DISSERTATION TITLE

EXTENDING WIFI ACCESS FOR RURAL REACH

SUBMITTED BY

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DECLARATION

I, Kribashnee Naidoo, Student Number 200202476, hereby declare that the dissertation entitled Extending WiFi Access for Rural Reach is a result of my own investigation and present my work unless specifically referenced in the text. This work has not been submitted in part or in full for any degree to any university.

Signed:____________________

Name:____ Date:____________________

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ABSTRACT

WiFi can be used to provide cost-effective last-mile IP connectivity to rural users. In initial roll-out, hotspots or hotzones can be positioned at community centres such as schools, clinics, hospitals or call-centres. The research will investigate maximizing coverage using physical and higher layer techniques. The study will consider a typical South African rural region, with telecommunications services traffic estimates. The study will compare several IEEE 802.11 deployment options based on the requirements of the South African case in order to recommend options that improve performance.

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LIST OF ABBREVIATIONS

ACK	Acknowledgement
AP	Access Point
BPSK	Binary Phase Shift Keying
BSS	Basic Service Set
BW	Bandwidth
BWA	Broadband Wireless Access
CCK	Complementary Code Keying
CoS	Class of Service
CRC	Cyclic Redundancy Check
cRTP	compressed Real-time Transport Protocol
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
CTS	Clear to Send
CW	Contention Window
DBPSK	Differential Binary Phase Shift Keying
DBSA	Development Bank of South Africa
DCF	Distribution Coordination Function
DIFS	DCF Inter Frame Spacing

DS	Distribution System
DSSS	Direct Sequence Spread Spectrum
DQPSK	Differential Quadrature Phase Shift Keying
ESS	Extended Service Set
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission (USA)
FCS	Frame Check Sequence
FHSS	Frequency Hopping Spread Spectrum
FIFO	First In First Out
FTP	File Transfer Protocol
IBSS	Independent Basic Service Set
ICT	Information and Communication Technologies
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSec	Internet Protocol security
ISA	Integrated Services Architecture
ISM	Industry, Scientific, and Medical
ISO	International Organization for Standardization
ISRDP	Integrated Sustainable Rural Development Programme

Kbps	Kilobits per second
LAN	Local Area Network
LLC	Logical Link Control
MAC	Media Access Control
MAN	Metropolitan Area Network
MANET	Mobile Ad-hoc Networks
Mbps	Megabits per second
MSDU	MAC Service Data Unit
NIC	Network Interface Card
NOS	Network Operating System
OFDM	Orthogonal Frequency Division Multiplexing
PCF	Point Coordination Function
PLCP	Physical Layer Convergence Procedure
PMD	Physical Medium Dependent
PPDU	PLCP Protocol Data Unit
PSTN	Public Switch Telephone Network
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency

RTP	Real-time Transport Protocol
RTS	Request to Send
SIFS	Short Inter Frame Spacing
SNMP	Simple Network Management Protocol
SS	Subscriber Station
STA	Station
TCP	Transport Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
ToS	Type of Service
UDP	User Datagram Protocol
VAD	Voice Activation Detection
VoIP	Voice over Internet Protocol
WEP	Wired Equivalent Privacy
WiFi	Wireless Fidelity
WiMax	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network
WPA	Wireless Protected Access
WSDP	Water Services Development Plan
4G	Fourth Generation Networks

CHAPTER ONE: INTRODUCTION

1.1 The Need for Rural Telecommunications

South Africa has very large disparity in rural versus urban and city telecommunication provisioning. The major cities enjoy the latest in telecommunication technology, while many regions of the country (especially the rural areas) have almost no service at all. To bridge this digital divide appears to be a daunting task. The abovementioned scenario and the Telecoms policy has become a major political issue in South Africa. The Telecoms policy is a government policy on the future development of telecommunications in South Africa. It balances the provision of basic universal service to disadvantaged rural and urban communities with the delivery of high-level services capable of meeting the needs of a growing South African economy. According to [1] the Integrated Sustainable Rural Development Programme (ISRDP), states the president in 2000, identified 13 District Municipalities (nodes) in South Africa as the priority focus areas for government to channel their expenditure in order to bring significant change to these poverty stricken areas. In order to provide universal access to disadvantaged and rural areas the South African Government has launched a drive to provide telecentres to communities and Internet access to schools. The telecentres are normally centrally located with respect to clusters of schools and other community services. It is important to recognize the potential of the Internet as an information source. The Internet is the tool to accessing wealth and knowledge and to empowering our nation.

Due to the rapid development of telecommunication technology, the benefits of the wireless industry in everyday life cannot go unnoticed. One of its greatest impacts is on the life in rural areas. Open rural and sparsely populated areas that lack basic communication facilities due to its harsh geographic and climatic environments can now overcome these barriers with the aid of wireless technology. The low cost of installing and maintaining a wireless communication system with ease in these problem areas makes it an attractive alternative for telecommunication carriers and service providers. The rapid increase in the use of cellular phones in rural areas substantiates the point that there is a potential telecommunication market in rural areas that is yet to be tapped into.

South Africa has an alarmingly low teledensity in some rural areas – less than 1% in certain rural areas [2] - making it difficult to connect schools that have computers to the Internet, even in the simple manner of a dial-up link. The benefits of schools having access to the Internet are immense. Students will no longer be at a disadvantage due to the limitations of their location, but will be able to interact with academics worldwide. This will open a whole new window of opportunities for the students.

The costs incurred in laying new copper or fibre-optic cables and building exchanges is high compared to potential revenue and thus is unattractive to telecommunication companies. For these reasons, wireless connectivity would be a great asset and service in South Africa.

The aim of this research is to provide a low cost, sustainable method of providing Internet access to these telecentres in a rural area. The rural area selected is in Zululand which is one of the thirteen nodes mentioned by the president in [1]. Many of the rural areas in Zululand do not have a wire-line infrastructure to provide for a dial-up link to the Internet. Hence the research proposes to implement a fixed wireless LAN (WiFi), and determine ways of extending the WiFi range to get optimal coverage. This WiFi LAN (hotspot) will be connected to the Internet via a WiMax backhaul. This extended range in a rural village or town will give more people access, to enjoy the benefits of the Internet. The use of VoIP, email, and web-browsing will greatly benefit the community in various ways. For example: VoIP and email can help people communicate with family, friends and associates in far away places; web-browsing can help students with their studies and research, as well as doctors and nurses in clinics, and many more.

Voice over Internet Protocol (VoIP) is one of the most widely used Internet applications and is still gaining popularity. Apart from the fact that VoIP drastically improves bandwidth efficiency, the combination of VoIP with other data applications provides new services such as video-conferencing, white boarding and so on. However, such a revolutionary application does have its flaws namely; a low VoIP capacity in WLAN and the performance degradation of VoIP in the presence of coexisting traffic from other applications. According to the theoretical figures, the 802.11 protocols will be able to support quite a substantial amount of VoIP sessions however the practical implementation shows a dramatically reduced amount of calls that can be sustained. This is due to various protocol overheads. Thus the key aspect of the research is to investigate the performance of real-time traffic such as voice and video in 802.11 solutions during heavy load. Voice and video packet end-to-end delay increases exponentially as the load on the network is increased. Voice capacity and quality is observed as the bandwidth is shared with data applications. The optimum solution as defined in chapter four is selected for the rural telecentre scenario.

The development of the Information Society depends on the accessibility of Information and Communication Technologies (ICTs) in business, education and government. Information technology will greatly benefit these sectors making them more efficient, transparent and providing the public, private sector and citizens with the ability to interact online. ICTs in business are used for electronic transactions and communications. ICT increases productivity of businesses and thereby boosts economic development. In addition, workers develop ICT skills which can be used in other areas apart from the workplace. In the education sector, students will become computer literate from an early age. E-forms used by government offices reduce the need to travel and eliminates the hassles of waiting in long queues at government buildings. Thus a more cost effective and time-saving alternative is provided.

1.2 The Region Selected For Study: Zululand

Zululand District is situated in the north-eastern part of KwaZulu-Natal. It has an area of 15 307 Km^2 and a population of about 954 020 people living in 866 dispersed settlements and six urban areas. Most of the rural settlements are small, making service delivery costly. About half the area falls under the jurisdiction of Traditional Authorities, the remainder being privately owned commercial farms, protected areas, or privately owned land in towns. The district experiences high levels of poverty. It needs investment in basic infrastructure and economic activities, which would create employment and, for this reason, has been selected as one of the focus areas for ISRDP. The area of jurisdiction of the new Zululand District Municipality is divided into five municipal areas. These are [3]:

eDumbe (Paulpietersburg) - KZ 261
uPhongolo (uPhongola) - KZ 262
Abaqulusi (Vryheid) - KZ 263
Nongoma - KZ 265
Ulundi - KZ 266

From the five municipal areas, the study will be limited to one of the most disadvantaged areas; namely, Nongoma.

1.2.1 The Nongoma Municipality

The geographical area of the Nongoma municipality is 2184 Km^2. Nongoma consists of 19 wards with 38 councilors.

Nongoma enjoys more sunshine hours and fewer rainy days than many other areas of its kind in South Africa. The area boasts a moderate climate all year round with temperatures ranging from 16 degrees Celsius in winter to 23 degrees Celsius in summer. The area has a mean annual precipitation of 750mm. Nongoma local municipality has a population in excess of 230 000 (WSDP, 2001), making it Zululand's second largest municipality in terms of population and the second largest in terms of area [3]. The rural communities use ward 19 of Nongoma as their primary service centre.

As far as social infrastructure and facilities are concerned, the Nongoma local municipality has one hospital and twelve clinics/health facilities. There is one fully-fledged police station, one social development office facility, one fully-fledged post office, one library and one sports facility. Nongoma municipality has about 25 community primary schools, and 13 high schools. There is also one technical college active in the area [4].

The level of education in the Nongoma municipality is low. Only 33% of the population in Nongoma has primary-level education and a mere 5.3% boasts of Grade 12 education [4]. The low level of education is reflected in the low level of development in the district. Education facilities are needed for both children and adults to further the chances for economic and social development in the rural and tribal areas. A further constraint is the lack of much needed facilities such as laboratories, libraries and sports fields. This limits the development of the skills of the children and their chances of bettering themselves in the future.

The telephone coverage in Nongoma is poor. Almost 56% of communities in Nongoma have no access to telephone network. Only 1% of households in Nongoma have telephones in dwelling. Reliable telecommunications will provide access to information, employment opportunities, education and health facilities, which in turn impact on productivity and social networks and which influence the ability of individuals and households to participate productively in the economic sphere.

The key statistical indicators for Nongoma can be found in Table 1.1 below.

Table 1.1: Key statistics of Nongoma

Item	Indicator	Totals
1.	Area	2184km^2
2.	Population	230 672
3.	Number of settlements	363
4.	Number of towns and small urban settlements	1
5.	% Rural population	98.34%
6.	% Urban population	1.66%
7.	% male population	44.61%
8.	% Female population	55.39%
9.	% Age group: 0 – 4 (years)	14.12%
10.	% Age group: 5 – 64 (years)	80.11%
11.	% Age group: 65+ (years)	4.35%
12.	% age unknown	1.42%

The research targets the group of people between the ages of 5 to 64 years. This estimates that 80.11% of the Nongoma population are potential telecommunication subscribers.

According to [4] a report prepared by the Development Bank of Southern Africa (DBSA) in 2000 provides the most useful statistics for estimating the future growth rate of Nongoma's population. It takes into account the high impact of HIV/AIDS and gives growth rates for 5-year intervals disaggregated by provinces. The KwaZulu-Natal figures have been used as the basis for projecting the population of Nongoma. Table 1.2 sets out the growth assumptions from the DBSA report and applies these to the WSDP population for 2001 for Nongoma as a whole [4].

Table 1.2: Population Growth Rate Projections for Nongoma Local Municipality, 2001 – 2021 [4]

Period	% Annual growth rate (based on DBSA report)	Growth factor	Population at start of period (2001 based on WSDP data)
2001-2005	1.44	1.0741	230 672
2006-2010	0.96	1.04893	247 765
2011-2015	0.58	1.02934	259 888
2016-2020	0.08	1.00401	267 513
2021			268 586

*High HIV/AIDS Impact Sept. 2000

1.3 Scope of the Study and Dissertation Structure

This research presents a solution to the much talked about "Digital Divide". The digital divide emerged out of the uneven spread of wealth & development in the country as a result of the apartheid aftermath. Rural and disadvantaged areas in SA have little to almost no telecommunication facilities. Infrastructure costs for implementation in certain areas are extremely high and in other cases the terrain makes it inaccessible to construct telephone networks. The emerging market of wireless technology with applications such as VoIP sheds a beam of light on this barren land. This research thus proposes to use this technology to provide a much needed service to the masses living in rural SA.

The benefits of providing Internet access and VoIP services to the population has been emphasised in this chapter. The case study selected a rural disadvantaged area north-east of Kwa-Zulu Natal called Nongoma. A profile and demographic analyses of the selected area has been done in this chapter. A profile of the solution is to implement WiFi WLANs in public locations to which a large percentage of the population will have access to. The benefits and features of WiMax make it the perfect candidate to do the backhauling of the WiFi WLANs to an Internet source. The solution is simulated to evaluate its feasibility. Apart from concluding that this is a feasible approach, the research calculates the maximum amount of VoIP calls that can be sustained in a network using the 802.11g protocol with CTS-to-self protection. It also proposes the use of mesh technology to extend the coverage in rural areas and evaluates a Star Mesh Network (SMN) model for the rural case.

The outline of the dissertation is:

Chapter Two provides the traffic estimates used in the simulation work of chapter four. The traffic dimensioning is done by using the Nongoma statistics to predict the traffic generated by the population. Traffic is calculated for each of the three telecentres provided in a hospital, school and a simple public telecentre. The network is dimensioned for a period of eight years, using typical PSTN and Internet trends.

Chapter Three provides the technical background for an understanding of the operation of an 802.11 WLAN. Properties of the different protocols highlight the applications that it is best suited for. The importance and operation of VoIP in a data network is reported. A discussion about the suitability of WiMax for use as a backhaul is given.

Chapter Four gives a brief description of the OPNET modeler 11.5 simulation package. A comparison of the 802.11b to 802.11g protocol is done via simulation. The 802.11g MAC layer enables the protocol to maintain low delays and high throughput during high loads on the network unlike 802.11b. The 802.11g protocol is used in the three telecentres and evaluated with the estimated traffic on the network over the dimensioned period. The hospital and school telecentres are heavily loaded to evaluate the threshold point before the networks fail. Research was done to test, for a telecentre only offering VoIP services, the maximum amount of simultaneous VoIP sessions that a network can sustain. The simulations mirrored the analytic results.

Chapter Five proposes the implementation of a mesh topology network to further extend the coverage to create WiFi hotspots or even hotzones. The advantages of a mesh topology are highlighted. Investigations of a proposed Star Mesh Network by Zhang and Wolff are done to evaluate its applicability to a South African rural scenario.

Chapter Six concludes the research by highlighting significant contributions and presents ideas for future work.

1.4 Original Contribution and Publications

Chapter Two -A rural South African case is identified. Traffic dimensioning is done for the specific case.

Chapter Four -Evaluation of the 802.11b and 802.11g protocol. QoS for the rural case is evaluated by examining the delay parameters. Voice sustainability investigation is done.

Chapter Five -Mesh studies are incorporated in the rural case.

Publications:

A paper titled *802.11 Mesh Mode Provides Rural Coverage at Low Cost* by K Naidoo and R Sewsunker was accepted for publication at IEEE Africon 2007 held in Windhoek, Namibia in September 2007.

A paper titled *Extending WiFi Access for Rural Reach* by K Naidoo and R Sewsunker was accepted for publication at the Southern African Telecommunications Networks and Applications Conference (SATNAC) held in Mauritius in September 2007.

CHAPTER TWO: THE RURAL CASE STUDY

2.1 Outline of Solution

The aim of the study is to explore a low cost, sustainable method of providing voice plus Internet access to telecentres in rural areas. The research is based on service provision using fixed wireless LANs (WiFi) in hotspot or hotzone (overlapping WLAN) configuration based on the service requirement. The area of interest is limited to Ward 19 of Nongoma. This ward hosts the town of Nongoma. There is only one hospital situated in the Nongoma district, namely the Benedictine Hospital in Nongoma town. In addition to the normal services rendered by the hospital, it also acts as a clinic and provides related service to surrounding communities. The town has basic municipal infrastructure such as electricity.

The intended plan to uplift this developing area is to create three wireless (WiFi) local area networks (LANs) in a school, hospital and public telecentre. WiMAX technology is a favourable option that should be considered for the backhauling of these telecentres to the Internet. These three telecentres will be connected via a WiMax backhaul to the Internet. The range of a typical WiFi LAN will be extended to create WiFi hotspots.

Figure 2.1 Topographical map of Nongoma [Google Earth]

The research forecasts the telecommunication growth of Ward 19 of the Nongoma region in KwaZulu-Natal over a period of 8 years based on current traffic estimates and these values give a starting point for dimensioning the network. The basis for these estimates are outlined below.
The projected population growth for the next 8 years of the Nongoma region was predicted using the relevant information from Table 1.2. The current and forecasted population for the region is shown in Table 2.1.

Table 2.1: Projected population growth for Nongoma

Year	Projected Population
2006	237 362
2007	240 780
2008	244 247
2009	247 765
2010	250 144
2011	252 545
2012	254 969
2013	257 417
2014	259 888

The population for ward 19 in Nongoma is 3847 with an area of 21.84km^2. We take into consideration the point that since ward 19 host the town and only hospital for Nongoma it will be frequented by the population from surrounding areas. Hence we shall dimension the network to cater for these numbers.

Table 2.2: Nongoma Population Distribution (2001) [4]

Urban	Rural Village	Rural Scattered	Scattered	Rural Farms	Total
3 842	183 525	43 305	0	0	230 672
1.66%	79.56%	18.78%	0.00%	0.00%	100.00%

Using the data from Table 2.2, the following estimates were made. Approximately 6% of the population that reside in rural villages and 1.5% of the population living in rural scattered areas are located in the surrounding areas of Ward 19. 1.66% of the population resides in an urban area which is Ward 19. These approximations give us a total of 15 951 people residing in Ward 19 and surrounding areas which is approximately 6.72% of the Nongoma population.

2.2 Traffic Projections

2.2.1 VoIP Traffic Predictions

The access network is to be dimensioned for the eight years starting 2006 and ending 2014. VoIP bandwidth calculations were done using the following methods and assumptions. In order to optimize the use of bandwidth the G729 codec was used for voice compression. The layer 2 protocol provides a unified framework for securing all wired and wireless connections using strong encryption and authentication. Voice Activation Detection (VAD) is a software application that allows a data network carrying voice traffic over the Internet to detect the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network. A VoIP packet is encapsulated by IP/UDP/RTP headers which is approximately 40 bytes. 40 bytes is a relatively large amount of overhead for the typical 20 byte VoIP payload. Thus compressed RTP reduces this overhead to 2 bytes for cases where UDP without checksum is being sent and to 4 bytes with checksum. The layer 2 header, cRTP and Voice Activation Detection (VAD) have also been accounted for by the following formulae:

(# calls/month X average holding time)/unit of time = # Erlangs (2.1)

Conversion from Erlangs to bits per second (bps):

Erlangs X 64Kbps (voice PCM) = # bits per second (2.2)

Compression:

bits per second X 0.125 (G729 codec) X 3 (CRTP & Layer2) X 0.6 (VAD) = # bits/sec (2.3)

It is necessary to take into account the population of the surrounding area of each telecentre to calculate the number of possible subscribers for each telecentre. This calculation was done using the following formula.

Number of subscribers = $\rho \, P \, \gamma \, \sigma \, \alpha$ (2.4)

Where:

P represents the population in Nongoma. (predicted from the 2001 census to be 237 362.)

ρ represents the percentage of the population that resides in Ward 19 and surrounding areas. (An estimated 6.72%)

γ represents the percentage of the population that will have access to the telecentre.

σ represents the percentage of the population that is active. (i.e. excluding children below 4 years, and adults over 65)

α represents the oversubscription ratio. (A typical oversubscription ratio of 1:6 is used.)

VoIP is a delay sensitive application which demands a lower oversubscription ratio as compared to data. The yearly percentage increases is predicted using previous PSTN growth trends in South Africa [6].

2.2.1.1 Simple Public Telecentre

For the Simple telecentre it is assumed each subscriber makes on average 10 calls/month with an average holding time of 3 minutes. 20% of the population of ward 19 has access to the simple telecentre of which 80% is active. Using the above formula, the number of subscribers is calculated:

$$\begin{aligned} \text{Number of subscribers} &= 0.0672 * 237362 * 0.2 * 0.8 * 0.167 \\ &= 425.35 \\ &\approx 425 \end{aligned}$$

425 subscribers will frequent this telecentre making on average 10 calls per month each resulting in a total of 4250 calls per month.(NB: there are 4 phones, so each phone makes 1062 calls per month). It can be said that each phone makes approximately 36 calls per day which is far below the threshold to eliminate long queues. The threshold for the number of calls a single telephone can make in one day is estimated via the following assumptions; if a simple telecentre is operating for 8 hours in a day, this implies that a single telephone will be active for 480 minutes. Dividing this value by the average holding time of three minutes plus a one minute interval between calls results in each telephone being able to sustain a maximum of 120 calls per day. A sample calculation for the VoIP bandwidth required for 2006 is as follows:

From equation (2.1): $\left(\dfrac{4250*3}{30*8*60}\right) = 0.8854E$

From equation (2.2), conversion: $0.8854*64Kbps = 56666.67bps$

From equation (2.3), compression: $56666.67*0.125*3*0.6 = 12750bps$

Hence the results for the consequent years follow in Table 2.3 below. The basis for the traffic assumptions have been provided earlier in the chapter.

Table 2.3: Simple Telecenter VoIP BW Requirements

Year	% Increase	Calls/month	Reqd. Bits/sec
2006		4250	12750
2007	13%	4803	14408
2008	20%	5763	17289
2009	23%	7088	21266
2010	21%	8577	25731
2011	20%	10292	30878
2012	15%	11836	35509
2013	16%	13730	41191
2014	14%	15652	46957

2.2.1.2 Hospital Telecentre

For the hospital telecentre it is assumed that 50% of the population of ward 19 has access of which 80% is active.

$$= 0.0672 * 237362 * 0.5 * 0.8 * 0.167$$

Number of subscribers $= 1065.51$

$$\approx 1066$$

It is assumed that the calculated 1066 subscribers will each make on average 8 calls per month with an average holding time of 3 minutes. Total number of calls made per month is a calculated 8528. Taking into account the number of phones available i.e. 6, each phone makes 1421 calls per month. Each phone makes 47 calls per day which is acceptable for short queues. The threshold for the hospital is calculated to sustain 180 calls per day per telephone (considering a 12 hour day). A sample calculation for the VoIP bandwidth required for the hospital telecentre in 2006 is as follows:

From equation (2.1): $\left(\dfrac{8528*3}{30*12*60}\right) = 1.1844E$

From equation (2.2), conversion: $1.1844 * 64Kbps = 75804.44bps$

From equation (2.3), compression: $75804.44 * 0.125 * 3 * 0.6 = 17056bps$

Hence the results for the consequent years follow in Table 2.4 below.

Table 2.4: Hospital Telecentre VoIP BW Requirements

Year	% Increase	Calls/month	Reqd. Bits/sec
2006		8528	17056
2007	13%	9636	19273
2008	20%	11564	23128
2009	23%	14223	28447
2010	21%	17210	34421
2011	20%	20652	41306
2012	15%	23750.	47501
2013	16%	27551	55102
2014	14%	31408	62816

2.2.1.3 School Telecentre

The school telecentre has γ=15% and σ=80%. Thus we assume that 320 subscribers each make on average 8 calls per month with an average holding time of 3 minutes. A total of 2560 calls per month are calculated resulting in each telephone making 21 calls per day. Threshold for the school given that the school telecentre is operational for 12 hours of a day, is 180 calls per day per telephone. A sample calculation for the VoIP bandwidth required for the school telecentre in 2006 is as follows:

From equation (2.1): $\left(\dfrac{2560*3}{30*12*60}\right) = 0.3555E$

From equation (2.2), conversion: $0.3555*64Kbps = 22755.56bps$

From equation (2.3), compression: $22755.56*0.125*3*0.6 = 5120bps$

Hence the results for the consequent years follow in Table 2.5 below.

Table 2.5: School Telecentre BW Requirements

Year	% Increase	Calls/month	Reqd. Bits/sec
2006		2560	5120
2007	13%	2893	5786
2008	20%	3471	6943
2009	23%	4270	8540
2010	21%	5166	10333
2011	20%	6200	12399
2012	15%	7130	14259
2013	16%	8270	16541
2014	14%	9428	18857

The growth trend for the VoIP traffic over a period of 8 years is shown in Figure 2.2.

