates for the region..

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APPENDIX A: Source Data for Graphs in Chapter 4

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	24.19	24.14	24.6	23.89	24.77	22.94	24.59	35.12	46.11
	UGS	14.12	14.09	14.16	13.95	14.11	14.07	13.95	13.90	13.43
	rtPS	25.61	24.50	25.12	24.41	24.87	23.83	26.77	28.03	26.17

Table A-1: Ideal Channel Case 1: Average UL Delay vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	6.087	6.53	9.390	7.291	7.276	7.186	34.17	35.12	46.11
	UGS	5.646	5.411	5.457	5.434	5.254	5.052	5.266	5.132	5.454
	rtPS	6.037	7.524	9.255	7.666	8.603	7.629	11.56	13.66	13.69

Table A-2: Ideal Channel Case 1: Average DL Delay vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Voice	86.71	86.56	86.41	86.67	86.50	86.66	87.29	87.10	87.95
	Video	19.01	18.20	21.37	18.63	19.95	18.66	29.46	33.80	33.57

Table A-3: Ideal Channel Case 1: Average End-to-End Delay vs. Total Network Load for each year

Delay (s)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Email	0.324	0.304	0.372	0.412	0.413	0.473	0.858	1.122	1.441
	FTP	0.543	0.581	0.829	1.066	1.044	1.762	3.596	3.779	4.183
	Web-Browsing	1.862	1.899	2.121	1.974	1.969	1.899	4.378	5.614	12.523

Table A-4: Ideal Channel Case 1: Average Download Response Time vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	24.53	24.60	25.33	24.15	24.94	23.61	24.76	29.45	35.13
	nrtPS	36.51	36.22	38.67	37.75	35.58	36.39	37.41	33.74	34.51
	UGS	14.17	14.15	14.07	14.15	14.10	14.04	13.76	13.95	13.93
rtPS	25.87	24.52	25.32	24.5	24.82	24.05	26.70	27.94	29.18	

Table A-5: Ideal Channel Case 2: Average UL Delay vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	6.52	6.70	9.12	7.42	8.20	9.37	34.77	36.39	55.84
	nrtPS	6.204	5.702	6.366	5.840	6.298	6.178	6.545	8.581	8.697
	UGS	5.469	5.455	5.013	4.269	5.399	4.998	4.671	5.586	4.791
rtPS	7.452	7.522	9.162	7.556	8.939	7.686	11.50	14.97	13.84	

Table A-6: Ideal Channel Case 2: Average DL Delay vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Voice	86.51	86.46	86.32	85.89	86.26	86.50	86.99	87.91	87.88
	Video	19.71	18.45	21.64	18.40	20.49	18.81	29.35	35.58	35.35

Table A-7: Ideal Channel Case 2: Average End-to-End Delay vs. Total Network Load for each year

Delay (s)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Email	0.293	0.332	0.371	0.378	0.433	0.477	0.835	1.224	2.093
	FTP	0.665	0.697	0.694	0.726	1.655	2.713	3.625	3.763	4.477
	Web-Browsing	2.015	1.974	2.148	2.01	2.081	2.059	3.767	4.784	5.039

Table A-8: Ideal Channel Case 2: Average Download Response Time vs. Total Network Load for each year

Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
% Packet Loss	0.008	0.008	0.008	0.008	0.008	0.009	0.009	0.009	0.012

Table A-9: Ideal Channel Case 2: Percentage Packet Loss vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	25.45	25.19	25.17	24.91	26.04	24.22	25.60	26.04	32.87
	nrtPS	32.86	33.66	31.40	33.90	31.18	31.34	31.56	30.77	30.38
	UGS	14.51	14.12	14.10	14.15	14.13	14.09	14.06	14.01	13.16
	rtPS	25.85	24.54	24.40	24.57	25.00	24.03	25.19	26.55	29.03

Table A-10: Non-Ideal Channel: Comparison between average UL WiMAX Connection Delays of each service flow for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	7.41	7.06	7.07	7.41	7.78	8.42	17.21	27.04	26.77
	nrtPS	6.691	5.778	6.298	6.126	6.242	6.198	7.106	7.569	7.446
	UGS	5.531	5.474	5.369	5.651	5.549	5.089	5.465	5.789	4.804
	rtPS	6.940	7.505	7.556	7.504	9.034	7.626	12.27	13.61	13.62

Table A-11: Non-Ideal Channel: Comparison between average DL WiMAX Connection Delays of each service flow for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Voice	86.46	86.60	86.47	86.71	86.70	86.56	87.49	87.88	87.87
	Video	19.30	18.42	18.61	18.28	21.51	18.74	31.31	33.51	35.15

Table A-12: Non-Ideal Channel: Average End-to-End Delay of real-time traffic on network vs. Total Network Load for each year