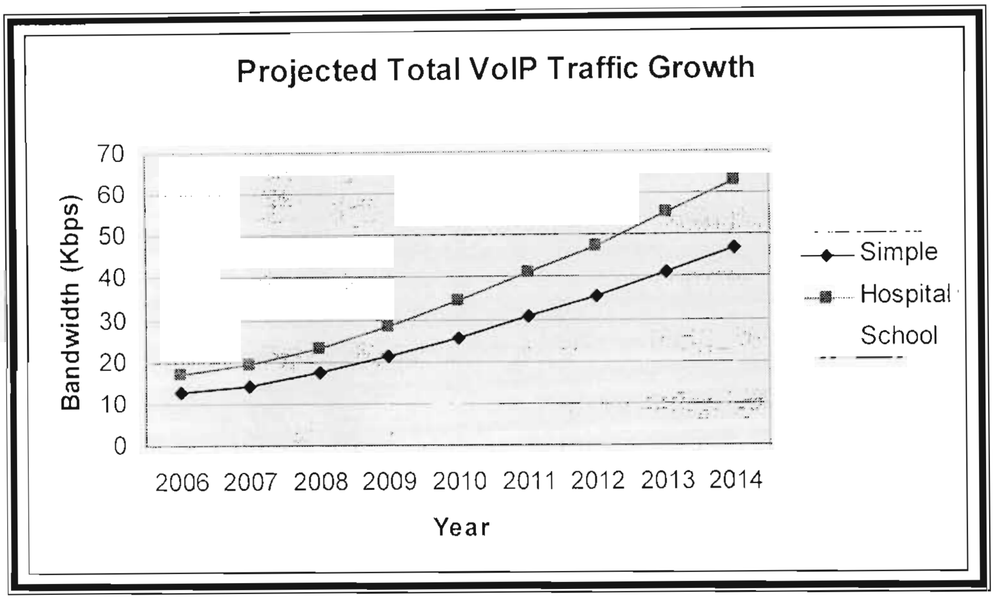

FIGURE 2.2: Graph of projected VoIP traffic

2.2.2 Data Traffic Predictions

All three telecentres will offer different applications according to user requirements. The applications offered at the telecenters may change with time, as user needs change. The annual increases in traffic for each of the applications, and hence total bandwidth required, are dependent on a number of complex interacting factors. These include technological, social as well as economic factors [7]. Technologically, web pages and web compilers are rapidly becoming more complicated with added graphics and increased efficiency, leading to greater volumes 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. The data and email percentage increase is estimated from previous trend patterns [6]. The values for 2006 are a good starting point for an emerging rural market. The minimum typical transaction volume for a formatted email text is 2 Kbytes and for file transfer it is about 50 Kbytes [8]. The 2006 telemedicine bandwidth requirement is the minimum bandwidth required to meet diagnostic needs which is 1.5Mbps [8]. Videoconferencing will be used for e-learning applications and for a South African rural scenario a typical 12 minute session will require a data transfer rate of 1.6Mbps [8]. The minimum average web-browsing bandwidth is 500kbps [9]. Table 2.7 below shows the percentage increase for the different applications over the 8 year time period.

Table 2.6: Yearly Percentage Increase of Data traffic for one station

Year	Email %incr.	Email File size (Kbps)	FTP & HTTP %incr.	FTP File size (Kbps)	HTTP (Kbps)	Telemed & e-learn. %incr.	Telemed (Kbps)	e-learn. (Kbps)
2006		16.0		400	500		1500	1600
2007	5	16.8	10	440	550	5	1575	1680
2008	15	19.3	30	572	715	10	1733	1848
2009	25	24.2	50	858	1073	16	2010	2144
2010	35	32.6	70	1459	1823	20	2412	2572
2011	32.5	43.2	65	2407	3008	26	3039	3241
2012	25	54.0	50	3610	4513	26	3829	4084
2013	20	64.8	40	5054	6318	30	4977	5309
2014	18	79.1	30	6570	8213	30	6471	6902

Real-time traffic is also known as a constant bit-rate application and utilises the UDP protocol. The requirements for real-time applications per station are illustrated in Figure 2.3.

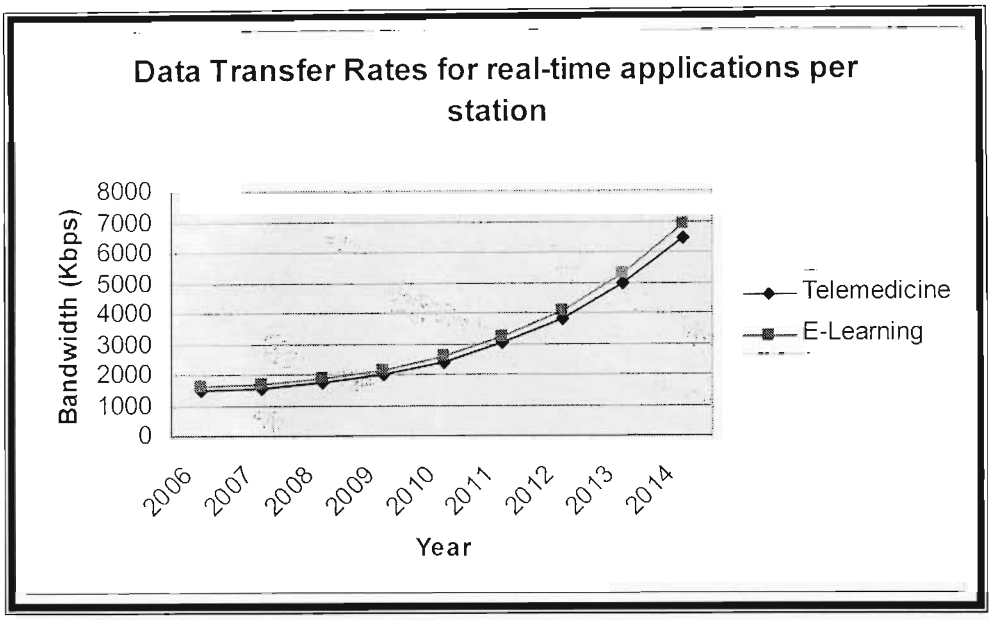

Figure 2.3: Bandwidth Requirements for Real-Time Applications for one station

To satisfy user requirements, the acceptable average tolerance of users to delays or download times is 10 seconds with a maximum of 15 seconds [8]. A 10 second delay tolerance time will be used to dimension the network for non-real time applications, resulting in the following bandwidth requirements. For example in 2006 the file transfer requires 400Kbps from Table 2.6, taking into account the 10s delay user tolerance time, FTP will only require 400Kbps/10s = 40Kbps. Similarly the bandwidth requirements for Email and HTTP are calculated.

Table 2.7: Bandwidth Requirements for Non-Real Time Applications based on 10s user tolerance

Year	FTP (Kbps)	Email (Kbps)	HTTP (Kbps)
2006	40	1.6	50.0
2007	44	1.7	55.0
2008	57	1.9	71.5
2009	86	2.4	107.3
2010	146	3.3	182.3
2011	241	4.3	300.8
2012	361	5.4	451.3
2013	505	6.5	631.8
2014	657	7.9	821.3

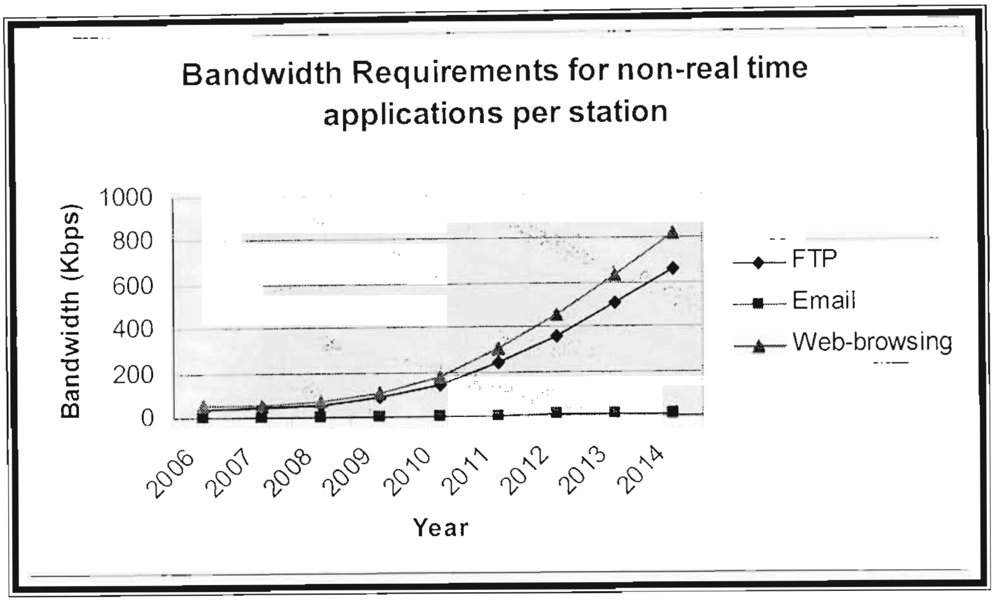

Figure 2.4: Graph of projected increase of data transfer rates for non-real time applications per station

The total bandwidth required for data applications can be calculated using the following formula:

Bandwidth = Number of Users * Data Transfer Rate (non-real-time application) * Oversubscription Ratio + Data Transfer Rate (real-time application)

(2.5)

Where:

Number of Users = number of PC's at a telecenter. (It is assumed that there will be one user per PC at any one time). These values can be found in Table 2.1

Oversubscription ratio = Overbooking factor. (Assumes that not everyone is online at the same time therefore 'peak' bandwidth has not been allocated to every PC/user.)

Real Time Applications = Videoconferencing, e-Learning, Telemedicine, streaming audio/video, VoIP.

Non-Real Time Applications = email, web-browsing, documents (downloads), limited audio/video.

If the above formula is assumed to be applied to the proposed network, the following results can be obtained for typical telecentre bandwidth requirements. The most bandwidth intensive non-real time application is used as the data transfer rate in all scenarios when calculating the total data bandwidth requirements. Literature indicates that a typical oversubscription ratio of 0.4 is reasonable. Using the above formula and taking into account the above mentioned considerations the following data bandwidth estimates were calculated for each telecentre. For example for all three telecentres web-browsing is selected as the most bandwidth intensive non-real time application since it requires 50Kbps as compared to FTP and Email. A sample calculation for 2006 is done for each telecentre.

Simple: Data BW = $6 * 50Kbps * 0.4 = 120Kbps$

School: Data BW = $(8 * 50Kbps * 0.4) + 1600Kbps = 1760Kbps$

Hospital: Data BW = $(8 * 50Kbps * 0.4) + 1500Kbps = 1660Kbps$

Table 2.8 Data Bandwidth Requirements for each telecentre

Year	Simple telecentre Data BW (Kbps)	School telecentre Data BW (Kbps)	Hospital telecentre Data BW (Kbps)
2006	120.0	1760.0	1660.0
2007	132.0	1856.0	1751.0
2008	171.6	2076.8	1961.3
2009	257.4	2486.9	2352.9
2010	437.6	3155.9	2995.1
2011	722.0	4203.9	4001.3
2012	1083.0	5528.0	5272.7
2013	1516.2	7330.8	6999.0
2014	1971.1	9530.0	9098.6

The values in Table 2.8 are illustrated in Figure 2.5 below.

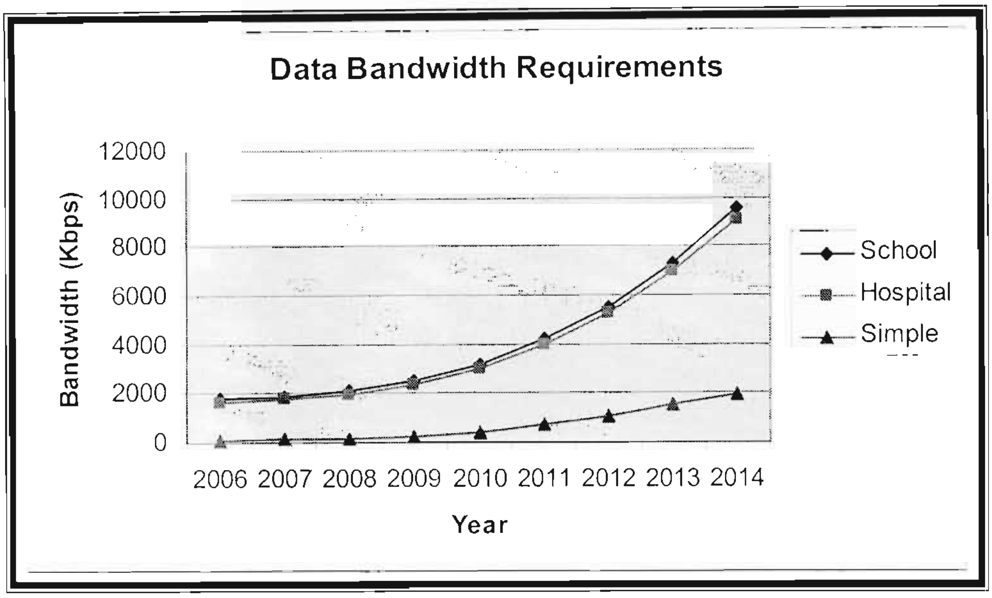

Figure 2.5: Total Data Bandwidth Requirements for each telecentre

Finally the total bandwidth required for the three telecentres over the 8 year time period was calculated by adding the Data and VoIP bandwidth requirements as shown in Table 2.9, 2.10, and 2.11 for each telecentre.

Table 2.9: Total Bandwidth Requirements for the Simple Telecentre

Year	VoIP BW (Kbps)	Data BW (Kbps)	Total BW (Kbps)
2006	12.75	120.00	132.75
2007	14.41	132.00	146.41
2008	17.29	171.60	188.89
2009	21.27	257.40	278.67
2010	25.73	437.58	463.31
2011	30.88	722.01	752.88
2012	35.51	1083.01	1118.52
2013	41.19	1516.22	1557.41
2014	46.96	1971.08	2018.04

Table 2.10: Total Bandwidth Requirements for the School Telecentre

Year	VoIP BW (Kbps)	Data BW (Kbps)	Total BW (Kbps)
2006	5.12	1760.00	1765.12
2007	5.79	1856.00	1861.79
2008	6.94	2076.80	2083.74
2009	8.54	2486.88	2495.42
2010	10.33	3155.86	3166.19
2011	12.40	4203.92	4216.32
2012	14.26	5527.98	5542.24
2013	16.54	7330.78	7347.32
2014	18.86	9530.01	9548.87

Table 2.11: Total Bandwidth Requirements for the Hospital telecentre

Year	VoIP BW (Kbps)	Data BW (Kbps)	Total BW (Kbps)
2006	17.06	1660.00	1677.06
2007	19.27	1751.00	1770.27
2008	23.13	1961.30	1984.43
2009	28.45	2352.90	2381.35
2010	34.42	2995.08	3029.50
2011	41.31	4001.34	4042.65
2012	47.50	5272.73	5320.24
2013	55.10	6998.96	7054.06
2014	62.82	9098.64	9161.46

Hence the results for the total bandwidth required for each telecentre is illustrated in Figure: 2.6.

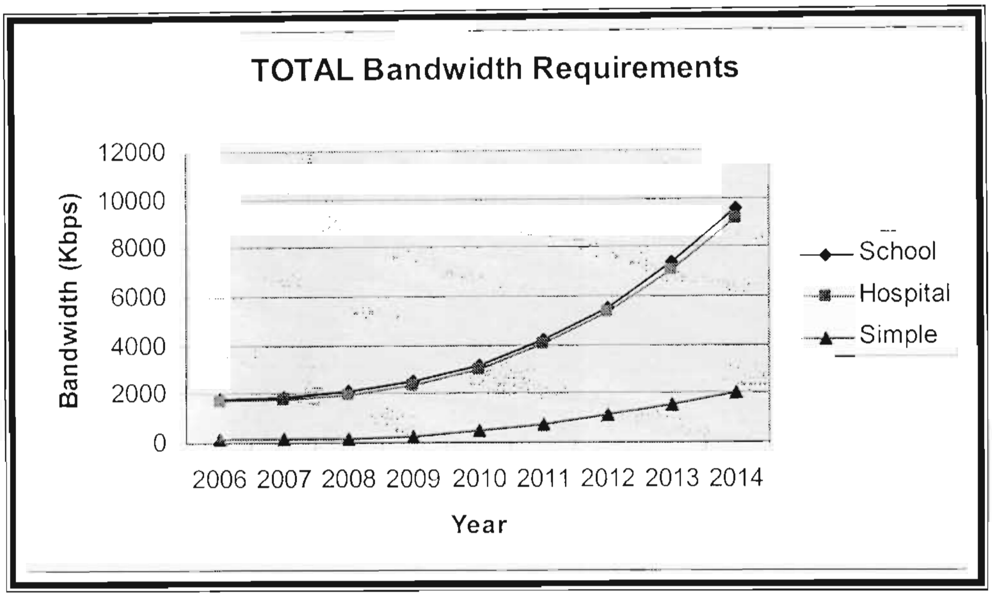

Figure 2.6: Total Bandwidth required for the telecentres

2.3 Telecentre Profiling

The telecentre profile is shown in Table 2.12 and explained below. Jensen and Esterhuysen [5] suggest that a telecentre with 5 computers is able to cater for a population of about 20000 people in rural SA. Thus for a similar population the research serves to eliminate long queues by providing a minimum of 6 to 8 computers in the telecentres. A similar estimation is used for the selection for the number of VoIP phones in each telecentre. The percentage of population that has access to the telecentre is estimated from the number of people that reside within 30 minutes walking distance from the telecentre. The percentage of the population that is active is estimated by excluding children under 4 years of age and adults over 65 years [4].

Table 2.12: Telecentre Profiles

Number	Telecentre	# of phones	# of pc's	%Pop. with access	%Pop. that is Active
1	Simple	4	6	20%	80%
2	Hospital	6	8	50%	80%
3	School	4	8	15%	80%

Each telecentre will offer different applications according to user requirements in that specific region. A Hospital telecentre will be situated at the only existing Hospital in the area at present, the Benedictine Hospital, located 15Km north of the base station. This telecentre will focus more on offering telemedicine applications and services to doctors, nurses and patients. An estimate of 50% of the population of Ward 19 will have access to this telecentre. This fixed wireless LAN will house 6 VoIP phones and 8 computers.

A School telecentre will be implemented at a Secondary School situated approximately 20Km away from the hospital and 15KM east of the base station. This telecentre will cater more for the needs of e-Learning and downloads. It is estimated that 15% of the Ward 19 population will have access to the school telecentre, majority of which will be students and teachers. The LAN will host 8 computers and 4 VoIP phones.

The third WiFi LAN will be a simple telecentre, which will cater for the needs of the community by providing the basic communications, namely VoIP phones, email, FTP and web-browsing. It will be situated approximately 20Km from the school and an estimated 15Km south of the base station. It is assumed that 20% of the population in Ward 19 will have access to this telecentre. This simple LAN will consist of 4 VoIP phones and 6 computers.

A telecentre will be implemented either as a hotspot or hotzone and can be connected to the outside world via a WiMax base station.

2.4 Summary

An outline of the solution for the rural case study was presented in this chapter. Using the demographic information of the selected rural area, estimates were made for the total bandwidth that will be required to support VoIP and data applications that are suited to the needs of the community. These traffic estimates will be used in Chapter four as the traffic that will be loaded onto the network in the simulation.

CHAPTER THREE: OVERVIEW OF WiFi / IEEE 802.11

3.1 Introduction / Background

WiFi is the abbreviated form of Wireless Fidelity. It is an IEEE standard that was ratified in 1997. The 802.11 protocol is a set of Wireless LAN standards developed by the working group 11 of the IEEE LAN/MAN standards committee [A]. The 802.11 family currently includes six over-the-air modulation techniques that all use the same protocol. 802.11b was the first widely accepted wireless networking standard, followed by 802.11a and 802.11g. Other standards in the family (c-f, h j, n) are service enhancement and extensions, or corrections to previous specifications. The 802.11e enhances the Quality of Service (QoS) performance of the standard, while 802.11i improves security and 802.11n gives faster data rates of 100Mbps. The 802.11s introduces the improved application of the standard in mesh mode, which is yet to be ratified in 2007. The 802.11b and 802.11g standards use the unlicensed 2.4GHz band. Operating in an unregulated frequency band, allows 802.11b and 802.11g to be susceptible to interference from appliances (e.g.: microwave ovens, cordless phones, Bluetooth devices, etc.) that use the same 2.4GHz band. The 802.11a standard uses the 5GHz band.

Raw data rates of 1Mbps and 2Mbps that are transmitted via infrared (IR) signals or in the Industrial Scientific Medical (ISM) frequency band at 2.4 GHz were specified in the original version of the standard IEEE 802.11. Currently IR has no actual implementations but remains a part of the standard.

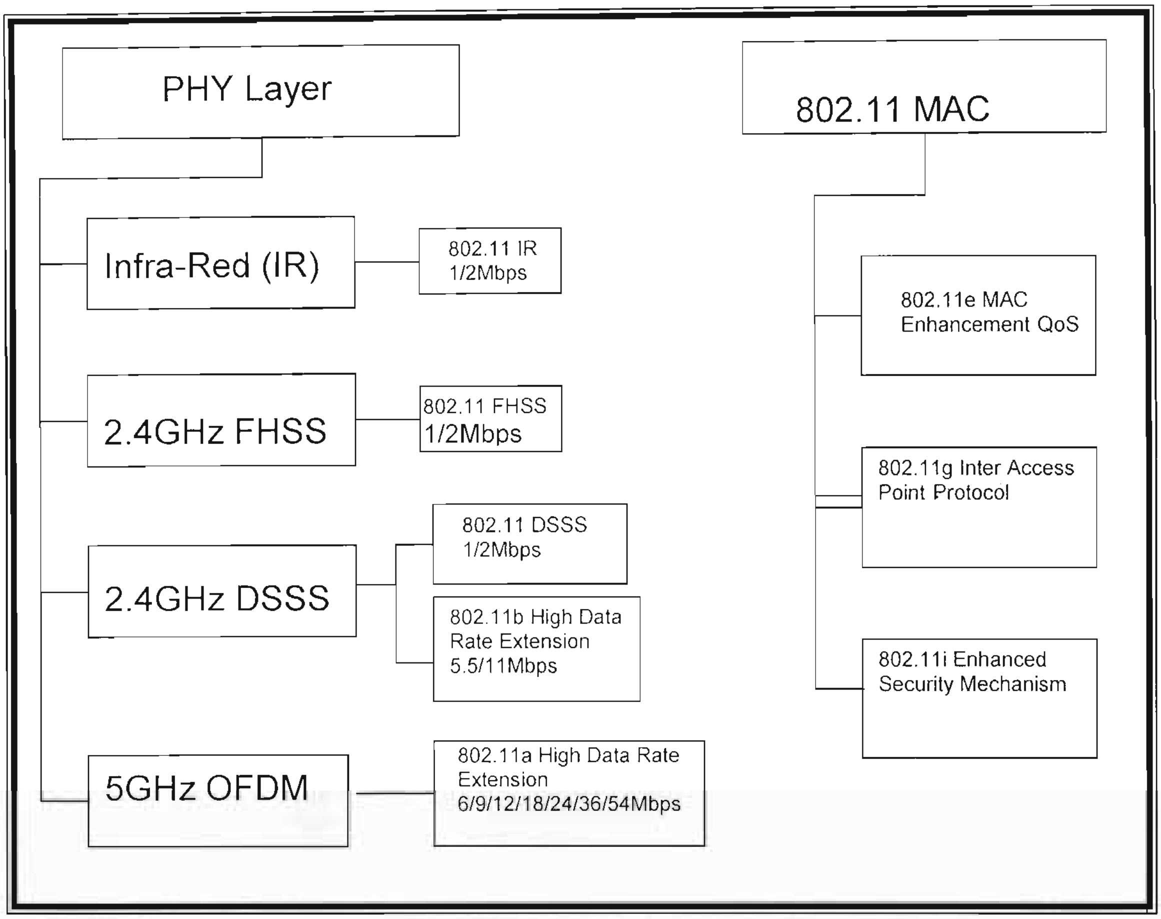

Figure 3.1: Snapshot of 802.11 PHY and MAC standard activities

The 802.11b standard uses Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol for medium access control. The standard includes a contention based and a polling based medium access protocol, called Distributed Coordination Function (DCF) and Point Coordination Function (PCF) respectively.

3.2 802.11 Architecture

The 802.11 architecture consists of two basic modes namely the Infrastructure mode and the Ad Hoc mode.

3.2.1 Infrastructure Mode

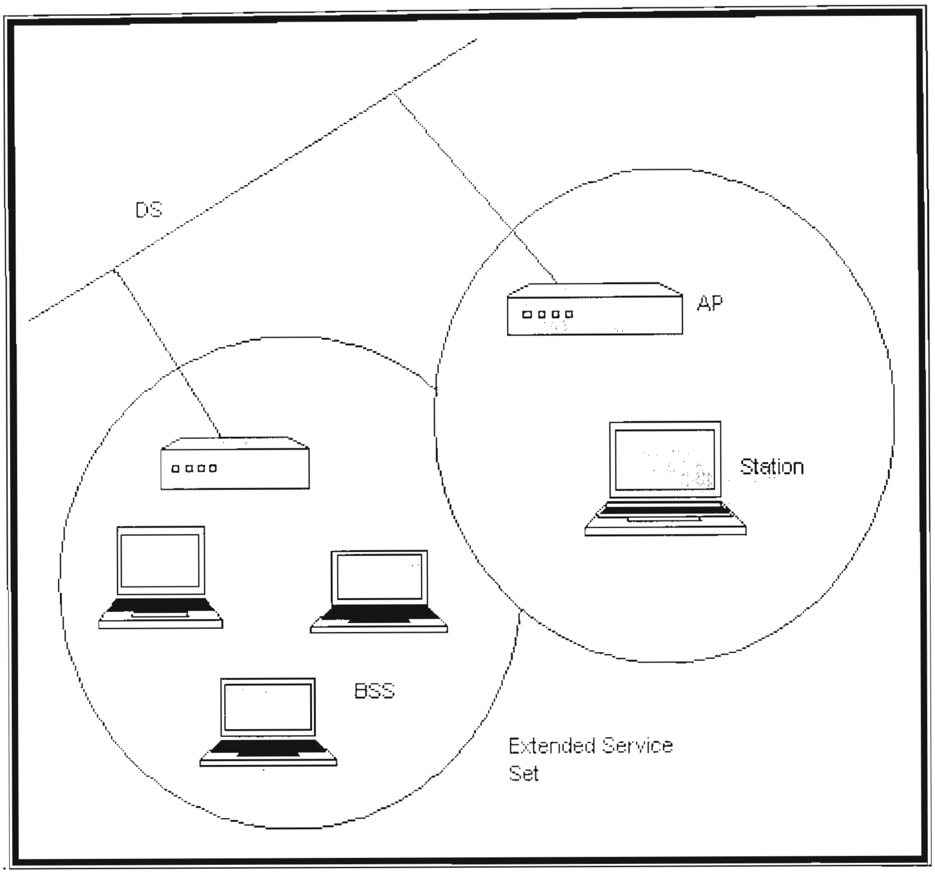

Figure 3.2: Infrastructure Mode

When two or more stations come together to communicate with each other, they form a Basic Service Set (BSS). Two or more BSS's are interconnected using a Distribution System (DS). The concept of DS increases network coverage. Each BSS becomes a component of an extended, larger network. Entry to the DS is accomplished with the use of Access Points (AP). Creating large and complex networks using BSS's and DS's leads us to the next level of hierarchy, the Extended Service Set (ESS). In the ESS, the entire network looks like an independent basic service set to the Logical Link Control layer (LLC). This means that stations within the ESS can communicate or even move between BSS's transparently to the LLC.

In order for 802.11 to be used with existing wired networks, a portal is used. A portal is the logical integration between wired LANs and 802.11. It also can serve as the access point to the DS. It thus functions as a bridge between wired and wireless LANs. The 802.11 protocols specify the services which the DS must support. These services are divided into two sections namely; Station Services (SS) and Distribution System Services (DSS).

There are five services provided by the DSS namely association, reassociation, disassociation, distribution and integration.

- A station must affiliate itself with the BSS infrastructure if it wants to use the LAN. This is done by *associating* itself with an access point. Associations are dynamic in nature because stations move, turn on or turn off. A station can only be associated with one AP. This ensures that the DS always knows where the station is. Association is when a station is either stationary or moving within its own BSS.
- Reassociation allows a station to switch from one access point to another when the station moves between BSS's within the same ESS.
- Disassociation is when the association between the station and access point is terminated. The station can move to another ESS but will have to reinitiate connections.
- Distribution is the transmitting of the data from the source via its access point through the DS to the intended destination via its access point.
- Integration is when the output AP is a portal.

Station services are authentication, deauthentication, privacy, and MAC Service Data Unit (MSDU) delivery [10].

- Before two stations in an IBSS or an AP of a BSS to a station can communicate with each other, the station needs to provide a network wide password, this is authentication. Open System Authentication provides anyone who attempts to authenticate with authentication, whereas Shared Key Authentication only provides users who possess a 'shared secret' with authentication. The Wired Equivalent Privacy (WEP) algorithm is used to implement the shared secret which is delivered to all stations ahead of time in some secure method.
- Deauthentication is the removal of a station's authentication by the station or access point. Deauthentication results in the station being automatically disassociated.
- 802.11 utilises the unlicensed frequency spectrum and the transmission of data over-the-air can be easily tapped into. In order to prevent other 802.11 users from eavesdropping on your LAN traffic an encryption algorithm is employed called Privacy. IEEE 802.11

specifies Wired Equivalent Privacy (WEP) as an optional algorithm to satisfy privacy. If WEP is not used then that traffic is not encrypted. Data is only encrypted after authentication.

- MSDU delivery ensures that the information in the MAC service data unit is delivered between the medium access control service access points.

3.2.2 Ad-Hoc Mode

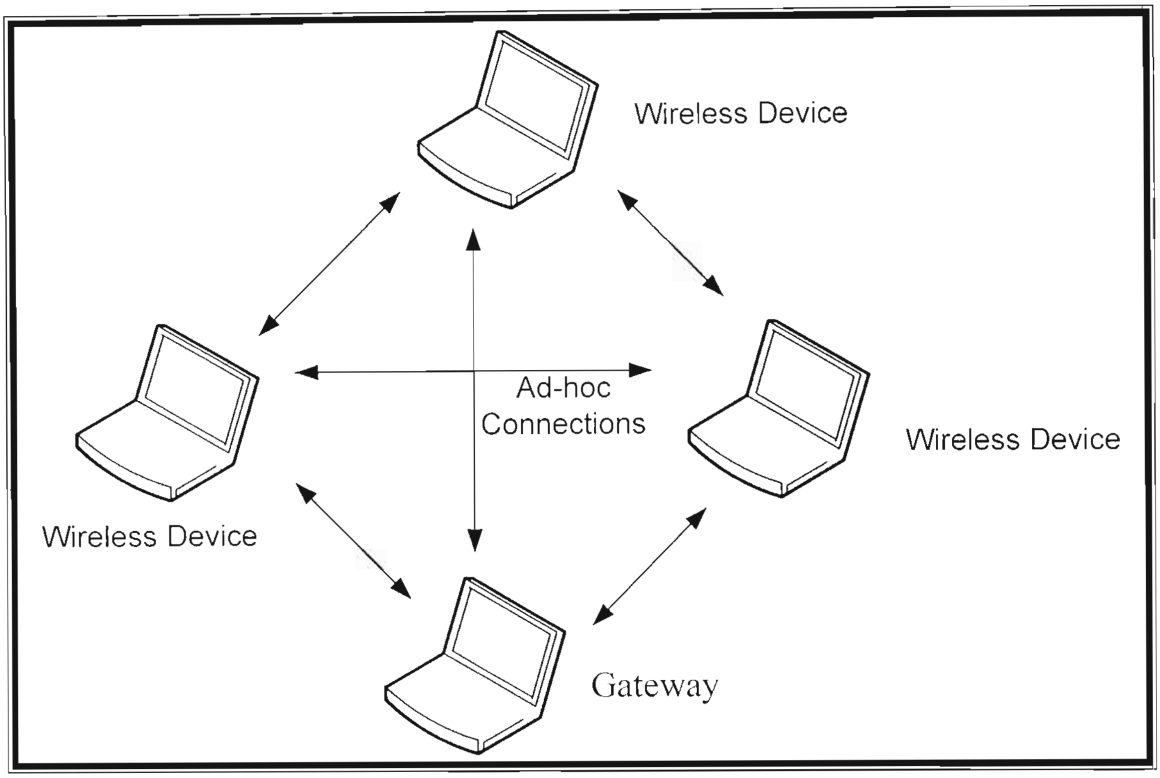

Figure 3.3: Ad-Hoc Mode

In an ad-hoc or peer-to-peer wireless network the computers are equipped with a Network Interface Card (NIC) which is used to communicate directly with each other in the same Basic Service Set (BSS) without the use of an access point. These computers can only share information amongst themselves and can not access the resources from the wired LAN. In order for these computers to gain access to the resources provided by the wired LAN, one of the computers has to act as a bridge to the wired LAN using special software. In peer-to-peer configuration one can communicate reasonably as long as the two computers are close enough to each other in order to avoid excessive Radio Frequency (RF) interference and signal loss.

3.3 Accessing the Media

The Open Systems Interconnection (OSI) networking suite is a seven layer model. In each layer, one or more entities have a specific function to perform. Each entity communicates directly with the layer immediately beneath it, and renders facilities available for use by the layer above it. Protocols are the links that allow an entity in one host to communicate with a corresponding entity at the same layer in a remote host. The OSI stack and the description and functions of each layer are thoroughly explained in [11]. This research focuses specifically on the Data-Link and physical layers for wireless LANs.

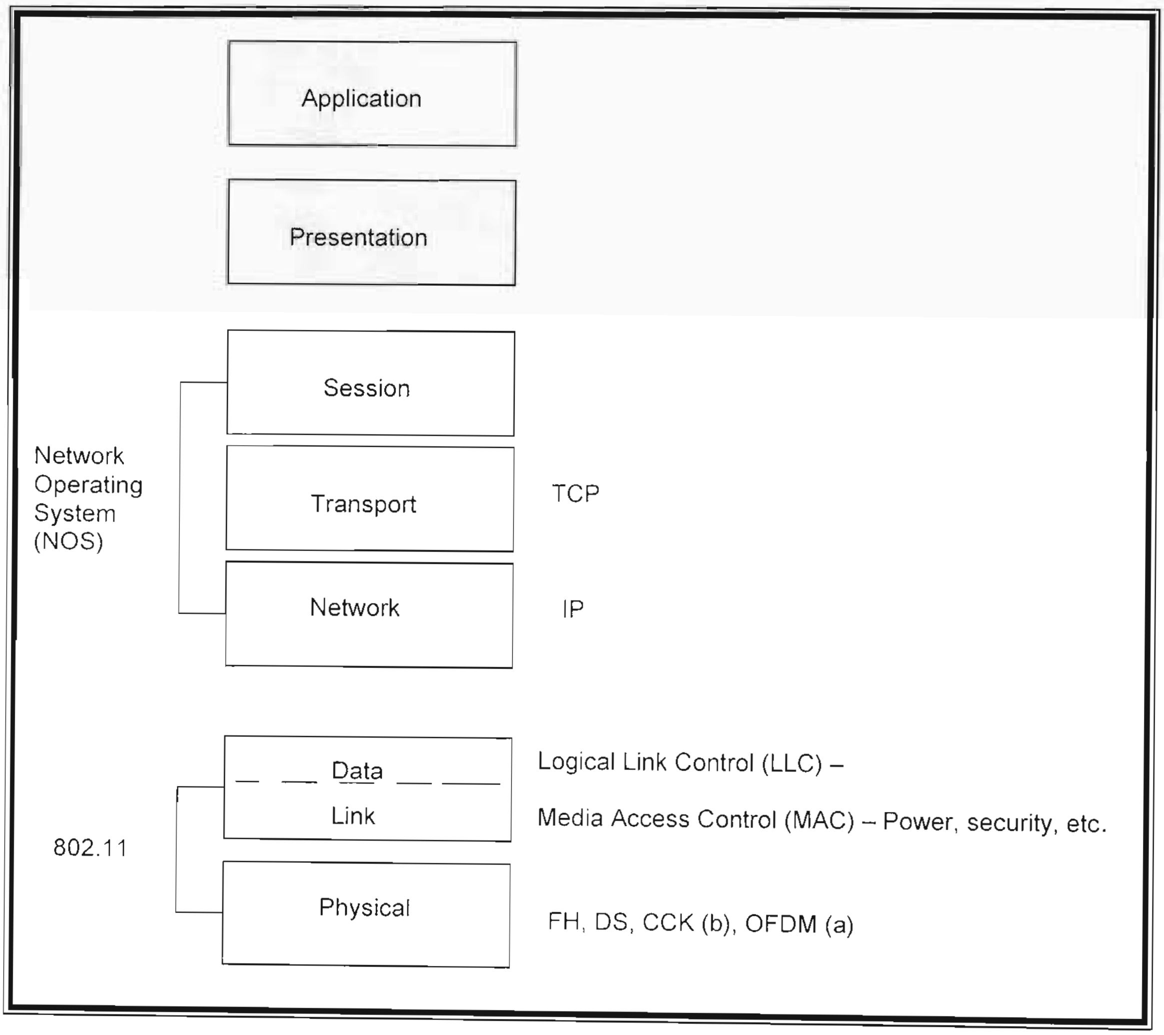

Figure 3.4: 802.11 and the OSI model

The 802.11 Data Link Layer is divided into two sublayers: Logical Link Control (LLC) and Media Access Control (MAC). The 802.11 LLC is the same as the LLC in the 802.2 standard with 48-bit addressing, making it easy to bridge from a wireless to a wired network. The MAC however is specifically created for the WLANs. Unlike Ethernet networks the 802.11 MAC layer does not support Carrier Sense Multiple Access with Collision Detection (CSMA/CD) due to its inability to 'hear' a collision which is drowned by the radio transmission as a result of the 'near/far' problem. The 'near/far' problem arises when signals from nodes close to the BS drown out the signals from nodes that are far away from the BS. Instead Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) or Distributed Coordination Function (DCF) is used. To confirm that a data packet was received intact the receiving station sends out an acknowledgement (ACK) packet to the sender. This is called explicit packet acknowledgement which is used in an attempt to avoid collisions by the CSMA/CA.

CSMA/CA works as follows. If a station wants to send a packet, it first senses its environment. If the channel is idle, the station waits an additional randomly selected period of time (backoff-period). After this backoff-period if the medium is still free, then the station transmits. If the packet is received intact, the receiver sends out an acknowledgement to the sender to confirm that the packet was received intact. Once the sender receives this acknowledgement, the process is complete. If however the sender does not receive this acknowledgement, it will assume that a collision has occurred therefore the original packet was not received intact regardless if it was just the acknowledgement packet that was not received intact and retransmits the packet after a random period of time.

This explicit ACK mechanism also handles interference and other radio-related problems very effectively. The advantages of the ACK mechanism comes at the cost of increased overhead which contributes to the fact that an 802.11 LAN will always have slower performance compared to an equivalent Ethernet LAN.

3.3.1 Distributed Coordination Function

Distributed Coordination Function (DCF) is contention based and provides asynchronous transmission which is mandatory in all 802.11 stations. DCF can be used in both ad-hoc and infrastructure modes.

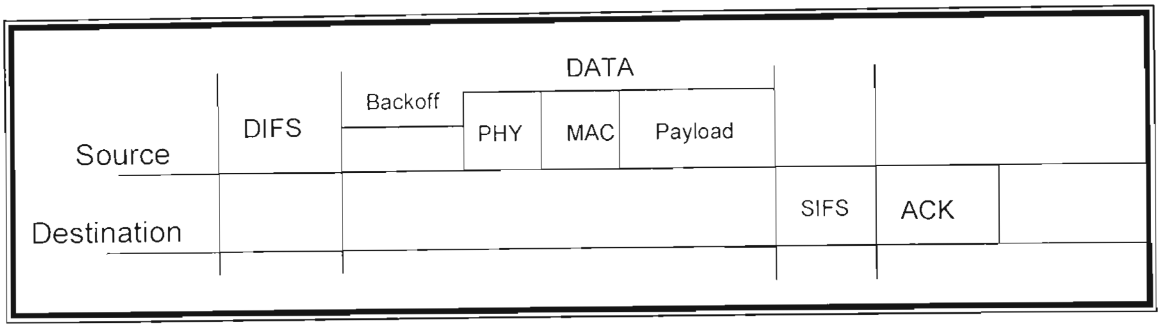

Figure 3.5: Basic Operation of 802.11 DCF

Another problem facing the 802.11 MAC is the 'hidden node' issue. When two nodes are on opposite sides of an access point, they can 'hear' activity from the access point but not from each other, usually due to distance or an obstruction. In order to solve this problem RTS/CTS was introduced. Request to Send (RTS) is when a node sends a message to the access point requesting to send a packet. Clear to Send (CTS) is the acknowledgement message the access point returns giving the node permission to transmit its packet. This action by the access point causes other nodes to delay their intended transmissions. This solution however, requires additional overhead, thus it is generally used only on the largest-sized packets, since retransmission would be expensive from a bandwidth standpoint.

Cyclic Redundancy Check (CRC) is another robustness feature offered in the MAC layer to ensure that the data is not corrupted in transit. Packet Fragmentation offers robustness by breaking down the large packets into smaller packets before transmitting them over the air in highly congested environments or in high interference areas where the larger packets are more susceptible to be corrupted. Thus packet fragmentation reduces the number of retransmissions and improves the overall performance of the wireless network.

The MAC layer is responsible for the client's selection of an access point upon joining a Basic Service Set. The client uses the signal strength and packet error rate parameters for weighing the selection process of an access point. The access point accepts the client. The client then tunes to the radio channel to which the access point is set. After a certain time interval the client does a periodic check and surveys the other access points. If another access point can offer the client better performance, the client reassociates with this new access point and tunes to the radio channel to which this access point is set. A client node may also reassociate if there is high network traffic on the original access point; this process is known as load balancing. Load balancing serves to distribute the total WLAN load most efficiently over the wireless network.

3.3.2 Point Coordination Function

Point Coordination Function (PCF) provides synchronous transmission which basically implements a polling-based access and is not mandatory. PCF can only be used in Infrastructure mode. PCF provides a contention free period that alternates with the contention period. PCF specified in the 802.11 MAC supports time bounded data such as voice and video. PCF allows a single access point to control access to the media. Any station, whether it agreed to operate in PCF or not, but is associated, can transmit data as long as the AP allows it to do so. Since PCF gives every station a turn to transmit in a predetermined fashion, a maximum latency is guaranteed. A downside to PCF is that it is not particularly scalable, in that a single point needs to have control of media access and must poll all stations, which can be ineffective in large networks.

3.4 802.11b Protocol

The 802.11b amendment to the original standard was ratified in 1999. 802.11b has a maximum data rate of up to 11 Mbit/s and uses the same CSMA/CA media access method defined in the original standard. Due to the CSMA/CA protocol overhead, in practice the maximum 802.11b throughput that an application can achieve is about 5.9 Mbit/s over TCP and 7.1 Mbit/s over UDP [12].

802.11b uses the Direct-Sequence Spread Spectrum (DSSS) modulation technique. The 802.11b was widely accepted as the wireless LAN technology standard due to its considerable increase in throughput compared to the original standard along with its low cost. 802.11b uses the Wireless Equivalent Privacy (WEP) security scheme as well as the upgraded Wireless Protected Access (WPA).

802.11b is most commonly used in a point-to-multipoint configuration. In this configuration the access point uses an omni-directional antenna. The omni-directional antenna gives an equally spaced coverage area around the access point. Within this area the access point can communicate with one or more clients. With high-gain external antennas, the protocol can also be used in fixed point-to-point arrangements, which extends the range. The range can be extended further when line of sight is established. This is usually done in place of costly leased lines or very cumbersome microwave communications equipment [12].

When signal quality becomes an issue, 802.11b cards that operate at 11 Mbit/s will scale back proportionately to a lower data rate. Lower data rates use less complex and more redundant methods of encoding the data and therefore are less susceptible to corruption due to interference and signal attenuation. This is known as Adaptive Rate Selection. Also it can be noted that the data rate drops as the client moves further away from the access point.

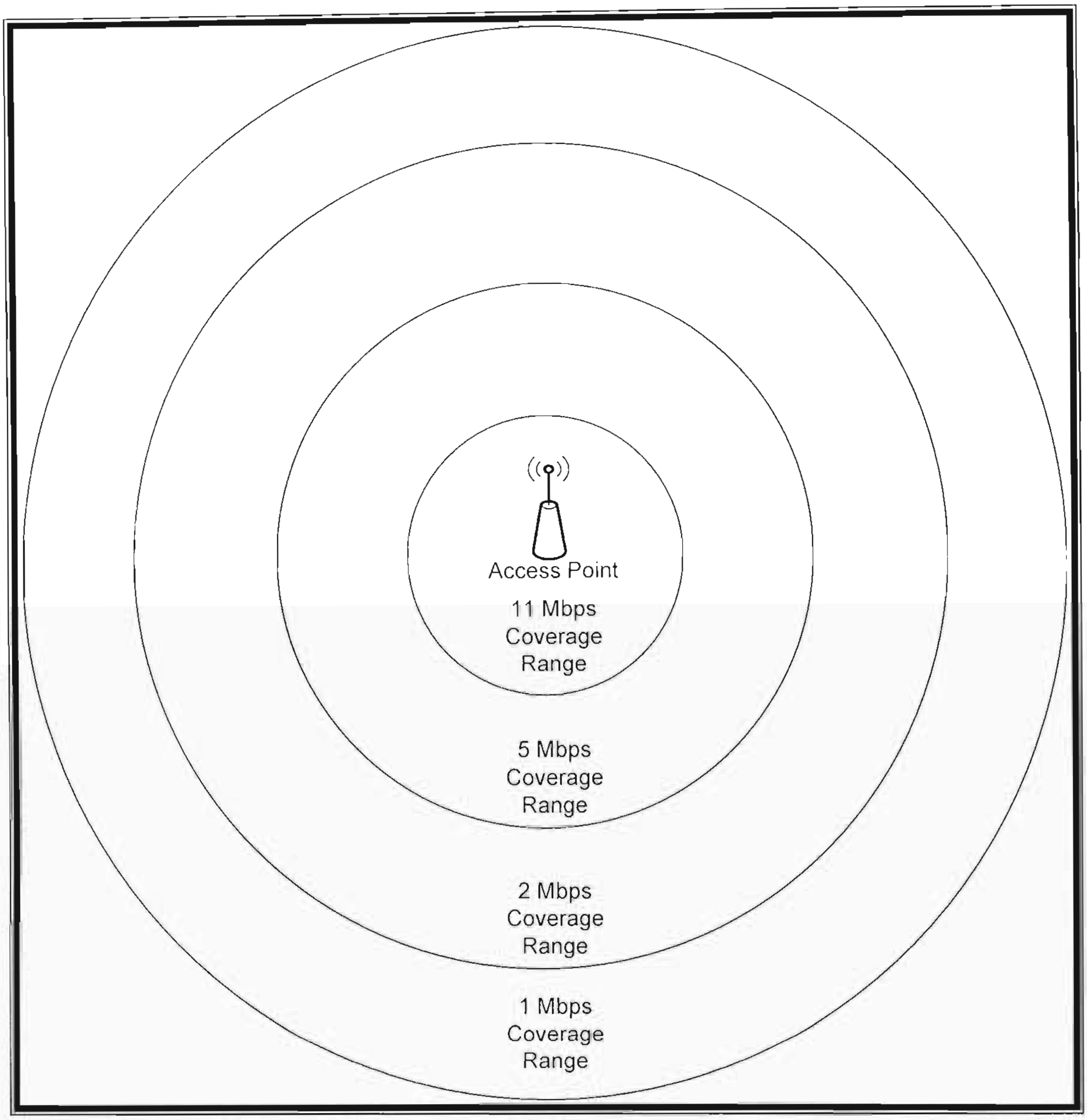

Figure 3.6: Dependence of Performance on Signal Quality and Range

3.4.1 802.11b Physical Layer

The 802.11b Physical Layer uses Direct Sequence Spread Spectrum (DSSS) technology which supports operation in the 2.4 GHz frequency band and gives a maximum data rate of up to 11 Mbps. 802.11b, like all other 802.11 physical layers, includes Physical Layer Convergence

Procedure (PLCP) and Physical Medium Dependent (PMD) sub-layers. The PLCP prepares 802.11 frames for transmission and directs the PMD to actually transmit signals, change radio channels, receive signals, and so on.

3.4.1.1 Physical Layer Convergence Procedure (PLCP)

When a station transmits an 802.11 frame the PLCP takes the frame and forms a PLCP protocol data unit (PPDU). The resulting PPDU includes the following fields:

- **Start Frame Delimiter**. The beginning of a frame is known by the field which is always 1111001110100000.
- **Signal**. The binary value of this field is equal to the data rate divided by 100Kbps. To ensure that the receiver uses the correct demodulation mechanism initially the PLCP fields are always sent at the lowest rate, which is 1Mbps.
- **Service**. This field is reserved for future use by the 802.11 standard and is always set to 00000000.
- **Length**. Time taken to transmit the contents of the PPDU. The number of microseconds is used to determine the end of the frame by the receiver.
- **Frame Check Sequence**. This field contains the 16-bit cyclic redundancy check (CRC) result for detecting possible errors in the Physical Layer header. Error detection is done on the PPDU contents by the MAC Layer as well.
- **PSDU**. Physical Layer Service Data Unit represents the actual 802.11 frame that is being sent which is the contents of the PPDU.
- **Sync**. When a receivable signal is present the sync alerts the receiver, which begins to synchronise with the incoming signal.

3.4.1.2 DSSS Spreading function

DSSS spreads the 802.11b data frame signal across approximately 30 MHz of the 2.4GHz frequency band, resulting in greater immunity to radio frequency (RF) interference as compared to narrowband signaling. For this reason the Federal Communications Commission (FCC) deemed the operation of spread spectrum systems as license free. To avoid channel overlap the 802.11b access points must be set to specific channels.

Through the use of a binary adder, an 802.11b transmitter combines the PPDU with a spreading sequence in order to actually spread the signal. The spreading sequence is a binary code. For 1Mbps and 2Mbps operation, the spreading code is the 11-chip Barker sequence, which is

10110111000. The Barker sequence is static. The binary adder effectively multiplies the length of the binary stream by the length of the sequence, which is 11. This increases the signaling rate and makes the signal span a greater amount of frequency bandwidth.

5.5Mbps and 11Mbps operation of 802.11b uses Complementary Code Keying (CCK) to provide the spreading sequences at these higher data rates. CCK derives a different spreading code based on fairly complex functions depending on the pattern of bits being sent.

3.4.1.3 DSSS Modulation

The modulator converts the spread binary signal into an analog waveform through the use of different modulation types, depending on which data rate is chosen. For example with 1Mbps operation, the PMD uses differential binary phase shift keying (DBPSK). The modulator merely shifts the phase of the centre transmit frequency to distinguish a binary 1 from a binary 0 within the data stream.

For 2Mbps transmission, the PMD uses differential quadrature phase shift keying (DQPSK), which is similar to DBPSK except that there are four possible phase shifts that represents every two data bits. This process enables the data stream to be sent at 2Mbps while using the same amount of bandwidth as the one sent at 1Mbps. The modulator uses similar methods for the higher, 5.5Mbps and 11Mbps data rates.

After RF amplification takes place based on the transmit power, the transmitter outputs the modulated DSSS signal to the antenna in order to propagate the signal to the destination. The trip en-route to the destination will significantly attenuate the signal, but the receiver at the destination will detect the incoming Physical Layer header and reverse (demodulate and despread) the process implemented by the transmitter. The DSSS system consumes more power than the FSSS system since it is more complicated. On the other hand, the DSSS is more resistant to interference than Frequency Hop Spread Spectrum (FHSS). Transmission time in DSSS is shorter than FHSS since it does not need to wait to change the frequency.

3.4.1.4 Complementary Code Keying (CCK) Modulation

In order to achieve the higher data rates in IEEE 802.11b, an eight-chip complementary code keying (CCK) modulation scheme is used. The High Rate Direct Sequence Spread Spectrum (HR/DSSS) uses a code word that is derived partially from data. Thus, it carries information as well as spreads the signal. Even in the presence of noise and interference they are able to be correctly identified from one another due to the mathematical properties incorporated within the CCK.

3.4.1.5 RF Interference

RF Interference involves the presence of unwanted, interfering RF signals that disrupt normal system operations. When the power of the interferer is significant, it may cause the station (STA) to become inactive for indefinite periods until the interferer disappears. An interfering signal basically causes an energy jump in radio and may start abruptly in the middle of a transmission causing a collision or the packet to be received by the destination with errors. When this occurs the destination does not send an acknowledgement to the sender, causing the sender to retransmit, adding overhead on the network. This causes delays. If the interference continues, the STA either automatically switches to a lower rate which slows the use of wireless applications, or holds off until the medium is clear which could cause infinite delays.