Delay (s)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Email	5.771	4.222	4.351	12.375	9.876	7.905	14.54	8.197	11.32
	FTP	6.545	3.423	15.59	12.65	16.43	17.64	29.12	32.25	49.94
	Web-Browsing	17.91	17.24	18.28	18.84	18.10	20.71	21.09	21.19	25.92

Table A-13: Non-Ideal Channel: Download Response Time of data traffic on network vs. Total Network Load

# Retransmissions	Simulation Time	200	240	280	320	360	400	440	480
	Ideal	1	7	5	10	9	2	3	4
	Non-ideal	1	37	17	22	35	19	35	26

Table A-14: Number of TCP retransmissions over simulation time for ideal channel Case 2 and non-ideal channel in 2006

Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
% Packet Loss	5.281	5.384	5.244	5.399	5.500	5.312	6.298	6.5	6.746

Table A-15: Non-Ideal Channel: Percentage Packet Loss vs. Total Network Load for each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	441.2	352.6	378.2	507.8	457.6	444.8	386.9	614.7	373.2
	nrtPS	32.89	39.65	33.49	36.08	32.06	46.45	31.01	35.24	34.18
	UGS	14.16	14.13	13.09	14.13	14.13	14.2	14.08	13.98	13.89
	rtPS	26.38	24.47	24.46	24.53	24.83	24	24.71	26.58	27.97

Table A-16: ARQ: Comparison between average UL Delays of different service flows for Total Network Load of each year

Delay (ms)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	BE	399.4	507.18	150.4	311.1	292.4	565.6	495.7	570.6	573.9
	nrtPS	13.73	12.45	12.89	8.575	12.93	12.7	16.6	13.38	23.91
	UGS	5.575	5.475	5.408	5.45	5.538	5.05	5.089	5.1	5
	rtPS	7.75	7.5	7.58	7.5	8.942	7.033	11.12	13.18	15.03

Table A-17: ARQ: Comparison between average UL Delays of different service flows for Total Network Load of each year

Delay (s)	Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
	Email	3.826	3.582	4.625	3.855	3.915	4.424	10.461	7.095	12.832
	FTP	3.904	3.661	1.462	4.807	6.279	5.992	11.569	10.408	19.654
	Web-Browsing	17.24	15.44	16.03	14.75	17.68	16.27	17.54	14.93	21.81

Table A-18: ARQ: Download Response Time of data traffic vs. Total Network Load for each year

# Retransmissions	Simulation Time	200	230	260	290	320	350	380	410	440	470
	ARQ	5	31	24	21	16	21	25	28	25	21
	Non-ideal	1	37	17	22	35	19	35	32	35	26

Table A-19: Number of TCP retransmissions over simulation time for non-ideal channel and non-ideal channel with ARQ enabled in 2006

Year	2006	2007	2008	2009	2010	2011	2012	2013	2014
% Packet Loss	10.9	11	11.4	12.8	13.3	19.3	23.4	25	29.4

Table A-20: ARQ: Total percentage Packet Loss vs. Total Network Load for each year

APPENDIX B: Calculations for Adaptive Modulation

B.1 System Parameters:

Operation Frequency $F_c = 3.5\text{GHz}$

Channel Bandwidth $BW = 20\text{MHz}$

Total Transmission Power $P_t = 35\text{dBm}$

Receive Antenna Gain $G_t = 15\text{dBi}$

$N_{\text{subchannels}} = 16$ (no subchannel)

Modulation = QPSK, 16 QAM, 64 QAM

Bits per symbol $b_m = 2$ (QPSK), 4 (16 QAM), 6 (64 QAM)

Coding Rate $c_r = 3/4$

Guard Time $G = 1/4$

Sampling Frequency $N = 57/50$ for BW multiple of 2.0MHz

$N_{\text{used}} = 192$

$N_{\text{FFT}} = 256$

The calculations for the maximum transmission data rate and maximum coverage use the methodology and equations defined in a study on coverage prediction and performance evaluation of wireless metropolitan area networks based on IEEE 802.16 in [33], as well as equations defined in the IEEE 802.16 Standard and assume ideal channel conditions.