For a WLAN, another WLAN that operates in the same channel is also an interferer. This situation is likely to occur in IEEE 802.11 and 802.11b since there are only three non-overlapping channels.

3.5 802.11g Protocol

802.11g was a late comer as it was only ratified in 2003 but has made a significant impact thus far. It operates in the 2.4GHz frequency range and can support up to a 54 Mbps data rate. Since it operates in the same unlicensed frequency band as 802.11b it suffers from the same interference problems from microwaves, Bluetooth and cordless phones. Like 802.11b, it also uses the 3 non-overlapping channels from the available 14 channels. 802.11g is backward compatible with 802.11b providing investment protection for 802.11b clients. 802.11g supports all the data rates of both 802.11a and 802.11b protocols. 802.11b uses Direct Sequence Spread Spectrum (DSSS) for 1, 2, 5.5, and 11 Mbps data rates which is adopted by 802.11g and likewise uses Orthogonal Frequency Division Multiplexing (OFDM) for data rates used by 802.11a.

Table 3.1: 802.11g Data rates, Transmission type and Modulation schemes

Data Rate (Mbps)	Transmission Type	Modulation Scheme
54	OFDM	64 QAM
48	OFDM	64 QAM
36	OFDM	16 QAM
24	OFDM	16 QAM
18	OFDM	QPSK1
12	OFDM	QPSK
11	DSSS	CCK2
9	OFDM	BPSK3
6	OFDM	BPSK
5.5	DSSS	CCK
2	DSSS	QPSK
1	DSSS	BPSK

In a network the effective throughput that one receives is considerably lower than the actual given data rate. When 802.11g is in the same network as 802.11b clients the throughput drops to 802.11b rates and when in the same network as 802.11a clients the 802.11g throughput increases to 802.11a rates. The effective data rates for 802.11g are as shown in the table below.

Table 3.2: Approximate throughput comparison of 802.11a, 802.11b, and 802.11g

	Data Rate (Mbps)	**Approximate** Throughput (Mbps)
802.11b	11	5.7
802.11g-with 802.11b clients in cell (CTS/RTS)	54	9.1
802.11g-with 802.11b clients in cell (CTS-to-self)	54	13
802.11g (no 802.11b clients in cell)	54	27
802.11a	**54**	**27.3**

The 802.11g standard provides an option called CTS to Self, which is able to provide greater throughput when in mixed-cell mode. As the name suggests, CTS to Self dispenses with the RTS and relies upon the 802.11b client's clear channel assessment capabilities to check for an open medium.

3.6 802.11a Protocol

IEEE 802.11a amendment to the original standard was ratified in 1999. It operates in the 5GHz frequency range using a 52 sub-carrier OFDM. 48 of those sub-carriers are for data whilst the remaining 4 are pilot sub-carriers with a carrier separation of 0.3125 MHz. Each of these sub-carriers can have different modulation types such as BPSK, QPSK, 16-QAM or 64-QAM. For a more technical description of the 802.11a protocol the reader can refer to the works of [12].

802.11a can provide a maximum data rate of 54Mbps but has an actual throughput of approximately 27 Mbps. 802.11a has 12 non-overlapping channels. Unlike 802.11b and 802.11g, the advantage of operating in the 5GHz frequency range is the lack of interference from other electronic devices. To counter this advantage the high carrier frequency limits 802.11a to almost line of sight, requiring more access points which in turn increase the cost of the network. The superior performance of 802.11a offers excellent support for bandwidth hungry applications, but the higher operating frequency equates to relatively shorter range as compared to 802.11b. 802.11a and 802.11b do not interoperate due to the different modulation schemes and radio frequencies.

3.7 Voice over IP (VoIP)

The role of VoIP in this research is an important one as it provides a cost-effective voice communication solution. Henceforth for rural regions where it is infeasible to bear the infrastructure costs to provide PSTN networks to the people, they may still be provided with voice communication services by VoIP.

Voice over IP (VoIP) uses the internet protocol (IP) to transmit voice as packets over an IP network. So VoIP can be achieved on any data network that uses IP, like Internet, Intranets and Local Area Networks (LAN). Here the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network. Signaling protocols are used to set up and clear calls, carry information required to locate users and negotiate capabilities.

The cost of a packet switched network for VoIP could be less than half that of a traditional circuit-switched network (such as PSTN) for voice transmission. This expected cost saving is the result of more efficient use of bandwidth requiring fewer long distance trunks between switches.

The traditional circuit-switched networks have to dedicate a full-duplex 64kbit/s channel for the duration of a single call, whereas with VoIP networks the bandwidth is used only when something has to be transmitted. More efficient use of bandwidth enables more calls to be carried over a single link, without requiring the carrier to install new lines or further augment network capacity. VoIP allows Internet access and voice traffic simultaneously over a single network which potentially eliminates the need for two separate networks.

This model uses Web telephones that can be used in conjunction with Internet services to bypass the public telephone network. This approach envelopes frames of compressed speech into IP packets. Typically, voice is compressed to 8kbps using standard methods. The IP overhead increases the data rate to 24kbps. Some devices use silence compression technology, so that bandwidth is required only when someone is actually talking. During periods of silence, bandwidth for voice is automatically freed up for other traffic on the enterprise network. This typically reduces the bandwidth utilization to about 16kbps. Some devices also use forward error correction methods to minimize loss, and jitter-buffer techniques to reduce latency variations. Quality of Speech is impacted by both the compression algorithms and by the lack of guaranteed QoS in the IP network. For this type of network, prioritization is directly related to QoS issues.

3.7.1 Sources of VoIP Network Delays

There are a number of factors that affect the application performance, as perceived by the end users. Perhaps the most significant of these is the end-to-end delay. The maximum recommended one-way delay for most user applications is 150ms, or a round trip delay of 300ms.

The speech encoder incurs delays based on the algorithm used. Other delays include switching, routing and other packet processing delays (typically a few milliseconds); transmission delays; signal propagation delays; jitter buffer delays (typically a settable parameter between 20-40ms); and decoding delays (typically half of the encoding delay).

Delay of packet transfer between the transmitter and the receiver will always exist. The delay should always be constant in order to minimize its impact on voice quality. Variation in delay is known as jitter, which is defined as the mean difference in packet spacing at the receiver as compared to the spacing at the sender. A jitter buffer is used to compensate for the difference in arrival time. Most jitter buffers can be selected to have a value between 0 and 255ms.

Packet loss and delay in the Internet degrades the quality of VoIP. Forward error correction (FEC) reduces packet loss at the cost of higher delay and bandwidth. An alternative is a loss-robust voice codec, but also at the expense of higher bit-rates.

The standard method of transporting voice samples through an IP based network requires the addition of three headers; one for each layer. These headers are IP, UDP and RTP.

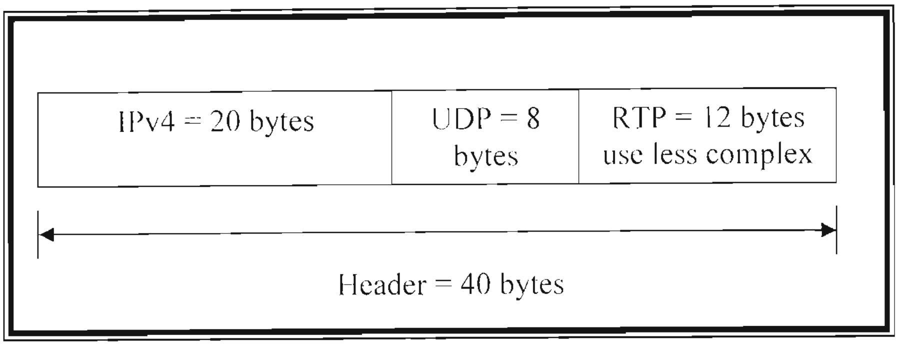

Figure 3.7: VoIP Header

The total length of this header information is 40 octets (bytes), or 320 bits. Each time a voice packet is transmitted, these headers are included, utilizing additional bandwidth which is determined by the number of packets sent each second.

Packet frequency is the number of packets containing voice samples which are sent per second. The packet frequency is the inverse of the duration in seconds represented by the voice samples. For example, if the voice samples in one packet represent a duration of 20 milliseconds, then 50 of these samples would be required each second. The packet frequency would therefore be 50.

The payload duration is reached by a compromise between bandwidth requirements and quality. Since the header length remains constant at 40 bytes the smaller payloads demand higher bandwidth per channel band. On the other hand if we increase the payload size, the overall delay of the system will increase, and the system will be more susceptible to the loss of individual packets by the network. The VoIP call will be sustained through the network provided the total latency including that of the terminating network is bounded to 300ms.

3.7.2 Voice Codecs

In VoIP systems several codecs are available to translate voice into packets. Research has determined the G.711, G.729 and G.723.1 as the most favourable codecs to use in an IP network. The G.711 codec operating at a data rate of 64Kbps gives PSTN quality voice. The G.723.1 speech vocoder passes less distortion while passing through DTMF tones as compared to other ITU-T codecs close to G.723.1 codec compression rates. The G.729A CS-ACELP has the same specifications as the G729 but offers much less complexity. The G729 and G729A provide less algorithm delay. G.729 and G.723.1 utilizes less bandwidth than G.711 hence there is always a trade-off between cost and quality. The properties of the abovementioned codecs are shown in the table below.

Table 3.3: Codec Properties

Codec & Bit rate	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Voice Payload Size (bytes)	Voice Payload Size (ms)	Packets per Second (pps)	Codec Lookahead (ms)
G.711 (64kbps)	80	10	160	20	50	
G.729 (8kbps)	10	10	20	20	50	5
G.723.1 (6.3kbps)	24	30	24	30	34	7.5
G.723.1 (5.3kbps)	20	30	20	30	34	7.5

Table 3.4: Shows the relevant characteristics of the most common coding algorithms.

Coding	algorithm	Bandwidth	Sample	IP bandwidth
G.711	PCM	64kbps	0.125ms	80kbps
G.723.1	ACELP	5.6kbps	30ms	16.27kbps
G.723.1	MP-MLQ	6.4kbps	30ms	17.07kbps
G.729(A)	CS-ACELP	8kbps	10ms	24kbps

Each codec provides a certain quality of speech. A common benchmark used to determine the quality of sound produced by specific codecs is the Mean Opinion Score (MOS) [14]. The MOS is determined using a large sample of subjective listening tests. It is judged on a scale of 1 (bad) to 5 (excellent). The E-Model is a passive objective method for estimating user satisfaction with the quality of narrowband handset telephony. The E-Model computes a "Rating Factor" R ranging from 0 to 100. R can be translated to a conversational quality MOS (MOS-CQ) according to the following equation:

MOS-CQ = 1 + 0.035R + 7R(R – 60) (100 – R) 10 – 6 (3.1) [15]

Note that the E-Model produces a theoretical maximum MOS-CQ of 4.5 when R is 100.

The table below taken from ITU-T G.109 illustrates the relationship between R, MOS-CQ, and user satisfaction.

Table 3.5: E-Model Relationships

R	MOS-CQ	User Satisfaction
90 – 100	4.3 – 5.0	Very Satisfied
80 – 89	4.0 – 4.3	Satisfied
70 – 79	3.6 – 4.0	Some Dissatisfied
60 – 69	3.1 – 3.6	Many Dissatisfied
50 – 59	2.6 – 3.1	Nearly All Dissatisfied
0 - 49	1.0 – 2.6	NOT Recommended

The G.723.1 has a 30ms payload size, transmitting only 34 packets per second and has an algorithm delay of 7.5ms thus the overall delay of the system will increase and the system will be more susceptible to the loss of individual packets by the network, resulting in leaving nearly all the clients dissatisfied. The G.729A speech codec software was primarily developed for use in multimedia simultaneous voice and data applications. The speech quality of a G.729 codec along with the good compression rates promises to be an excellent choice for many applications including VoIP.

Table 3.6: Codec Properties [4]

Codec	Algorithm	Bit Rate (Kbps)	Maximum Voice Quality
G.711	PCM	64	Very Satisfied
G.723.1 @ 6.3Kbps	MP-MLQ	6.3	Satisfied
G729A	CS-ACELP	8	Some Dissatisfied

The E-Model is very useful in revealing the behavior of the different codecs as a function of one-way delay or loss. Under ideal network conditions, it reveals the expected speech quality provided by different codecs.

Table 3.7: Estimated User Satisfaction with One-Way delay T_a and no loss

Codec	T_a =0msec	T_a =150ms	T_a =300ms	T_a =400ms
G.711	Very Satisfied	Very Satisfied	Some Dissatisfied	Many Dissatisfied
G.729A+VAD @ 8Kbps	Satisfied	Some Dissatisfied	Many Dissatisfied	Nearly All Dissatisfied
G.723.1 @ 6.3Kbps	Some Dissatisfied	Some Dissatisfied	Nearly All Dissatisfied	Nearly All Dissatisfied

It illustrates that most codecs experience a gradual roughly 4 point reduction in R as delay increases from 0 to 150ms with a much faster reduction beyond 175ms.

Table 3.8: Shows the relationship between Codecs and MOS scores.

Compression Method	Bit Rate (Kbps)	MOS Score	Compression Delay (ms)
G.711 PCM	64	4.1	0.75
G.726 ADPCM	32	3.85	1
G.728 LD-CELP	16	3.61	3 to 5
G.729 CS-ACELP	8	3.92	10
G.729 x 2 Encodings	8	3.27	10
G.729 x 3 Encodings	8	2.68	10
G.729a CS-ACELP	8	3.7	10
G.723.1 MP-MLQ	6.3	3.9	30
G.723.1 ACELP	5.3	3.65	30

The G.729A CS-ACELP has a MOS of 3.92 which means it has an R value in the range of 70-79 which is acceptable. One of the main drawbacks to compressing voice is signal distortion due to multiple encodings (called tandem encodings). For example, when a G.729 voice signal is tandem encoded three times, the MOS score drops from 3.92 (very good) to 2.68 (unacceptable). Another drawback is codec-induced delay with low bit-rate codecs [14].

The number of codec samples per packet is another factor in determining the bandwidth and delay of a VoIP call. The codec defines the size of the sample, but the total number of samples placed in a packet affects how many packets are sent per second. Typically one voice sample is sent per packet. When you increase the voice payload size the VoIP bandwidth reduces and the overall delay increases as will be seen in the simulation work in chapter four.

3.8 WiMax as a Backhaul

WiMax is defined as Worldwide Interoperability for Microwave Access by the WiMax Forum, formed in June 2001 to promote conformance and interoperability of the IEEE 802.16 standard [16]. The original WiMax standard (IEEE 802.16) specified WiMax for the 10 to 66 GHz range. 802.16a was updated in 2004 to 802.16d (also known as 'fixed' WiMax) specified for the 2 to 11 GHz range. 802.16d was updated to 802.16e in 2005 (known as 'mobile' WiMax) and uses scalable orthogonal frequency division multiplexing (OFDM) as opposed to the OFDM version with 256 sub-carriers used in 802.16d. WiMax can provide broadband wireless access (BWA) up to 50 km for fixed stations, and 5 - 15 km for mobile stations [60]. WiMax is a second-generation protocol that allows for more efficient bandwidth use, interference avoidance, and is intended to allow higher data rates over longer distances. The 802.16 MAC uses a scheduling algorithm for which the subscriber station need only compete once for initial entry into the network. Thereafter the base station assigns an access slot to the subscriber station. The size of the access slot can be varied according to the network environment but will remain assigned to the subscriber station. The 802.16 scheduling algorithm is stable under overload and over-subscription. QoS parameters are controlled by a scheduling algorithm which assigns time-slots among the applications according to the needs of the subscriber stations. The bandwidth and reach of WiMax make it suitable for the following potential applications [16]:

- Connecting Wi-Fi hotspots with each other and to other parts of the Internet
- Providing a wireless alternative to cable and DSL for last mile (last km) broadband access.
- Providing high-speed mobile data and telecommunications services (4G)

In line with these possible applications is the technology's ability to serve as a high bandwidth "backhaul" for Internet or cellular phone traffic from remote areas back to an internet backbone. Areas of low population density and flat terrain are particularly suited to WiMax and its range. For countries that have skipped wired infrastructure as a result of inhibitive costs and unsympathetic geography or lack of provision (due to political background) as in SA, WiMax can enhance wireless infrastructure in an inexpensive, decentralized, deployment-friendly and effective manner.

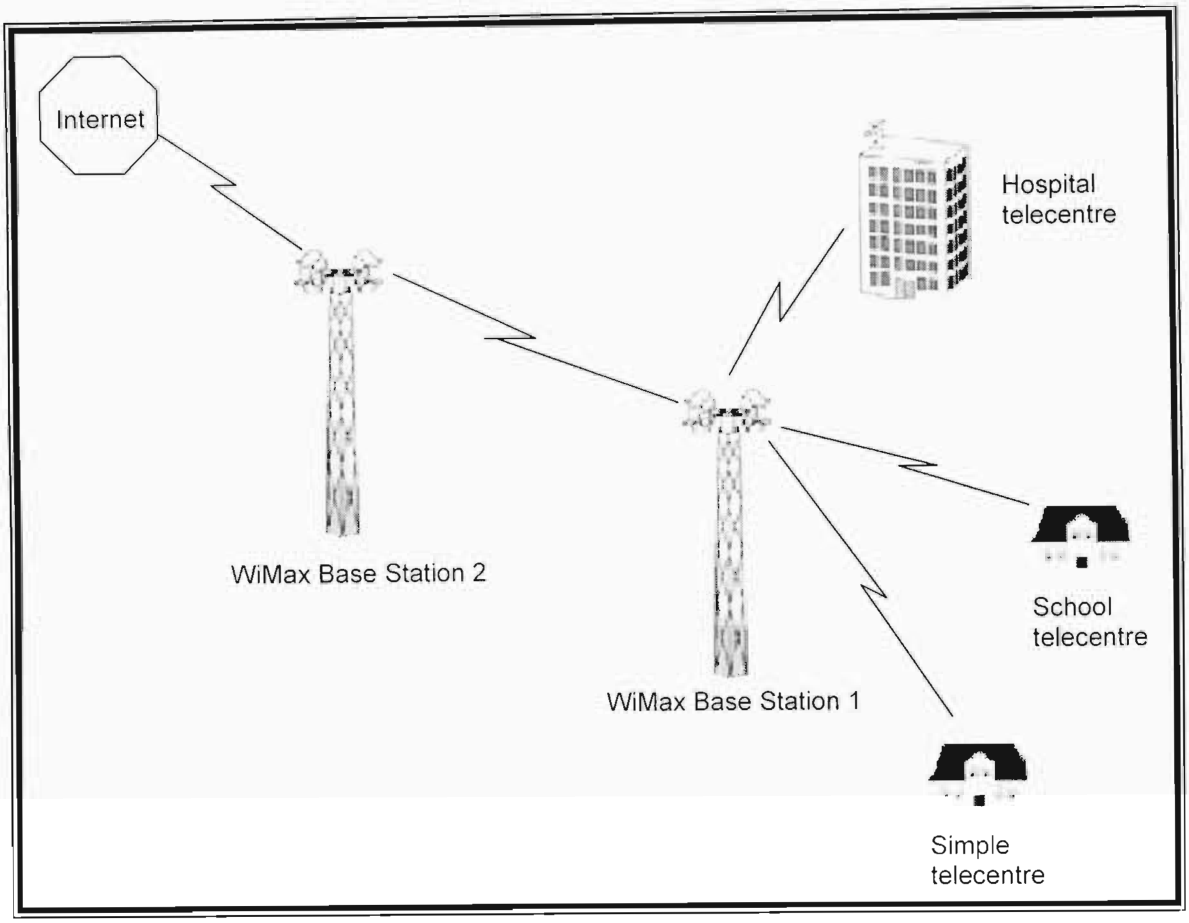

Figure 3.8: Illustrates the use of WiMax as a backhaul to Interconnect three telecentres in a rural area to the Internet.

3.9 SUMMARY

This chapter explained the operation of the protocols used in WiFi WLANs. Literature suggests that 802.11b is best suited for low cost networks with low data rate applications and where range is an issue. However 802.11g in compatibility mode exhibits a number of similar features to 802.11b yet offers a higher data rate that is able to cater for high bandwidth applications. Hence the performance of these two protocols is evaluated via simulation in Chapter Four.

How voice is transmitted over the internet and how the choice of Codec may affect the speech quality according to the Mean Opinion Score (MOS) was also discussed. The G.729 CS-ACELP has a MOS score of 3.92 which means most of the clients are satisfied. It also has a 10ms compression delay as compared to the G.723.1 which has a 30ms compression delay. Theory boasts that the G.729 codec is a perfect choice for VoIP applications in IP networks.

WiMax is the proposed solution to be used as the backhaul in these rural scenarios due to its remarkable qualities. One of its most admirable features is the ability to wirelessly transmit higher data rates over longer distances.

CHAPTER FOUR: SIMULATION RESULTS

4.1 Introduction

The objective of our simulation studies is to characterize some of the proposed WLANs mentioned in chapter two. The projected traffic for the time period between 2006 until 2014 will be applied to the proposed LANs. Service Differentiation will be applied to establish QoS in an 802.11 WLAN. Delay values of the applications will be plotted against the throughput. This is done to evaluate if voice applications can be sustained over the 8 year period with reasonable quality of service.

All the simulations for network performance characterization have been performed using the discrete event simulator OPNET Modeler version 11.5 [17]. OPNET is commonly used for network simulation and provides a rich library of models for implementing wired and wireless simulation scenarios. In this chapter we begin with a brief discussion of the modeling and simulation methodology used in OPNET.

4.2 Network Modeling Using OPNET

OPNET is among the leading discrete event network simulators used both by the commercial and research communities [17]. It provides a comprehensive framework for modeling wired as well as wireless network scenarios. The hierarchy of the simulation model consists of three main levels. The simulation network is the highest level which defines the network layout, the nodes and the configuration of attributes of the nodes comprising the scenario. The second level refers to the node model which consists of an organized set of modules describing the various functions of the node. The modules in the nodes are implemented using process models, the lowest level in the hierarchy. Process models consist of finite state machines, definitions of model functions, and a process interface that defines the parameters for interfacing with other process models and configuring attributes. Finite state machine models are implemented using Proto C, which is a discrete event library based on C functions [17].

OPNET Modeler uses an object-oriented approach for the development of models and simulation scenarios. The OPNET parameter settings, descriptions and queuing system concepts have been copied into Appendix B1 and B2.

4.3 Hospital Telecentre Simulation Scenario

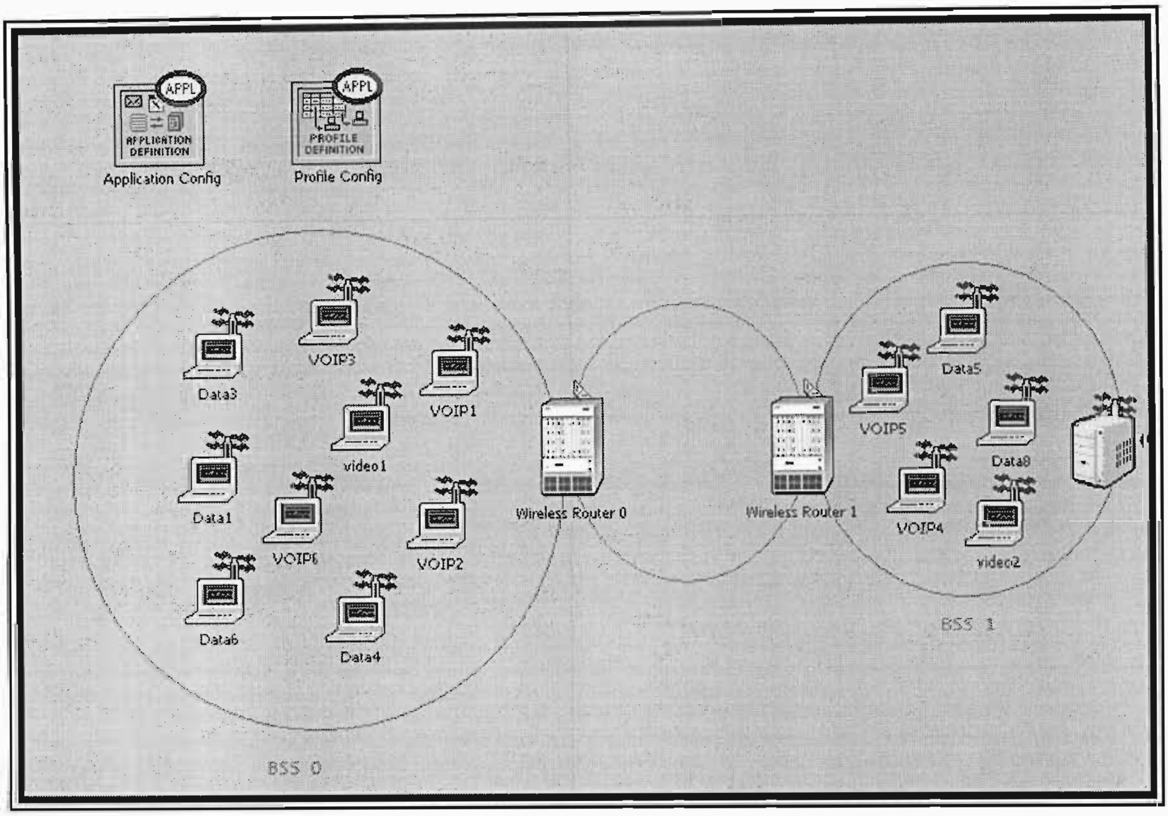

Figure 4.1: OPNET snapshot of the Hospital telecentre scenario

The above scenario was set up in OPNET. The application configuration box in the top left corner defines all the applications and their respective parameters that will be available on the network. It is configured to support five applications, namely; FTP, email, web-browsing, voice and videoconferencing. The task at hand is to represent a population of users. User behaviour or any attribute that involves time taken by the user to think can best be modeled using an exponential distribution. To correctly represent a population of users, a distribution with a memoryless property is required and one that can be used when outcomes are independent of each other. Thus an exponential distribution successfully meets the requirements.

FTP is classified as a Best-Effort (0) application. The File inter-arrival times has an exponential distribution with a mean of receiving a file every 5 minutes. The file sizes are exponentially distributed with a mean set for each year according to the file size increases as predicted in chapter two.

Email is considered another Best-Effort (0) Type of Service (ToS). The Email Send Inter-arrival time and the Receive Inter-arrival time are exponentially distributed with a mean of sending and receiving one email every 5 minutes. Just like FTP, the email sizes are exponentially distributed with a mean set for each year according to the email size increases as predicted in chapter two.

Web-browsing is set at searching mode which is approximately 560.96 kbps. Type of Service is Best-Effort (0).

Video-Conferencing is initially set at low resolution which has a frame size of 128 x 120 pixels and transmits frames at a rate of 10 frames per second. Using the OPNET standard of 9 bits per pixel, a frame size of 138.24 Kbps is determined. Thus video-conferencing produces a load of 1.38 Mbps on the network. The frame rate and resolution is adjusted according to the percentage increases predicted in chapter two. High resolution has a frame size of 128 x 240 pixels. Type of Service is Interactive Multimedia (5), assigning video the second highest priority.

Voice is transmitted using a G.729 codec with a frame size of 20ms and a look ahead size of 5ms. It has a coding rate of 8 Kbps. Transmission is set at 2 voice frames per packet. The talk-spurt length (incoming/outgoing) is set at exponential 0.35s and the silence length (incoming/outgoing) is set at 0.65s. It has a compression and decompression delay of 0.02s. It is assigned Interactive Voice (6) Type of Service, assigning voice with highest priority in this network. It should be noted that in an IP network a VoIP phone can make and receive many simultaneous calls and can have multi-way conversations. One may also think of voice devices as gateways that can handle multiple calls and redistribute them into the internal network.

Real-time applications (Voice & Video) use UDP protocol as compared to non-real time applications (FTP, email, web-browsing) which use TCP.

Once the applications are configured in the application configuration node, then the profiles of the users in the profile configuration are set. Three profiles are supported, namely My_Data, My_VoIP and My_Video. My_Data profile supports the three data applications, i.e.: FTP, Email and Web-browsing. My_Video supports the videoconferencing application and My_VoIP supports the voice application. All applications have a Start time offset of uniform distribution between 0 and 5 seconds. The duration of the data and videoconferencing applications are set to 'end of last task' and the repeatability of these applications are set to unlimited. The duration of the voice application is set to an exponential distribution of 180 seconds which equates to the mean holding time of a call. For the voice application the sub-parameters for the repeatability, which is the inter-repetition time (i.e. the time between each call), number of repetitions (i.e. the number of calls) and repetition pattern are set. The inter-repetition time has an exponential distribution while the number of repetitions has a Poisson distribution. These values are set according to the traffic predictions in chapter two. The tabled values can be found in Appendix A3. All the profiles are set to run simultaneously and have a start time with a uniform distribution between 100 and 110 seconds. The duration of each profile is till the end of its last application and the profile can only be repeated once at start time.

The parameters explained in Appendix B1 are then set accordingly on each node. The 802.11g (extended physical rate) protocol at 54 Mbps is applied on all the nodes. In this scenario a WLAN (BSS0) is connected to another WLAN (BSS1) using a wireless backbone configuration (BSS2), thus forming an Extended Service Set (ESS). The typical maximum range for the 802.11b and 802.11g protocol is 300m. The range was extended to approximately 600m using the wireless backbone configuration. This extended range is enough to cover an entire floor in a hospital, thus creating a hot-spot. In order to evaluate the delays a videoconferencing session between two nodes and three two-way conversations between the six VoIP phones was set up. At the same time FTP and Email downloads as well as web-browsing is set to take place across the network.

From the simulation results displayed in Table 4.1 it is observed that for the first six years the delay for voice remains fairly constant whilst the network load is increased due to increasing traffic according to user demand. FTP delays however increase steadily as the network load increases. The end-to-end network model has not been considered in detail but it is accepted that the VoIP call will be sustained on the network provided the total latency including that of the terminating network is bounded to 300ms.

Table 4.1: Simulation Results for the Hospital telecentre

Year	Voice Delay (ms)	Video Delay (ms)	FTP Delay (ms)	Network Load (Kbps)
2006	81.16	16.32	188.14	2332
2007	81.67	18.51	242.45	2565
2008	81.70	17.88	300.63	2721
2009	81.99	17.95	239.75	3123
2010	82.45	18.44	722.00	3830
2011	81.01	10.27	699.60	4663
2012	81.97	20.53	1161.00	6354
2013	82.38	20.71	957.40	7454
2014	83.11	20.80	1490.90	9896
2015	83.97	21.28	3359.00	10649
2016	84.22	21.12	7778.00	12974
2017	85.60	22.13	8708.00	14856
2018	91.61	32.85	18370.00	16667
2019	166.40	108.69	16820.00	19456

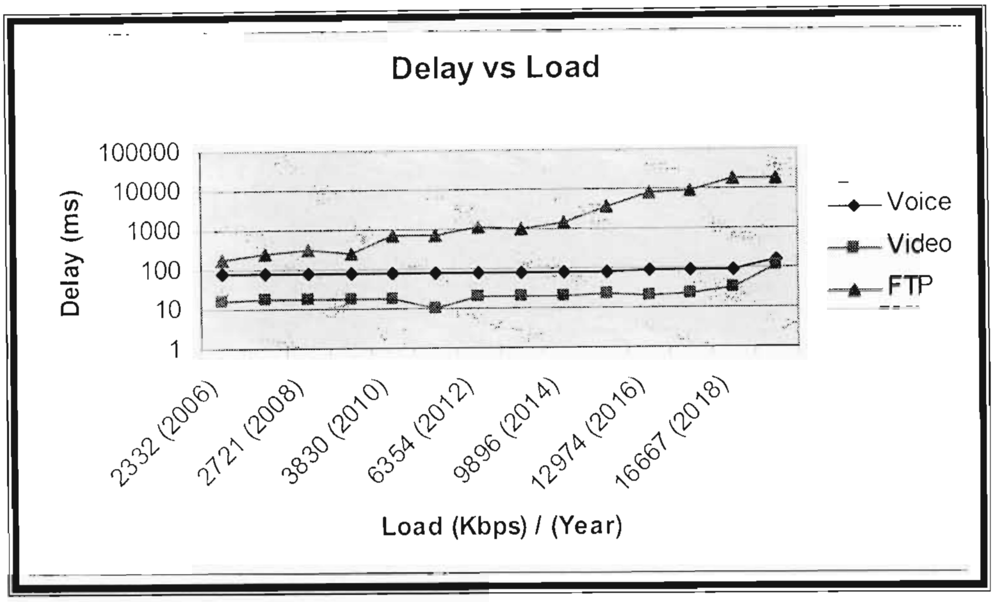

Figure 4.2: Delay vs. Load for the Hospital telecentre

Figure 4.2 shows the end-to-end delay of the various applications as the load is increased on the network. The voice packet delay remains fairly constant around 80ms which is way below its threshold of 150ms. Although the round trip delay for voice is usually 300ms, a 150ms round trip delay budget is proposed to maintain good quality of service whilst allowing room for the inter-LAN and WiMax backhaul delays. Since voice is assigned highest priority, it gains preference over the other applications and maintains a constant delay within its budget. Similarly videoconferencing is assigned second highest priority and also maintains a fairly constant delay below its budget of 100ms [19]. FTP is assigned Best Effort traffic which is the lowest priority; hence it suffers immensely compared to voice and video. FTP delays increase considerably as the load on the network is increased. To evaluate the maximum capacity this network can handle it was decided to heavily load the network beyond 2014. At 2014 the total load on this network is 9.896Mbps. Thereafter the network is loaded according to the trend increases for the following years. At 2017 one observes a slight degradation in the QoS as the delays increase. Good quality can be observed via simulation by the smoothness of the voice or video application curve. 2017 can be considered the threshold at 14.856Mbps since beyond this point there is a rapid deterioration in the QoS and the delays increase rapidly. This correlates to the work of Gast [20] which states that for an 802.11g protocol operating at 54Mbps with CTS-to-self protection the effective TCP payload throughput is 13Mbps. 2018 can be seen as the point where the wheels begin to fall off on this network dimensioning. At 2019 voice and video cannot be sustained with reasonable QoS as both applications exceed their delay budgets. One can observe from the graph that after 2017 the delay for video begins to increase more sharply than voice due to voice having priority over video. FTP reached a threshold in 2018 where the delay was so high that FTP data starts to get dropped thus decreasing the load on the network and at 2019 the delay for FTP also starts to decrease due to the reduced load.

4.4 School Telecentre Simulation Scenario

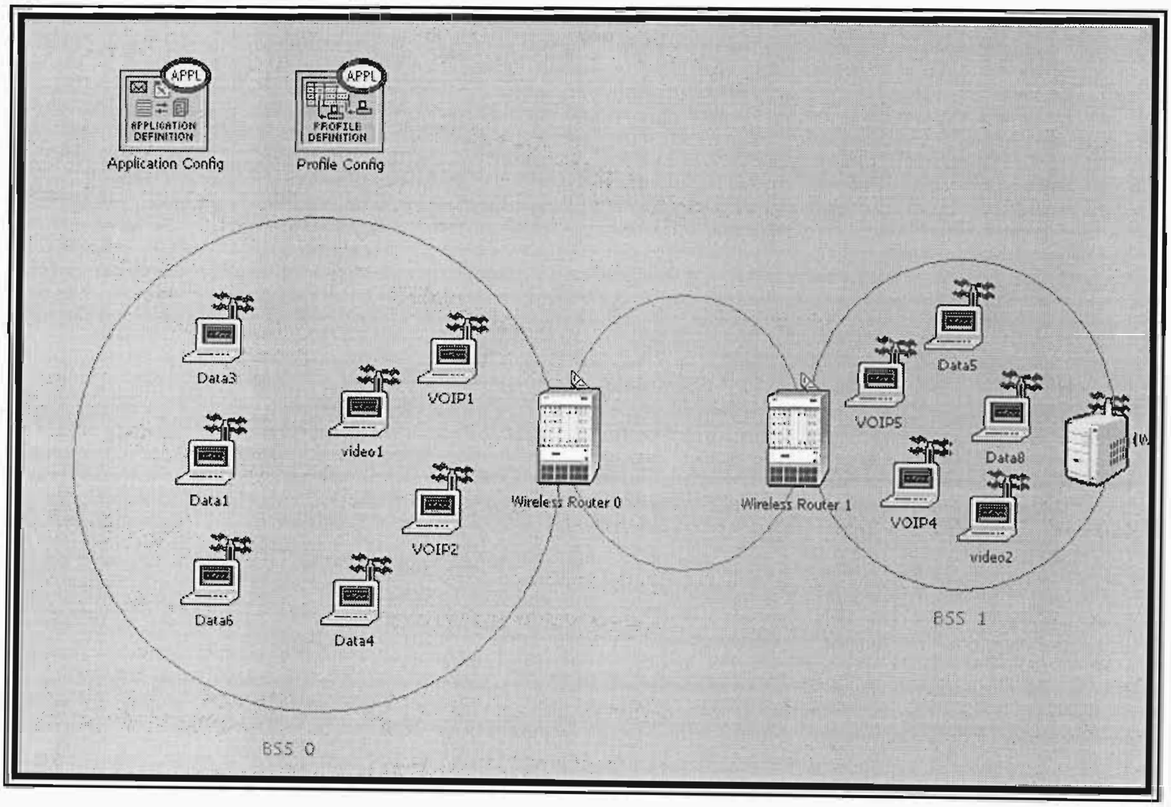

Figure 4.3 OPNET snapshot of the School telecentre scenario

The school telecentre scenario is very similar to the hospital telecentre scenario with regard to the set-up and applications offered. The profile and applications configuration is configured the same as for the hospital. The school telecentre will be equipped with 8 personal computers and 4 VoIP phones. Two of the 8 computers are assigned to run e-learning applications only. The following results evaluate the delays experienced by the applications in the network. FTP, email and web-browsing applications are running simultaneously with VoIP and videoconferencing sessions. The duration of the simulation run is 360 seconds. The 802.11g (extended physical rate) protocol at a data rate of 24 Mbps is applied on all the nodes.

Table 4.2: Simulation Results for the School telecentre

Year	Voice Delay (ms)	Video Delay (ms)	FTP Delay (ms)	Network Load (Kbps)
2006	80.53	9.94	190.09	3226
2007	80.57	10.09	189.29	3469
2008	80.63	10.13	195.14	3775
2009	81.43	16.81	365	4398
2010	81.59	16.36	538	5203
2011	81.99	16.43	1240	6414
2012	83.89	33.65	1630	8702
2013	84.55	33.44	2420	10380
2014	88.61	36.7	7720	10030
2015	94.79	42.68	23130	14283
2016	96.03	43.63	22330	15058
2017	200.22	152.99	20300	17020

According to Gast [20] the effective data rate of 24 Mbps is approximately 12 Mbps. It is observed from Table 4.2 that from 2006 till 2011 the delays for voice and video remain fairly constant whilst FTP delays increase stealthily as the load on the network increases. Beyond 2011 as the load approaches its capacity of approximately 12 Mbps the delays increase significantly resulting in a slight degradation of the QoS.

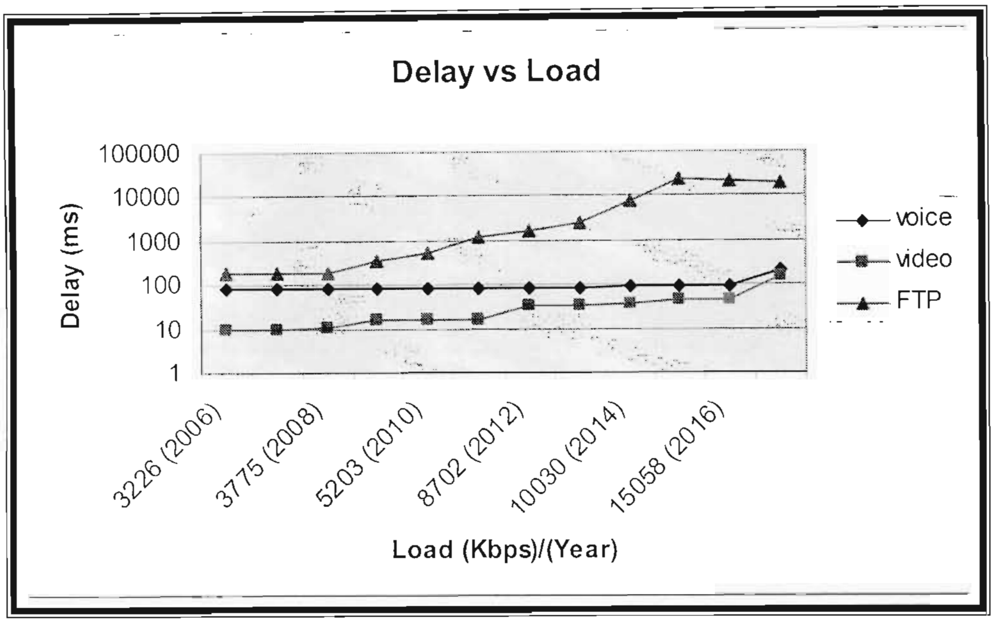

Figure 4.4: End-to-end Delay vs. Load for the School telecentre

It can be observed from Figure 4.4 and Table 4.2 that the voice packet end-to-end delay of voice begins to increase more sharply beyond 2013. As the load on the network increases significantly, so to does the FTP delay. Beyond 2012 FTP experiences large increases in the seconds range. FTP is assigned Best Effort traffic which is the lowest priority; hence it suffers from large delays as compared to voice and video. FTP reaches its threshold at 23.1sec in 2015 after which the FTP data starts to get dropped thus decreasing the load on the network and at 2016 and 2017 the delay for FTP also starts to decrease due to the reduced load.

To evaluate the maximum capacity this network can handle it was decided to heavily load the network beyond 2014. At 2014 the total load on this network is 10.030Mbps. Thereafter the network is loaded according to the trend increases for the following years. 2015 can be considered the threshold at 14.283Mbps since beyond this point there is an observable deterioration in the QoS and the delays increase rapidly. At 2017 voice and video cannot be sustained with reasonable QoS as both applications exceed their delay budgets of 150ms and 100ms respectively.

4.5 Simple Telecentre Simulation Scenario

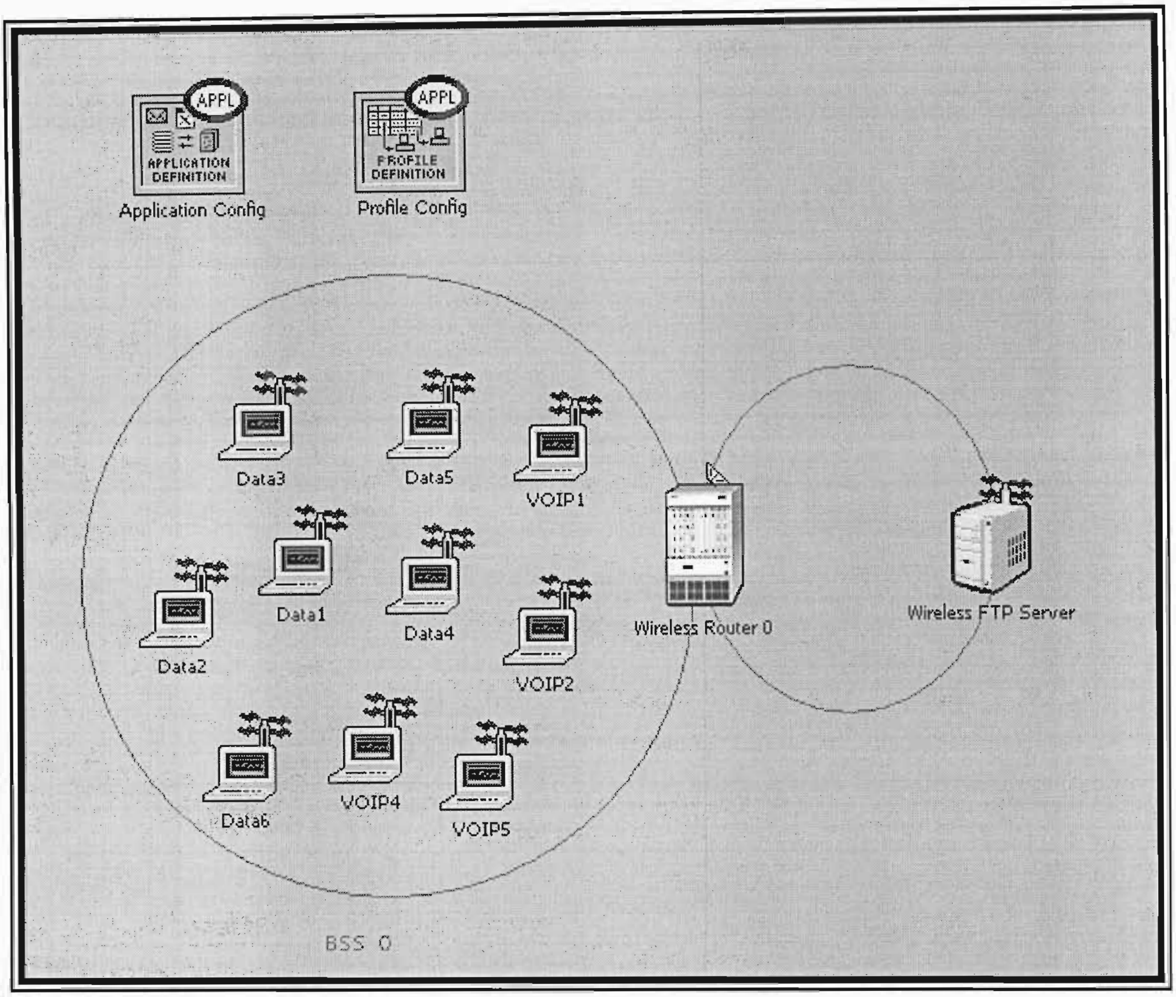

Figure 4.5: OPNET snapshot of the Simple telecentre scenario

The simple telecentre is equipped with 4 VoIP phones and 6 computers offering data applications such as email, web-browsing and FTP. The 10 nodes are placed within a single basic service set (BSS) with a wireless router enabled as the access point. It is connected wirelessly to an FTP server which supports FTP, email and web-browsing applications. This wireless FTP server lies in another BSS and is assumed to have access to the internet via WiMax backhaul. In the Simple telecentre only two profiles are defined namely; My_Data and My_Voice. The specific applications and profiles are configured according to the discussion of the hospital telecentre. The 802.11g (extended physical rate) protocol at a data rate of 2 Mbps is applied to all nodes.

In the Simple telecentre scenario voice is the only real-time application and is assigned the highest priority. Hence, the voice delay remains fairly constant regardless of the increasing network load as seen in Table 4.3.

Table 4.3: Simulation Results for the Simple telecentre

Year	Voice Delay (ms)	FTP Delay (ms)	Email Delay (ms)	Network Load (Kbps)
2006	81.08	555	173	272
2007	80.91	359	182	254
2008	80.57	280	135	261
2009	80.54	340	130	303
2010	80.63	784	164	400
2011	80.82	1499	164	547
2012	81.00	2877	207	821
2013	80.89	1935	207	594
2014	81.05	2829	243	714
2015	83.14	11660	325	1180

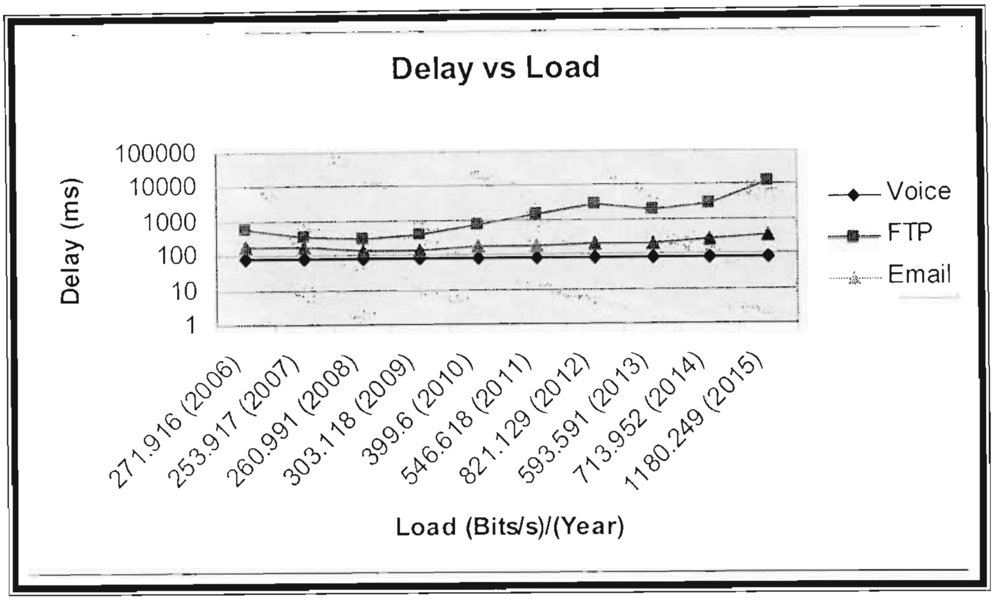

Figure 4.6: Graph of end-to-end delay plotted against network load for the Simple telecentre

From the simulation results the following observations were made; Voice is sustained over the 9 year period from 2006 until 2015 with reasonable QoS. The average voice packet end-to-end delay is maintained below the 100ms delay budget at 81ms. FTP and email services take the stress in a network when having to cater for high priority voice or real-time services. Whilst the delay for voice remains fairly constant, email and FTP delays increase significantly owing to their best-effort type of service. It is seen that since FTP has much larger file sizes as compared to email sizes (tabled in chapter 2), it experiences much higher delays.

4.6 Comparison of 802.11b and 802.11g protocol via Simulation

Details of the 802.11b and 802.11g protocol have been discussed in Chapter three. To evaluate the performance of the two protocols, simulations were run of the Simple telecentre scenario.