B.1.1 Data Transmission Rate

The IEEE 802.16 Standard defines the maximum transmission data rate R for the OFDM physical layer as:

$$R = \frac{N_{\text{used}} * b_m * c_r}{T_s} \quad (\text{B. 1})$$

where:

T_s is the OFDM symbol time given by:

$$\begin{aligned} T_s &= T_g + T_b \\ &= (G + 1) T_b \end{aligned} \quad (\text{B.2})$$

where:

T_b is the useful symbol time given by:

$$T_b = 1/\Delta f \quad (B.3)$$

where:

Δf is the subcarrier spacing given by

$$\Delta f = F_s/N_{FFT} \quad (B.4)$$

where:

F_s is the sampling frequency in MHz in by:

$$F_s = \frac{8000 * n * BW}{8000} \quad (B.5)$$

B.1.2 Coverage Prediction

The maximum propagation loss in a system is determined using the following link budget equation:

$$P_{r, \min} = P_t + G_t - L_t$$

where:

$P_{r, \min}$ is the receiver sensitivity in dBm

P_t is the total transmission power

G_t is the total system gain

L_t is total system loss

The receiver sensitivity $P_{r, \min}$ (dBm), is defined in [33] as:

$$P_{r, \min} = SNR + 10 \log (W) + F + N_0 \quad (B.6)$$

where:

N_0 is the thermal noise level in dBm defined as

$$N_0 = 10 \log (kT/10^{-3}) \quad (B.7)$$

where:

k is the Boltzmann's constant = 1.38×10^{-23}

T is temperature in Kelvin (a value of 290K is used)

F is the noise figure in dB (a value of 12dB is used as defined by the IEEE 802.16 standard)

W is the effective bandwidth given by:

$$W = \frac{F_s * N_{used} * N_{subchannels}}{16 * N_{FFT}} \quad (B.8)$$

SNR is the required SNR as defined by the IEEE 802.16 standard. Table A-1 below shows the required SNR for each modulation scheme:

Modulation Scheme	Coding Rate	SNR (dB)
BPSK	1/2	6.4
QPSK	1/2	9.4
	3/4	11.2
16 QAM	1/2	16.4
	3/4	18.2
64 QAM	2/3	22.7
	3/4	24.4

Table B-1: Required SNR for each modulation scheme and coding rate as defined by IEEE 802.16 Standard

Thus, according to [33], the receiver sensitivity $P_{r, min}$ is given by:

$$P_{r, min} = -102 + SNR + 10 \log \left(\frac{F_s * N_{used} * N_{subchannels}}{16 * N_{FFT}} \right) \quad (B.9)$$

Since LOS conditions are considered, the total system loss is given by the free space loss in dB is defined in [33] as:

$$L_0 = 32.45 + 20 \log (F_c) + 20 \log (d) \quad (B.10)$$

where:

F_c is the operation frequency

d is the distance in km between the SS and BS

An example calculation for QPSK modulation is shown below:

$$F_s = (57/50) * (20\text{MHz}) = 22.8\text{MHz}$$

$$\Delta f = F_s/256 = 89.0625\text{kHz}$$

$$T_b = 1/\Delta f = 11.23\mu\text{s}$$

$$T_s = T_b + G(T_b) = 14.035\mu\text{s}$$

$$R = \frac{N_{\text{used}} * b_m * c_r}{T_s}$$

$$= \frac{\left(192 * 2 * \left(\frac{3}{4}\right)\right)}{14.035 \mu\text{s}}$$

$$= \underline{20.52 \text{ Mbps}}$$

$$P_{r, \text{min}} = -102 + \text{SNR} + 10 \log \left(\frac{F_s * N_{\text{used}} * N_{\text{subchannels}}}{16 * N_{\text{FFT}}} \right)$$

$$= P_{r, \text{min}} = -102 + 11.2 + 10 \log \left(\frac{22.8 * 192 * 16}{16 * 256} \right)$$

$$= -78.47 \text{ dBm}$$

$$P_{r, \text{min}} = P_t + G_t - L_t$$

$$-78.47 = 35 + 15 - (32.45 + 20 \log(3500) + 20 \log d)$$

$$-78.47 - 35 - 15 + 32.45 + 70.88 = -20 \log d$$

$$-25.14 = -20 \log d$$

$$d = \underline{18.07 \text{ km}}$$

Similarly, the data rates and coverage diameters were obtained for the other modulation types. The coverage diameter is 2*d. These values are shown in the Table A-2 below:

Modulation	Max Data Rate R (Mbps)	Distance d (km)	Coverage Diameter (km)
QPSK	20.52	18.07	36.14
16 QAM	41.04	8.01	16.02
64 QAM	61.56	3.95	7.9

Table B-2: Maximum data rate and coverage for different modulation schemes

APPENDIX C: Source Data for Graphs in Chapter 5

Max Efficiency	Payload (bytes)	57	102	147	332	652	1292	2051
	CS	0.770	0.860	0.895	0.940	0.957	0.966	0.970
	DS	0.779	0.872	0.907	0.951	0.969	0.978	0.981

Table C-1: Maximum Efficiency vs. Payload for $n_c = 1$

Max Efficiency	Payload (bytes)	57	102	147	332	652	1292	2051
	CS	0.705	0.788	0.811	0.861	0.876	0.884	0.888
	DS	0.729	0.815	0.848	0.891	0.907	0.915	0.918

Table C-2: Maximum Efficiency vs. Payload for $n_c = 6$