Table 4.4: Simulation results of the Simple telecentre using 802.11b protocol at 2Mbps

Year	Voice Delay (ms)	FTP Delay (ms)	Email Delay (ms)	Network Load (Kbps)	Throughput (Kbps)
2006	81.08	556	181	268.88	268.62
2007	80.92	343	178	250.77	250.49
2008	80.58	290	129	207.43	150.32
2009	80.54	423	124	291.46	154.89
2010	83.37	2229	417	344.33	344.22
2011	88.11	10157	1722	505.54	505.35
2012	90.07	7460	1208	519.32	519.08
2013	88.91	9508	490	513.13	513.87
2014	92.33	12230	1502	542.47	542.07
2015	98.45	28036	1327	714.91	713.85

Table 4.5: Simulation results of the Simple telecentre using 802.11g protocol at 2Mbps

Year	Voice Delay (ms)	FTP Delay (ms)	Email Delay (ms)	Network Load (Kbps)	Throughput (Kbps)
2006	81.08	554.72	173.08	271.92	271.779
2007	80.91	359.34	181.91	253.92	253.519
2008	80.57	280.16	135.00	260.99	260.02
2009	80.54	399.50	130.00	303.12	303.02
2010	80.63	784.00	164.00	399.60	399.28
2011	80.82	1499.00	164.05	546.62	541.11
2012	81.00	2876.60	207.30	821.13	820.36
2013	80.89	1935.30	207.30	593.59	593.61
2014	81.05	2829.00	243.12	713.95	713.08
2015	83.14	11660.00	325.00	1180.25	1180.25

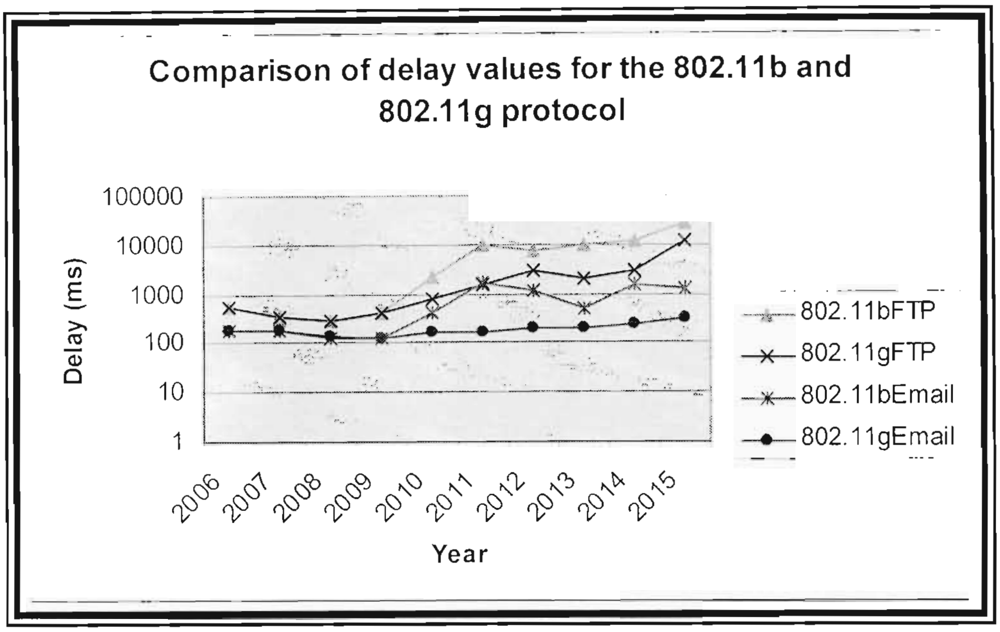

Figure 4.7: Comparison of 802.11b & 802.11g protocols evaluating FTP & Email Delays

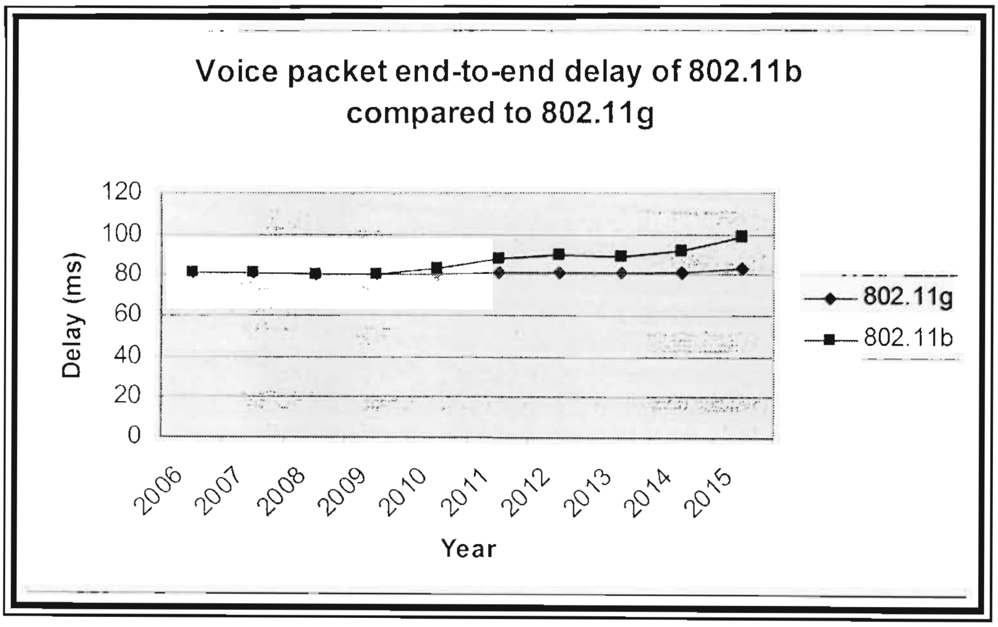

Figure 4.8: Voice packet end-to-end delay of 802.11b compared to 802.11g

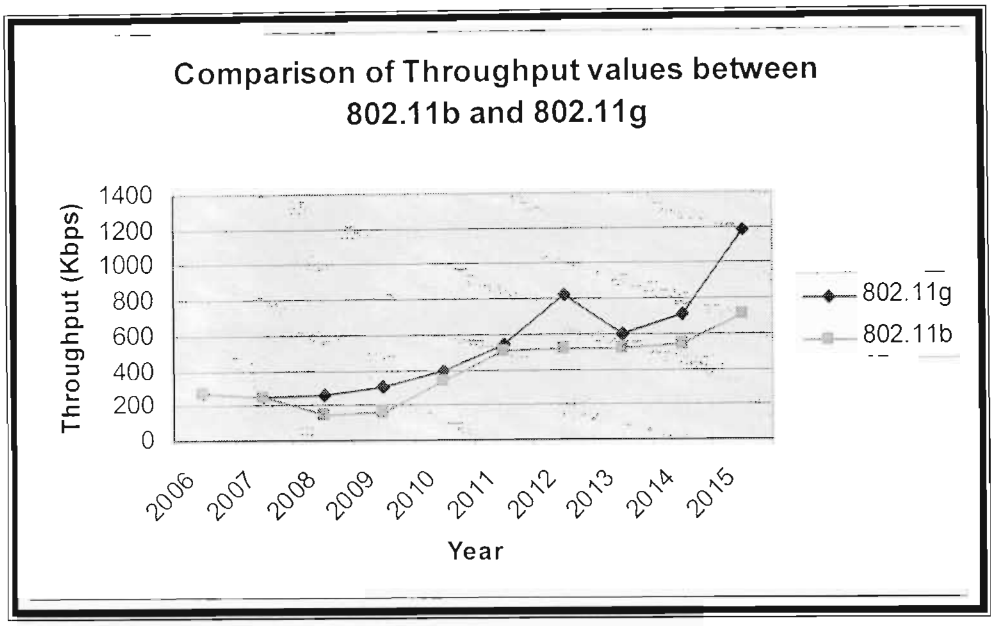

Figure 4.9: Comparison of throughput values between 802.11b and 802.11g

It can be observed from the above plots that 802.11g offers a higher throughput and lower delays as compared to 802.11b for higher loads. At low loads both protocols offer the same performance but beyond 2010 when the load on the network increases significantly the FTP, email and voice packet end-to-end delays for 802.11b increase substantially compared to 802.11g. The increased performance of 802.11g compared to 802.11b may be attributed to its MAC layer which has a preamble of 20us whilst in compatibility mode compared to 802.11b's long preamble of 192us. The minimum contention window (CWmin) for 802.11g is 15 time slots compared to 802.11b's 31 time slots, which ultimately reduces the delay. In DCF mode at low loads there is no contention in the network hence the delays remain the same for both 802.11b and 802.11g, but at high loads there is contention in the network and the back-off period for 802.11b will be longer than 802.11g due to its larger CWmin. In the Nongoma scenario 802.11g would be a more suitable protocol due to the requirement for higher throughput and lower delays to cater for real-time applications.

4.7 Voice Capacity of IEEE 802.11g Wireless LANs

This section evaluates the inherent limitations of the 802.11g protocol using DCF mode in supporting VoIP calls over a wireless LAN. Although PCF mode was designed specifically for isochronous traffic such as voice, it is specified as optional in the standard and seldom implemented in devices, furthermore previous studies [21] have indicated that PCF provides marginal improvement in performance when compared to DCF. 802.11g boasts a data rate of 54Mbps yet has an effective data rate of only 50% of this value. In principle, each VoIP stream typically requires less than 10Kbps and therefore an 802.11g WLAN operated at 27Mbps should be able to support more than 1350 VoIP sessions. In reality however only a few sessions can be supported due to various protocol overheads. Analytically, the upper bound on the number of simultaneous VoIP calls that can be placed in a single cell of an 802.11g network using CTS-to-self protection is derived. If one more VoIP call is added above the limit in that cell, the quality of all the VoIP calls will degrade. The Nongoma hospital, school and simple telecentre WLANs that were simulated above were configured using the 802.11g protocol in DCF mode. It is a pure 802.11g network, meaning that all nodes in the network are 802.11g; there are no 802.11b nodes active in the network. The CTS-to-self protection mechanism is enabled which allows the 802.11g nodes to hear if there is an 802.11b node in or entering the BSS. Thus it is important to establish the maximum number of simultaneous VoIP calls that can be sustained in that network.

The work done in this section is closely related to the work of Hole and Tobagi [22], where the capacity of an IEEE 802.11b network carrying voice calls in a wide range of scenarios were evaluated. The research of [22] considered both G.711 and G.729 voice encoding schemes and a range of voice packet sizes. An analytical upper bound was presented and using simulation it was shown to be tight in scenarios where channel quality is good and delay constraints are weak or absent. It was also shown that capacity is highly sensitive to the delay budget allocated to packetisation and wireless network delays. Hole and Tobagi derived the upper bound on the value of *N*, which is the number of calls in progress as:

$$N = \{1 / R\ [2(T_{VOICE} + SIFS + T_{ACK} + DIFS) + (T_{SLOT}\ X\ CW_{MIN} / 2)]\} \qquad [4.1]$$

In the above equation, R is the number of packets generated by each encoder per second. T_{SLOT} is the slot duration, and CW is the contention window. T_{ACK} is the duration of the acknowledgement frame, SIFS is the short inter-frame spacing and DIFS is the DCF inter-frame spacing. These parameters have been discussed in Chapter three. Values for the above parameters are found in Table 4.6 and Table 4.7.

Table 4.6: Parameter values of 802.11b and 802.11g

	802.11b	**802.11g**	
		802.11g only	**802.11b compatible**
DIFS	50μs	28μs	50μs
SIFS	10μs	10μs	10μs
Slot Time	20μs	9μs	20μs
CWmin	31	15	15
RTS	14 bytes	14 bytes	14 bytes
CTS	14 bytes	14 bytes	14 bytes
ACK Frame	14 bytes	14 bytes	14 bytes
Physical Layer Header Length	**192μs**	20μs	20μs
Signal Extension	**N/A**	6μs	6μs

Table 4.7: Component Times of T_{VOICE} & T_{ACK} for 802.11b at 11Mbps & 802.11g at 27Mbps

		802.11b (11Mbps)	**802.11g (27Mbps)**
Tvoice	PLCP Preamble & Header	192μs	26μs
	MAC Header & FCS	20.4μs	8.3μs
	IP/UDP/RTP header	29.1μs	11.85μs
	Voice Data	(Voice octets x 8/11)μs	(Voice octets x 8/27)μs
Tack	PLCP Preamble & Header	192μs	26μs
	ACK Frame	10.2μs	4.15μs

Hole & Tobagi assessed the tightness of this upper bound by comparing the value of the upper bound with the capacity obtained by simulation. Their results are shown in Table 4.8 (the calculated upper bound is in parentheses). The ns-2 network simulator was used in [22] for the simulations.

Table 4.8: Capacity results from simulation (analysis) [22]

	Voice Data per frame			
	10ms	20ms	30ms	50ms
G.711	6(6)	12(12)	17(18)	25(26)
G.729	7(7)	14(14)	21(22)	34(35)

To verify these results, the exact scenario and settings were duplicated in the OPNET Modeler 11.5 simulation package and the following results were obtained. Firstly the G711 codec with a 10ms and 20ms frame sizes was considered and thereafter a G.729 codec with a 10ms and 20ms frame size was evaluated to compare with the results from [22].

Table 4.9: Results of a G711 codec with a 10ms frame size

# Phones	**Voice Traffic Sent (Kbps)**	**Voice Traffic Received (Kbps)**	**Delay (ms)**	**WLAN Load (Kbps)**	**WLAN Throughput (Kbps)**
4	503	503	20	768	768
5	640	640	21	960	960
6	752	733	232	1152	1121
7	896	752	235	1344	1147

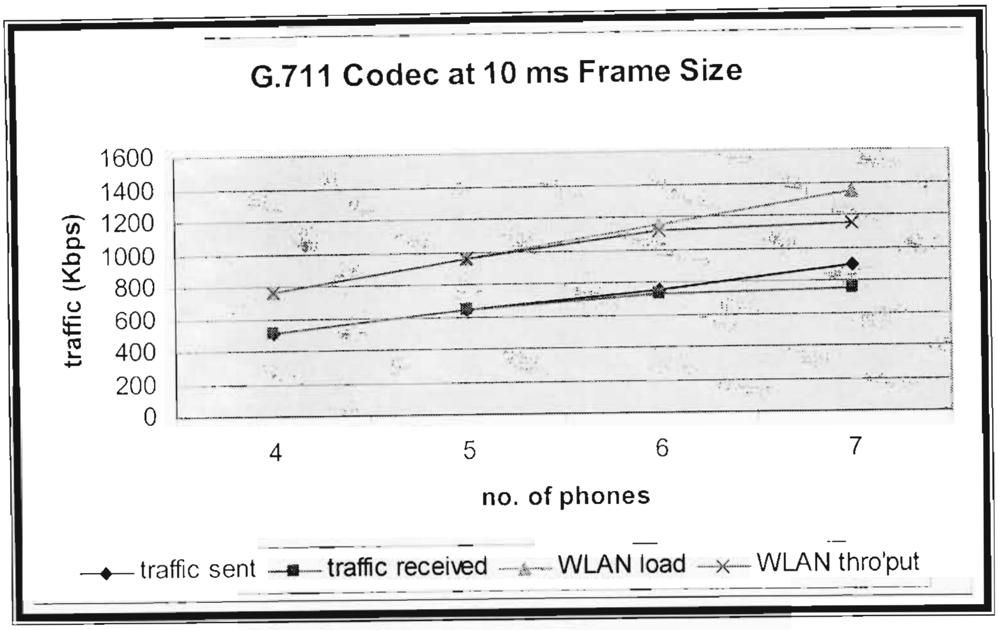

Figure 4.10: Throughput of the voice payload using a G.711 codec with a 10ms frame size

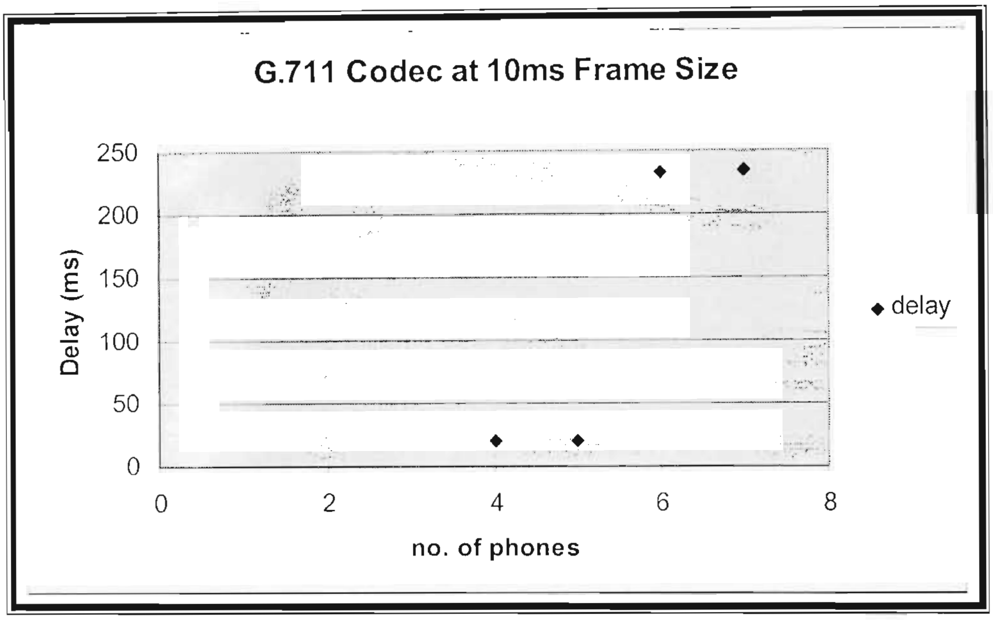

Figure 4.11: The delay increase for a G.711 codec with a 10ms frame size

It is observed that when 6 VoIP phones are added to the network, the delay increases sharply and there is a slight drop in throughput. As mentioned in [22] selecting the optimum amount of voice data per packet is shown to be a trade-off between throughput and delay constraints. The value of 6 VoIP phones corresponds to the value obtained by Hole & Tobagi as the maximum capacity for voice calls using a G.711 codec with a 10ms frame size. It can also be observed that as the number of phones is increased, the amount of bandwidth required increases. A single VoIP stream typically requires approximately 10 Kbps and an 802.11b WLAN operated at 11 Mbps could in principle support approximately 550 VoIP sessions. In actuality, no more than a few sessions can be supported due to various protocol overheads. The 802.11 MAC/PHY layers have additional overhead of more than 800 μsec, attributed to the physical preamble, MAC header, MAC backoff time, MAC acknowledgement and inter-transmission times of packets and acknowledgements [38].

Table 4.10: Results of a G.711 codec using a 20ms frame size

# Phones	Voice traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput (Kbps)
11	1408	1395.842	168.33	1760	1744.809
12	1536	1409.646	178.23	1920	1762.057
13	1664	1415.517	177.69	2080	1769.401

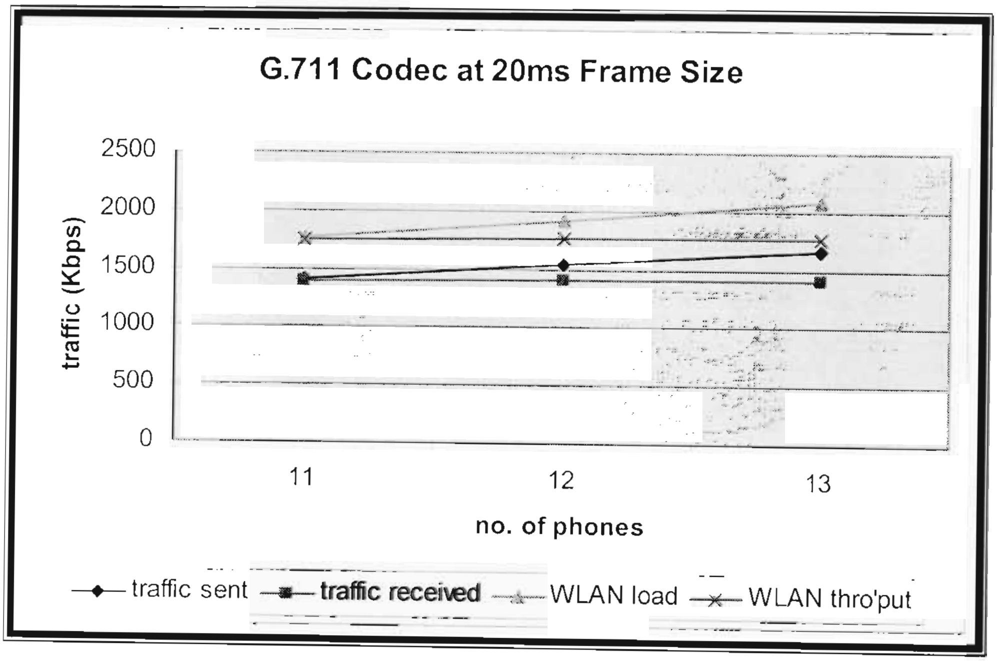

Figure 4.12: Throughput of the voice payload using a G.711 codec with a 20ms frame size

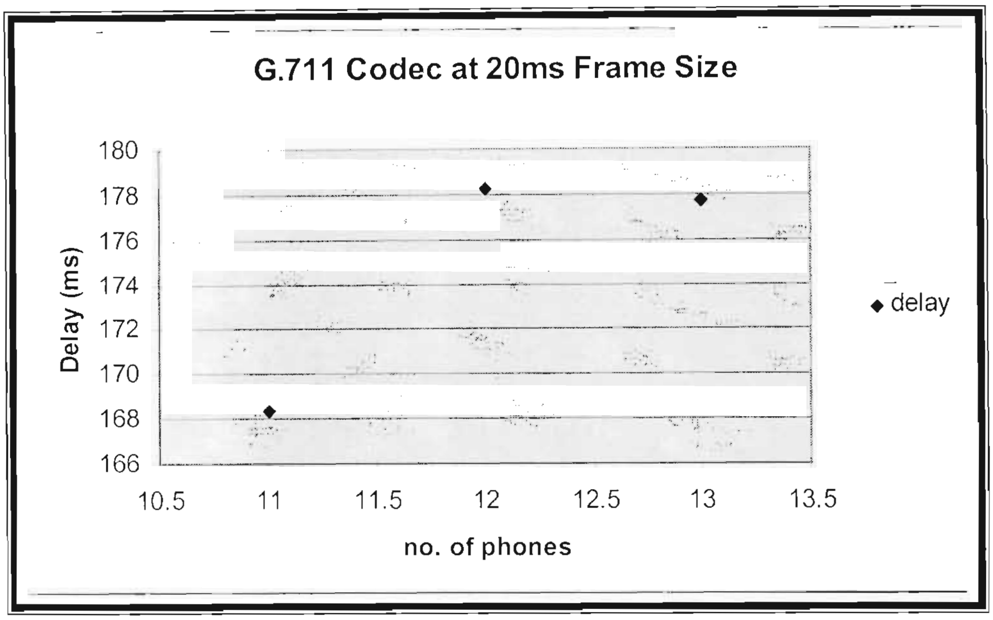

Figure 4.13: The delay increase for a G.711 codec with a 20ms frame size

Figure 4.13 shows that 12 VoIP phones is the maximum voice capacity which again matches the results in [22]. The delay increases rapidly when 12 VoIP phones are placed in the network and the throughput starts to decrease at this same point.

Table 4.11: Results of a G.729 codec using a 10ms frame size

# Phones	Voice Traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput (Kbps)
6	96	96	21.66	480	480
7	112	101.244	499.51	560	506.218
8	128	102.733	496.28	640	513.668

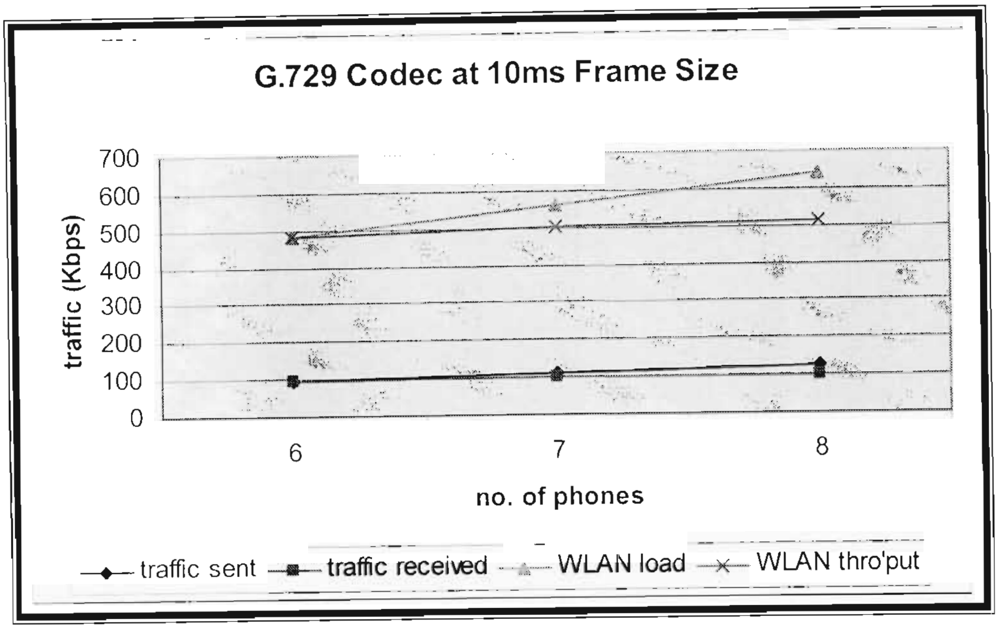

Figure 4.14: Throughput of the voice payload using a G.729 codec with a 10ms frame size

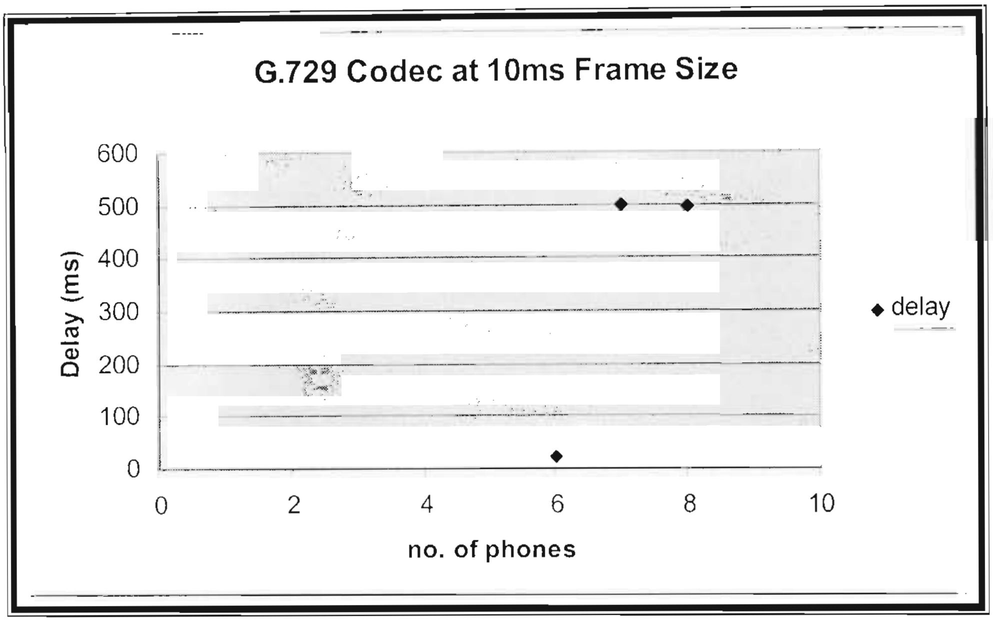

Figure 4.15: The delay increase for a G.729 codec with a 10ms frame size

As observed similar to the above scenarios, when 7 VoIP phones are added to the network the delay increases drastically and there is a drop in throughput. To maintain good QoS on all VoIP calls, 6 VoIP phones are determined as the maximum capacity.

Table 4.12: Results of a G.729 codec using a 20ms frame size

# Phones	Voice Traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput
12	192	191.992	42.73	576	575.977
13	208	197.674	460.7	624	593.022
14	224	200.414	440.82	672	601.239
15	240	201.465	463.8	720	604.395

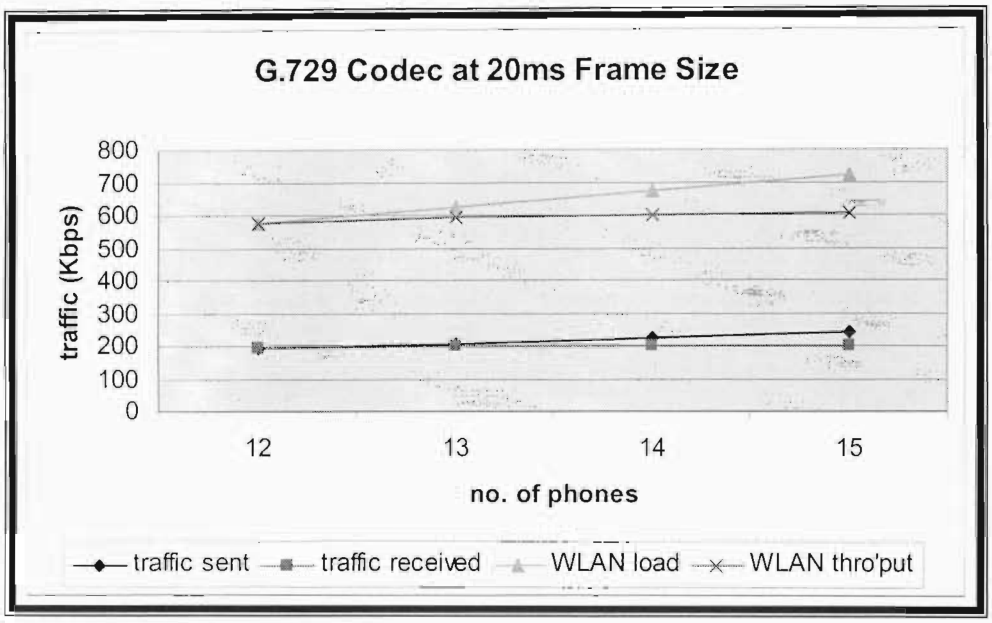

Figure 4.16: Throughput of the voice payload using a G.729 codec with a 20ms frame size

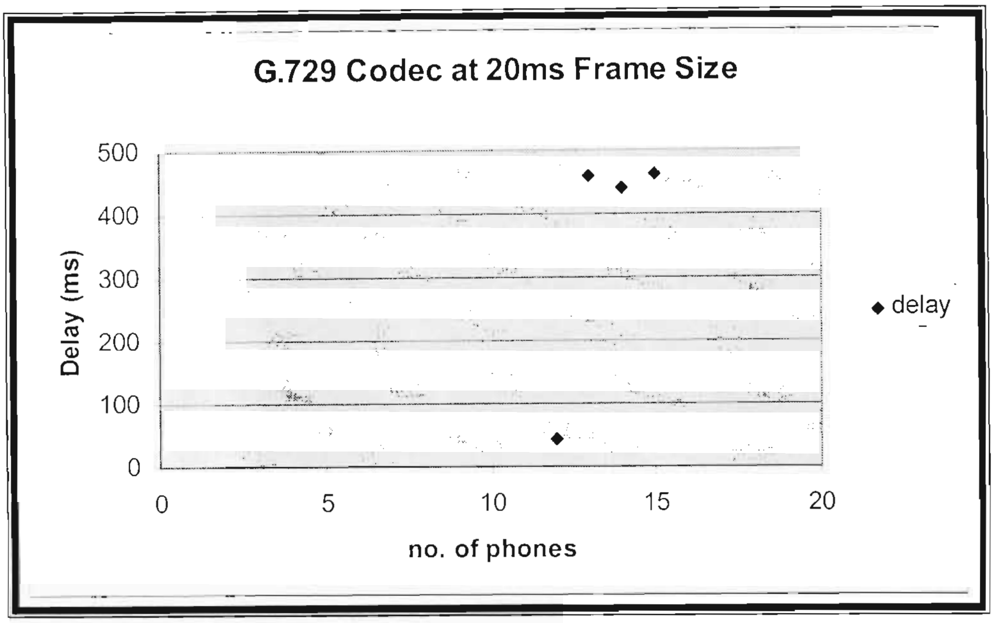

Figure 4.17: The delay increase for a G.729 codec with a 20ms frame size

From the above results, one can conclude that the maximum capacity is reached at 13 VoIP phones, bearing in mind the throughput and delay constraints to maintain good QoS.

Thus Hole & Tobagi's results have been verified using the OPNET simulator. It should also be noted that unless a very high voice quality requirement precludes its use, the G.729 is shown to allow a capacity greater than that of the G.711 codec, for a given quality requirement.

In this research, equation 4.1 is modified to establish the upper bound for an 802.11g network using the CTS-to-self protection mechanism as used in the Nongoma telecentre scenario. From equation 4.1, the parameter T_{CTS} which is the CTS duration is included in the calculation of the total transaction time in equation 4.2.

$N = \{1 / R\ [2(T_{VOICE} + SIFS + T_{ACK} + DIFS + T_{CTS}) + (T_{SLOT}\ X\ CW_{MIN} / 2)]\}$ [4.2]

The values for the parameters in equation 4.2 can be found in Table 4.6 (802.11g/ 802.11b compatible) and Table 4.7 (802.11g 27Mbps). 27Mbps is used as the effective data rate for 802.11g which is 50% of the actual data rate of 54 Mbps. The CTS and ACK frame size are both 14 bytes, thus;

$T_{CTS} = T_{ACK}$ = preamble + signal extension for 802.11g + CTS/ACK frame = $20\mu s + 6\mu s + (14 \times 8 / 27 \times 10^6) = 30.15\mu s$.

The backoff time = (T_{SLOT} X CW_{MIN} / 2) = (20μs x 15)/2 = 150μs.

An example of the calculation for the component times of T_{VOICE} are as follows:

The duration for a 28 byte MAC header & FCS = 28 x 8/ 27x10^6 = 8.3μs.

The duration for the 8 byte IP, 20 byte UDP and 12 byte RTP headers = 40 x 8/ 27x10^6 = 11.85μs.

The PLCP preamble = Physical Layer Header Length + Signal Extension = 20μs + 6μs = 26μs.

Equation 4.2 is used to calculate the maximum number of VoIP connections a single 802.11g access point can support. Three standard codecs were evaluated, namely; G.711, G.729 and G.723.1. The G.711 codec operates at 64Kbps and transmits 10ms of audio data in 80 bytes of payload with no compression. The G.729 codec compresses 10ms of audio in 10 bytes of payload at 8Kbps. The G.723.1 codec compresses 30ms of audio data in 24 bytes of payload at 6.4Kbps. Table 4.13 tabulates the maximum number of VoIP connections for the three codecs.

Table 4.13: Maximum number of VoIP connections for various codecs

Frame Size	G.711	G.729	G.723.1
10	18	20	
20	34	40	
30	47	59	60
40	59	78	
50	69	97	
60	79	118	120
70	87	136	
80	94	154	
90	98	167	171

The voice payload is increased by increments of 10ms for the G.711 and G.729 codecs whilst it is varied in increments of 30ms for the G.723.1 codec. The VoIP call capacity increases as the payload size increases. One voice frame per packet implies as the frame size is increased the packet size increases. The number of VoIP connections increase with frame size due to the decreased overhead per payload. For example for a 10ms payload size, the packet size will be x and for a 20ms frame size the packet size will be 2x and so forth. The overhead will decrease as the frame size increases. The overhead comprises of packet headers from various networking layers and the backoff and deferral time imposed by the DCF. Thus the capacity of the WLAN depends on the size of packets comprising the load. As a result of this overhead the achievable throughput for 802.11g is far less than its maximal 54Mbps data rate it currently supports. It is important to note, as mentioned in [24], that the most commercial implementations of IP phones use a payload size of either 20ms or 30ms voice payload in each RTP packet. The G.711 codec has a maximum call capacity of 34 calls for a 20ms frame size and 47 calls for a 30ms frame size. Since VoIP calls traverse both wireless and wired networks, the larger the payload, the worse are the delay, loss and jitter characteristics adversely affecting the VoIP call quality [24]. The G.729 is the obvious codec of choice as it uses less compression than the G.723.1 and offers a higher capacity than the G.711.

The G.729 codec is used in the Nongoma telecentres. Hence the calculated G.729 maximum VoIP capacity shall be verified via simulation using the OPNET modeler. The G.729 codec is set to transmit various frame sizes at a rate of 1 voice frame per packet. For example the voice data and R parameters from equation 4.2 is calculated for a 10ms frame size as shown:

Voice Data = $10 \times 8/ 27\times10^{6} = 2.96\mu s$

R (G.729) = 8000bps/ 10x8bps = 100pps

Thus the calculated results for the G.729 codec at frame sizes of 10ms, 20ms, and 30ms are evaluated by way of simulation as shown below. The smaller frame sizes are selected due to the fact that the longer voice payload sizes add to the overall delay. Another important point to note is that when large voice payload sizes are lost, it is more difficult to conceal by the speech decoder as compared to smaller voice payload sizes.

Table 4.14: Results of the G.729 codec with a 10ms frame size

# Phones	Voice Traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput (Kbps)
20	320	320	21.5	1600	1600
21	336	331.371	168.88	1680	1656.849
22	352	330.414	174.49	1760	1652.069

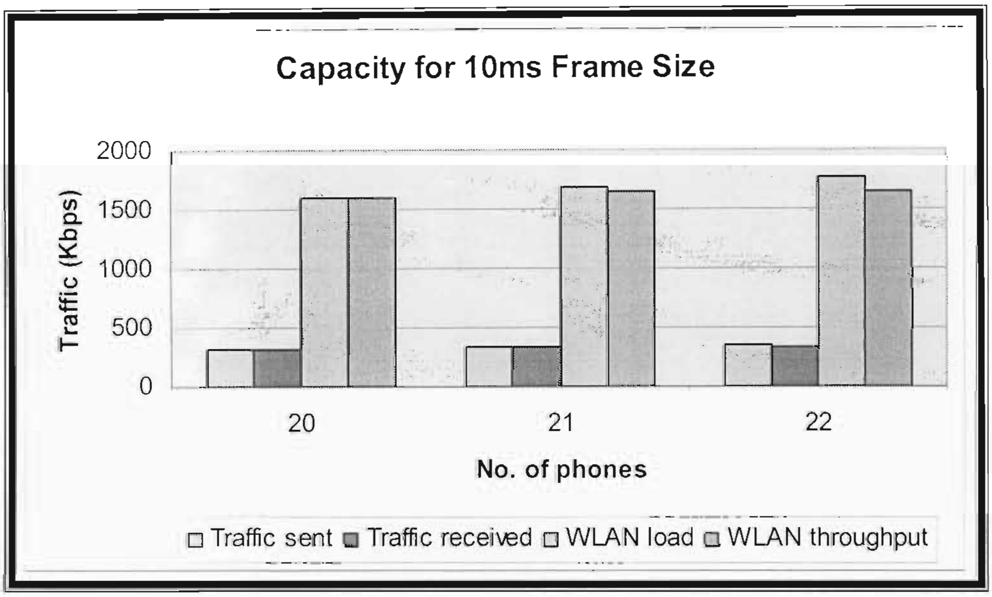

Figure 4.18: Voice capacity for 10ms frame size

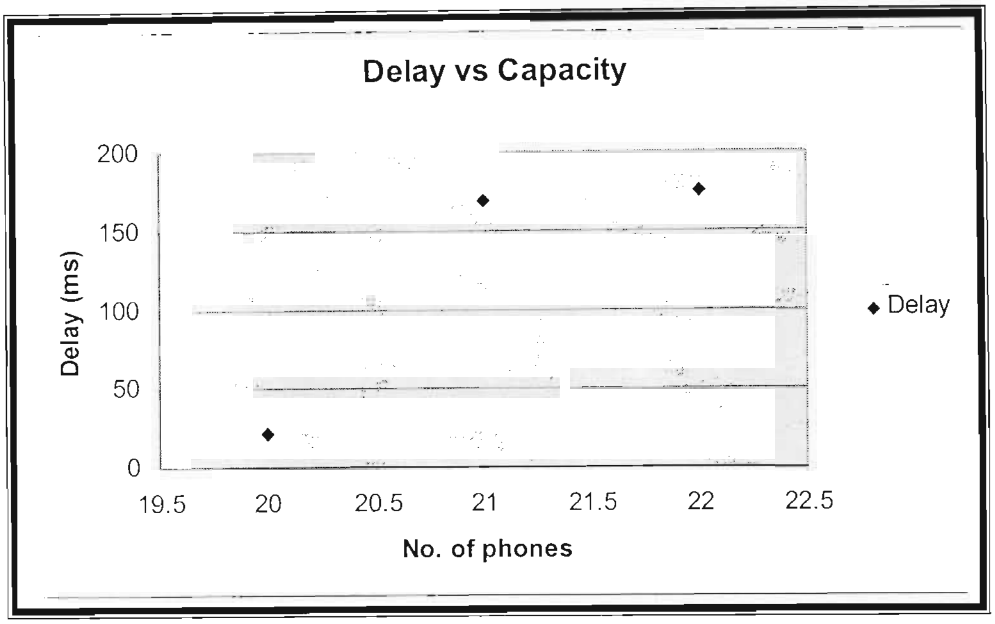

Figure 4.19: Delay analysis for a 10ms voice frame size

Figure 4.18 shows the throughput of the voice payload. It is easy to observe that there is 100% throughput when 20 VoIP phones are connected and packets start to get dropped as soon as one more phone is added to the network. Figure 4.18 compares the WLAN Load at the access point to the WLAN throughput and also displays the voice payload throughput again relative to it. The WLAN throughput begins to drop when 21 VoIP phones are connected. Figure 4.19 shows the delay constraint. The delay with 20 VoIP phones of 21.5ms shoots up to 168.88ms when another phone is added. 168.88ms exceeds the round trip delay budget of 150ms. Hence the deterioration of throughput and delay values severely impact the QoS of a VoIP connection, implying 20 VoIP phones is given as the maximum capacity of a VoIP network maintaining good QoS.

Table 4.15: Results of the G.729 codec with a 20ms frame size

# Phones	Voice Traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput (Kbps)
38	608	607.986	41.48	1824	1823.958
39	624	623.772	42.13	1872	1871.307
40	640	639.905	43.2	1920	1919.712
41	656	647.877	162.08	1968	1943.622
42	672	644.626	169.78	2016	1933.878

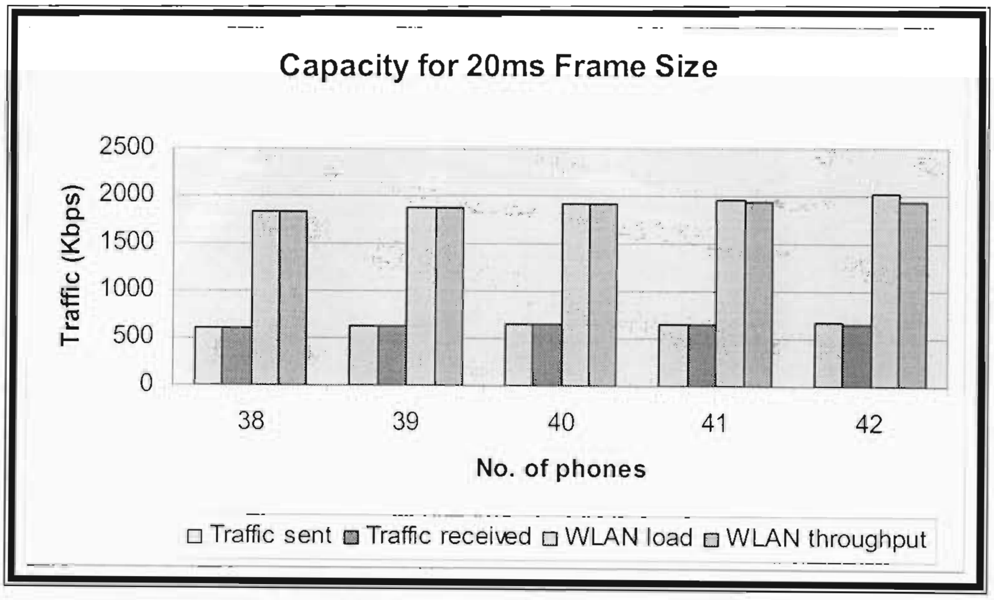

Figure 4.20: Voice capacity for 20ms frame size

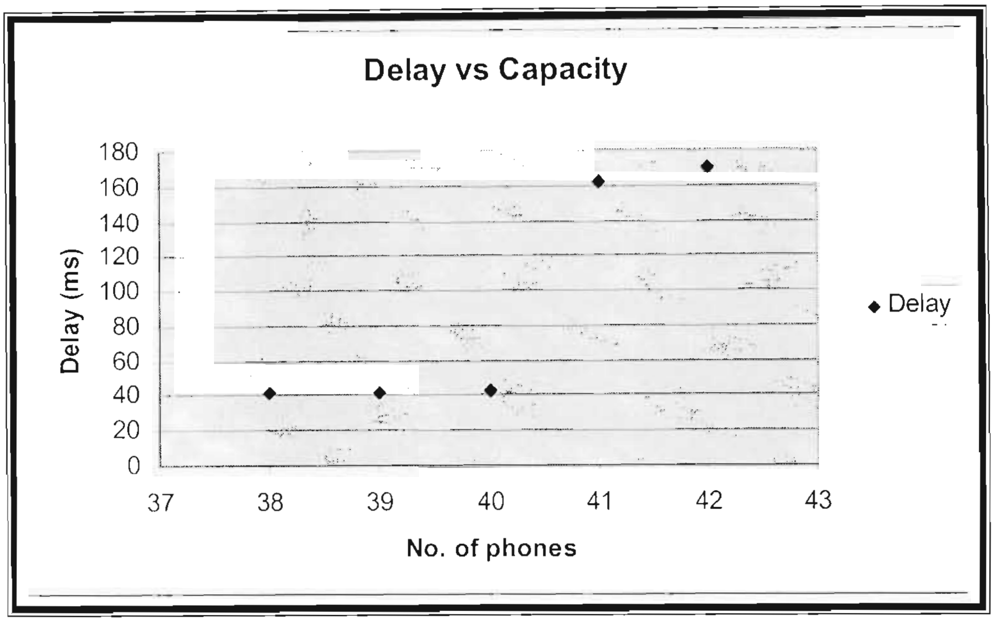

Figure 4.21: Delay analysis for a 20ms voice frame size

As observed from figure 4.20 the voice payload and WLAN throughput starts to degrade beyond 40 VoIP phones. Figure 4.21 shows the rapid increase beyond the delay budget of 150ms when 41 VoIP phones are connected in the network. Thus 40 VoIP phones are assumed to be the maximum capacity for a network using a frame size of 20ms.

Table 4.16: Results of the G.729 codec with a 30ms frame size

# Phones	Voice Traffic Sent (Kbps)	Voice Traffic Received (Kbps)	Delay (ms)	WLAN Load (Kbps)	WLAN Throughput (Kbps)
59	**944**	944	**63.78**	2202.667	2202.667
60	**960**	951.852	**165.78**	2240	2220.987
61	**976**	962.428	**169.88**	2277.333	2245.666
62	**992**	960.781	**172.33**	**2314.667**	2241.822

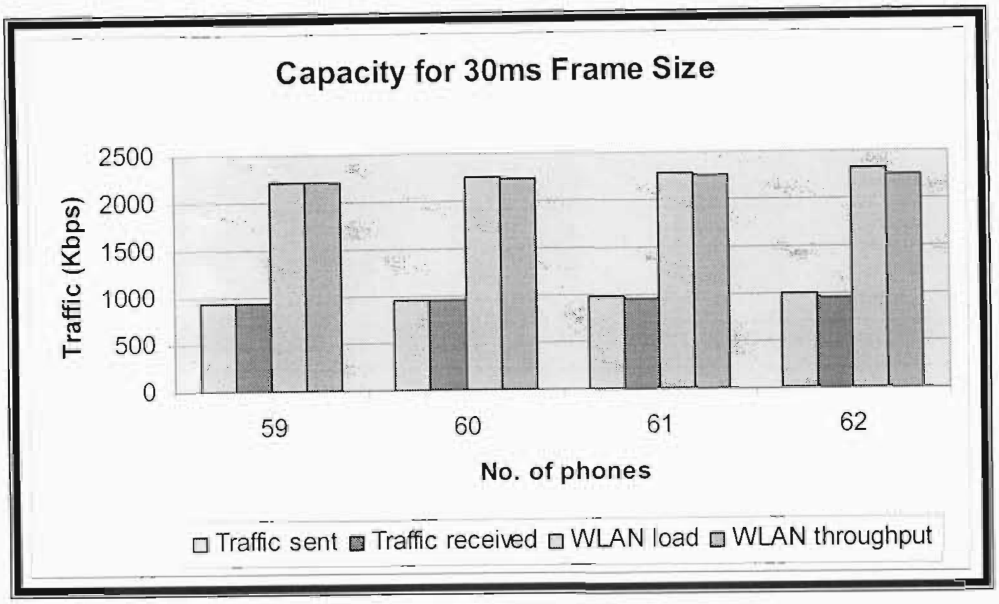

Figure 4.22: Voice capacity for 30ms frame size

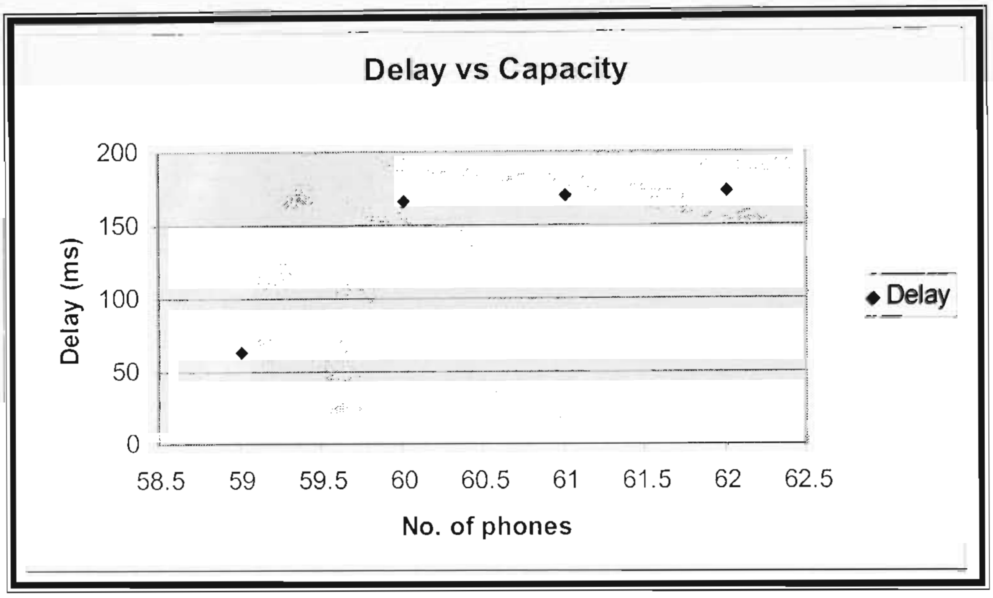

Figure 4.23: Delay analysis for a 30ms voice frame size

Figure 4.22 demonstrates that from 60 VoIP phones onwards the voice traffic received and WLAN throughput starts to decrease. It is also noted from figure 4.23 that the delay increases significantly when there are 60 VoIP phones connected. 59 VoIP phones are selected as the maximum capacity for a VoIP network operating in 802.11g DCF mode with CTS-to-self protection using a 30ms voice frame size. 59 VoIP phones exhibit a 100% throughput and a delay below the end-to-end delay budget of 150ms which maintains a good QoS.

Table 4.17 presents a summary of the analytical and simulation results for the conversational speech capacity of WLANs using a G.729 codec.

Table 4.17: Maximum Capacity of a VoIP network using a G.729 Codec with various frame sizes

Frame Size (ms)	**Maximum Capacity (# of VoIP calls)**	
	Analytic	Simulation (Delay)
10	20	20 (21.5ms)
20	40	40 (43.2ms)
30	59	59 (63.78ms)

The analytical results are in very close agreement with those obtained via simulation. The tightness of the analytical upper bound is demonstrated in the simulations due to the delay constraints. The appropriate choice for a VoIP network should be the G.729 Codec using a 20ms frame size. As the frame size is increased the overall delay increases. From the simulation results it is observed that the delay value is approximately equal to twice the frame size. In the Nongoma telecentres the delay for a 20ms frame size in the presence of data traffic was found to be approximately 80ms which is approximately twice the delay in a pure VoIP network. The delay value for a 50ms frame size was obtained by experiment to be 103.95ms. In the presence of data traffic this delay value will increase rendering poor QoS, hence smaller frame sizes are selected over larger frame sizes and a higher capacity.

In the Nongoma telecentre scenarios, the use of a G.729 codec with a voice frame size of 20ms gives a maximum capacity of 40 VoIP simultaneous sessions. This value however will be further reduced in the presence of data traffic [21] and [24].

4.8 Summary

The OPNET Modeler version 11.5 simulation package was used to model the telecentres. A thorough explanation of the simulation scenarios for the various telecentres was presented along with the simulation results. The traffic placed on the simulated network was in accordance with the traffic estimates predicted in chapter two. Initial dimensioning in chapter two was done for an 8 year period from 2007 until 2014. In all the telecentres voice data is sustained with good QoS over this period with data traffic on the network. The hospital and school telecentres were then heavily loaded following traffic trends for the consequent years to establish at which point the network will fail to sustain voice applications with reasonable QoS. An evaluation of the performance of 802.11b and 802.11g was done via simulation to verify if the selection to use the 802.11g protocol in the Nongoma telecentres was adequate. The 802.11g protocol offers better throughput and lower delays for high loads. The choice of protocol is therefore partially dependent on the type of applications that will be provided on the network. For high data rate applications such as videoconferencing, 802.11g would be a better protocol to use. To accommodate for the growing demand of VoIP applications in rural regions the research proposes a solution of providing larger telecentres to provide mainly VoIP applications. Based on the substantial work of Hole & Tobagi [22] for the 802.11b protocol, in this study we have been able to extend the same concepts to calculate the maximum capacity of an 802.11g with CTS-to-self VoIP network using a G.729 codec with various frame sizes.

CHAPTER FIVE: WIRELESS MESH NETWORKS

5.1 Introduction

This chapter discusses the use of Mesh technology as a strategy to extend coverage and hence provide cost effective rural services. Wireless Mesh Networks (WMNs) provide a reliable & cost effective solution to extending coverage in a rural fixed network. The research investigates range extension using a hybrid WLAN architecture running both infrastructure & client wireless mesh networks. Chapter four of the research investigated the use of the 802.11g protocol in a rural WLAN. This chapter serves to extend the use of the 802.11g protocol in a WMN for a rural application.

5.2 Wireless Mesh Background

The mesh architecture has no central service provider. Mesh mode is relatively inexpensive, very reliable and resilient. In the mesh infrastructure mode, a node only transmits data as far as to the next node. Data transmitted from a node at one end is relayed from one node to the next until it reaches its destination node at the far end, thus extending the coverage range of the network, especially over rough or difficult terrain. Hence it can be said that the nodes behave as repeaters. Mesh networks may involve either fixed or mobile devices.

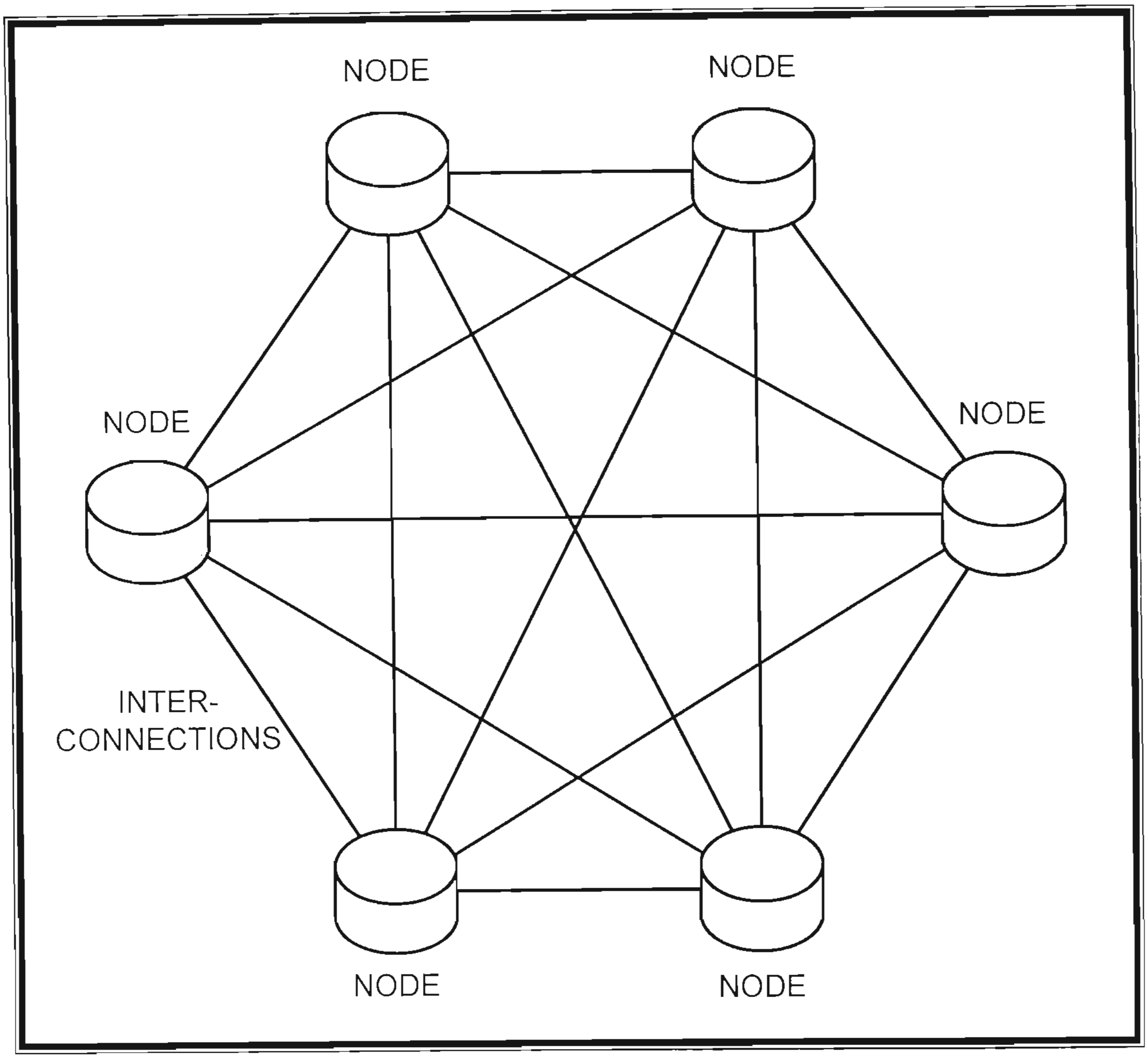

Figure 5.1: Conceptual Mesh Topology Showing the Complexity of Physical (or wireless/virtual) interconnections among network nodes

The conceptual mesh topology shown in figure 5.1 exhibits a fully connected network, which typically would improve overall performance and resiliency. Implementing a wired network with a mesh topology will be highly impractical or even impossible due to the high complexity and added cost of wires. However a wireless mesh network is very simple to implement and operate due to the fact that when a node is switched on it automatically connects itself to the network, senses its environment and detects all its neighbour nodes. It constantly senses the network at regular intervals to detect if any new nodes have joined the network. The hassles of moving, adding or removing nodes and their attached Ethernet devices are eliminated, making it quick and easy. The routing tables are constantly updated as the routing paths in the network change. The most optimal path is chosen based on the shortest or quickest path with minimal traffic. If a node fails or a specific link becomes congested the mesh network automatically redirects traffic on an alternate path. The multiple redundant paths of the mesh topology add resiliency and when properly arranged, eliminate single points of failure and potential bottlenecks within the mesh.

A wireless mesh network delivers scalable performance because it can be expanded easily and incrementally as needed. The overall performance and reliability of the mesh is improved when more nodes are included in the network. To maximize performance and reliability, each node should have at least two neighbors.

The Wireless Mesh takes advantage of the existing capability of the 802.11 standard in the Ad-Hoc mode of communications. Compared to the Infrastructure mode, the Ad-Hoc Mode uses much less overhead and thereby enables each node to function as a router capable of passing traffic to other nodes. The peer-to-peer approach provides a solid foundation for the high performance required in a mesh network. The implementation of multiple spectrums, specifically in the 2.4 GHz and 5 GHz bands in the 802.11 standard is innovatively used by mesh networks to provide optimal range and minimal interference. The 5 GHz spectrum is more suited to situations that require high capacity and low interference. The 2.4 GHz spectrum provides better overall coverage whilst dealing with obstacles such as walls and solid building materials.

Ad-Hoc Mode utilizes dynamic routing. To implement such dynamic routing capabilities, each device needs to communicate its routing information to every device it connects with, "almost in real time". This information is then analysed by each device from which it determines if it is the destination node and should keep the data it received or relay it along to the next device. The routing algorithm used should attempt to always ensure that the data takes the most appropriate (fastest) route to its destination.

Security is a major concern for wireless mesh networks. Traffic within the mesh must be secured and outside devices, including those that use the mesh's Ethernet services, should be prohibited from accessing internal mesh traffic. Security features like digital signatures, encryption and industry standards such as 128-bit and 256-bit AES encryption can provide very effective security comparable to the level of security provided by wired networks. The mesh should also be able to support other security standards available on other Ethernet-based and wireless networks.

To implement QoS in a wireless environment is quite complex. Various architectures have been adopted to facilitate end-to-end QoS provisions. One such design consideration is the ability to support standard Ethernet Class of Service (CoS) tagging to prioritize traffic. Support for CoS will allow, for example, Video and VoIP traffic to take priority over data traffic to achieve the low latency required [25].

When more nodes are added to the mesh the bandwidth is increased. This also potentially adds more hops to certain traffic flows. It is, therefore, critical that each node add a minimal amount of latency, ideally below 3-4 milliseconds [25].

A mesh network is a networking technique which allows inexpensive peer network nodes to supply back haul services to other nodes in the same network. It effectively extends a network by sharing access to higher cost network infrastructure. Mesh networks differ from other networks in that the component parts can all connect to each other via multiple hops, and they generally are not mobile.

In a rural scenario we are faced with the problem of low density and low concentration of population. There is a lack of infrastructure and services such as electricity in many parts. This is further aggravated by accessibility issues. Conventional technologies are costly and appear impractical to implement whereas wireless mesh networks provide a reliable and cost effective solution.

IEEE 802.11s working group is developing a mesh networking extension for WiFi networks. IEEE 802.11s Extended Service Set (ESS) Mesh Networking standard, will use mesh networking techniques to extend the range of wireless LANs securely and reliably [26].

5.3 Hybrid Wireless Mesh Network (HWMN)

The HWMN architecture is the combination of infrastructure and client meshing. Mesh clients can access the network through mesh routers as well as directly meshing with other mesh clients.

While the infrastructure provides connectivity to other networks such as the Internet, WiFi, WiMax, cellular, and sensor networks; the routing capabilities of clients provide improved connectivity and coverage around the WMN.

The hybrid architecture will be most applicable to the rural solution.

The wireless backbone provides large coverage, connectivity, and robustness in the wireless domain.

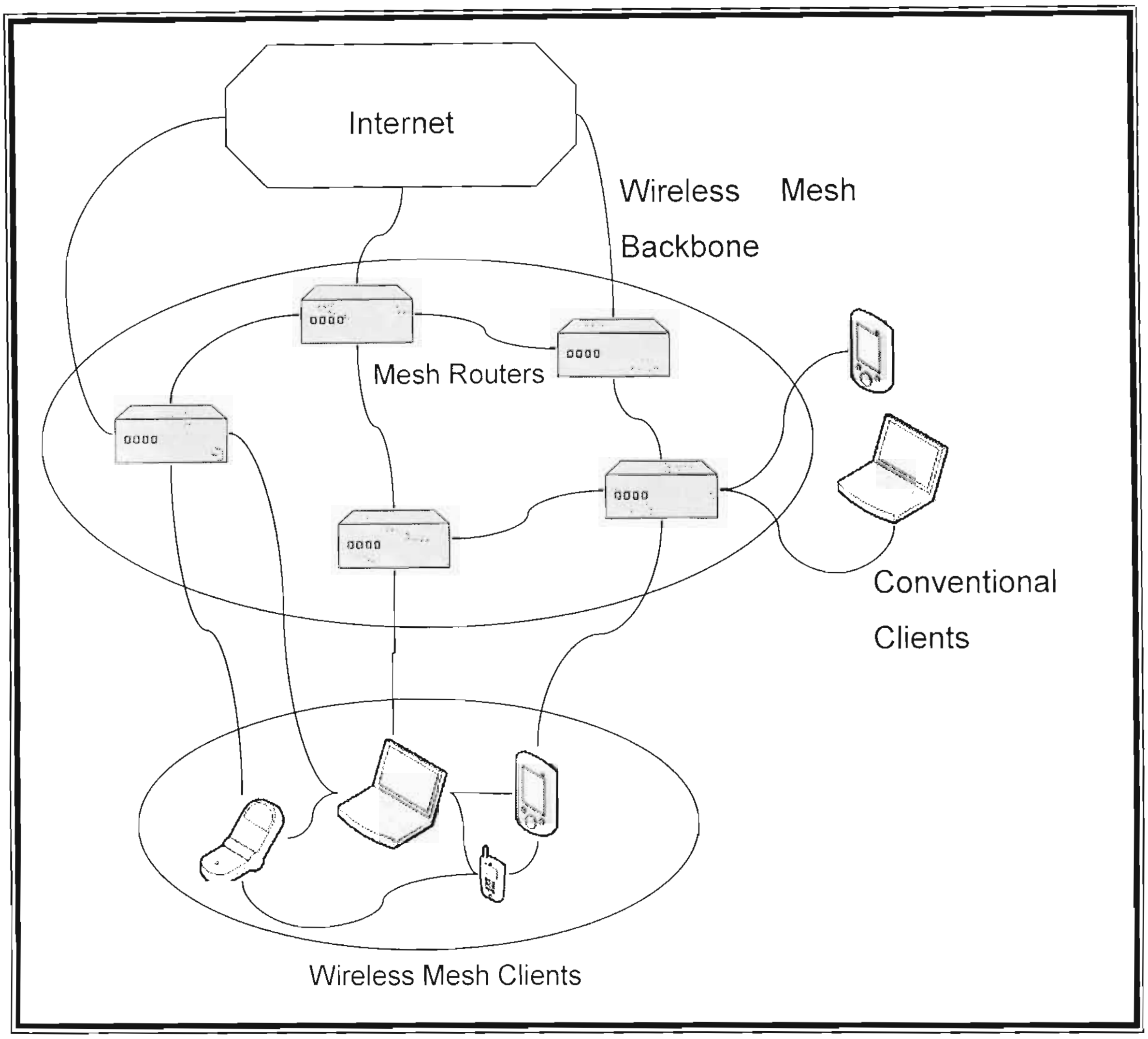

Figure 5.2: Hybrid Wireless Mesh Network

5.4 Mesh Rural Scenario

The basic range for an 802.11b or 802.11g protocol is approximately 300m. Previously, in chapter 4, the range was extended to 600m using a WLAN backbone configuration for an indoor environment. Here the research proposes to extend the WiFi range (outdoor) to the neighbourhood using mesh technology.

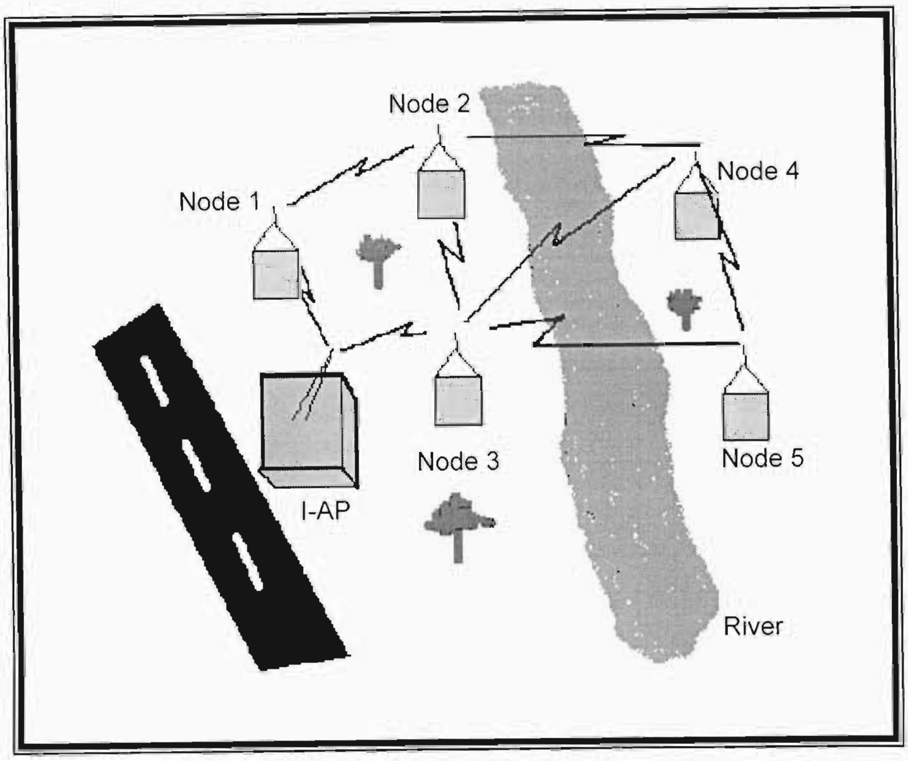

Figure 5.3: A wireless mesh network composed of WLAN APs connected by wireless links.

A cost-effective, rural solution is typically characterized by a small number of access points, a large coverage area and the use of a wireless backhaul. The key challenge is to improve the AP coverage area. For the Nongoma case study a fixed wireless network topology similar to the scenario as seen in figure 5.3 is proposed. The nodes are connected with each other via radio links operating on different channels. As long as a node is connected to the Infrastructure AP (I-AP) which is connected to the Internet via WiMax back haul, the network can provide Internet access to the rest of the nodes by means of multiple hops. It is typically not necessary for a user node to be in the coverage range of an AP to reach the outside network. In this scenario, local calls is a benefit in terms of cost, geography, etc. Hence fewer APs and associated backhaul links are required which reduces the cost of the network and also increases the coverage area. In the Figure 5.3 the I-AP functions as the gateway to the Internet and the user traffic either goes to or from this centralized AP. In a multi-hop application it has been stated in [27] that spatial reuse provides no obvious throughput advantage due to centralized traffic.

According to Zhang & Wolff [27] it is found that multi-hop systems become capacity limited due to congestion on the hop closest to the Internet connection point, favouring the use of high-bandwidth but shorter range technology such as 802.11g. In [28] the nominal capacity of wireless mesh networks is determined based on the concept of the Bottleneck Collision Domain (BCD), where the throughput of each node decreases as $O(1/n)$, where n is the total number of nodes in the network. Zhang & Wolff [27] propose two models, a basic multi-hop cell model called a Star Mesh Network (SMN) that analyses the capacity limitations whilst applying the 'bottleneck collision domain' concept from [28] and a SMN model with sector antennas at the AP in order to mitigate the bottleneck problem and to improve the average throughput of each node.

This model is evaluated to determine if it is indeed suitable for a South African rural area. Thus this model is applied to the Nongoma scenario. As mentioned earlier, the focus is on Ward 19 of Nongoma. Since Ward 19 hosts the town of Nongoma, it has a key urban area and is surrounded by rural settlements and farms. The total population in Ward 19 is 3847. The land area is an estimated 1/100 of the total area of Nongoma which is approximately 21.84km^2. Hence the population density is about 176 person/km^2. From the demographic data there is on average approximately 7 persons per household resulting in a household density ρ_h = 25 households/km^2. Taking into account the percentage increase over the 8 year time period on which the study is based, the household density at the end of the 8 years will be about 28 households/km^2.

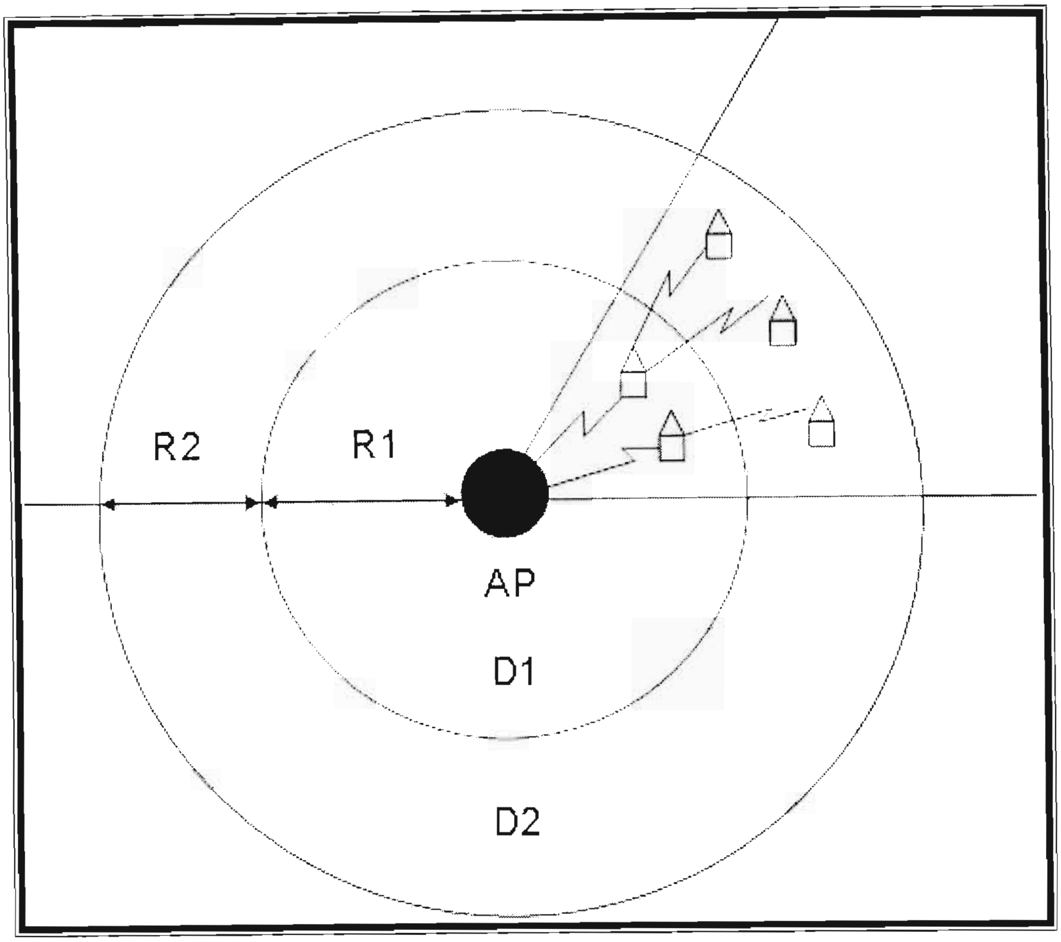

Figure 5.4: Star Mesh Network (SMN) model for the multi-hop WLAN [27]

For the households, the following Internet services will be provisioned for, namely VoIP, FTP, Email and Web-browsing. Taking into consideration the fact that increasing the path length i.e. the number of hops, increases the end-to-end delay and reduces the throughput of the flow, for the Nongoma scenario the mesh topology is limited to just two hops. The area is divided into two regions D_1 and D_2. R_1 is the maximum range between the AP to the first hop user and R_2 is the maximum distance from the first hop to the second hop. An assumption is made such that $R_2 = R$ which is the average distance between the neighbour user nodes determined by the user density ρ_h. We assume R=300m for the average hop distance. The area of region D_1 and D_2 are:

$$A_{D1} = \pi R_1^2 \tag{5.1}$$

$$A_{D2} = \pi(R^2 + 2RR_1) \tag{5.2}$$

The number of households in each region are:

$$\mathrm{N}_1 = \rho_h A_{D1}, \quad \mathrm{N}_1 = \rho_h A_{D2} \tag{5.3}$$

The average load offered by each user node is assumed to be G and the traffic on each link including the forwarded traffic and the originating traffic from each node is calculated. The total traffic load (L_T) on the links of the bottleneck collision domain is derived in [27], given by:

$$\mathrm{L}_T = (2\mathrm{N}_2 + N_1)G \tag{5.4}$$

According to the theorem presented in [28], the nominal MAC layer capacity, given by B, is the maximum capacity that can be forwarded by the bottleneck collision domain. Thus, $L_T \leq B$, and according to [4] the maximum average offered load G_{max} is related to B in a 2-hop network as follows:

$$\mathrm{G}_{max} \leq \frac{\mathrm{B}}{2\mathrm{N}_2 + \mathrm{N}_1} \tag{5.5}$$

Using the above equations and parameters the maximum range for the first hop of the Nongoma scenario was calculated. The maximum 802.11g MAC layer throughput was set at 27 Mbps for wireless links with a data rate of 54 Mbps as was calculated in chapter four.

Equations (5.1) to (5.5) were reduced to the following formula to solve for the maximum distance of the first hop.

$$2(\rho_h \pi(\mathrm{R}^2 + 2\mathrm{RR}_1)) + \rho_h \pi \mathrm{R}_1^2 \leq \frac{\mathrm{B}}{\mathrm{G}_{max}} \tag{5.6}$$

Let $\mathrm{K} = \frac{\mathrm{R}_1}{\mathrm{R}}$

$$2(\rho_h \pi(R^2 + 2R^2K)) + \rho_h \pi R^2 K^2 \leq \frac{B}{G_{max}} \quad (5.7)$$

An example calculation for R_{1max} at 2007 is shown below. R = 300 m; B = 27 Mbps;

$G_{max} = 100 Kbps$; $\rho_h = 25$

From (7), the values are plugged into the equation and it is simplified to:

$$\begin{aligned}
&7.07(2 + 4K + K^2 \leq 270 \\
&(K+2)^2 - 2 \leq 38.19 \\
&K + 2 \leq 6.34 \\
&K \leq 4.34
\end{aligned}$$

$$R_{1max} = K_{max} R = 4.34 * (0.3) = 1.302 Km$$

Similarly R_{1max} was calculated for the subsequent years in Table 5.1. The average traffic per node used in [27] is 250 Kbps, scaling this value to a South African rural scenario and referring to the traffic estimates in chapter two, the maximum traffic per node is estimated to be 100 Kbps and the percentage increases are predicted from the trends mentioned in chapter two.

Table 5.1: The calculated maximum first hop range, as the maximum offered load (G) per node and household densities increase over the years.

Year	ρ_h	G_{max}(Kbps)	% increase	R_{1max}(Km)
2007	25	100	0	1.302
2008	26	125	25	1.081
2009	26	150	20	0.945
2010	27	200	33	0.734
2011	27	250	25	0.608
2012	27	333	33	0.467
2013	27	400	20	0.390
2014	28	500	25	0.290

Table 5.2 looks at G_{max} based on ρ_h. Equation (6) was used where the R_1 values were fixed and G_{max} values were calculated for different ρ_h. An observation to be noted is that, for a larger household density the throughput per node decreases. This can be clearly seen in Figure 5.5.

Table 5.2: Calculated G_{max} for different ρ_h.

	$\rho_h = 25$	$\rho_h = 28$
R_{1max}(Km)	G_{max}(Kbps)	G_{max}(Kbps)
3	26.899	24.017
2.5	36.455	32.549
2.0	52.245	46.647
1.5	81.27	72.563
1.0	144.443	128.967
1.75	209.299	186.874
0.6	272.837	243.604
0.5	333.762	298.001

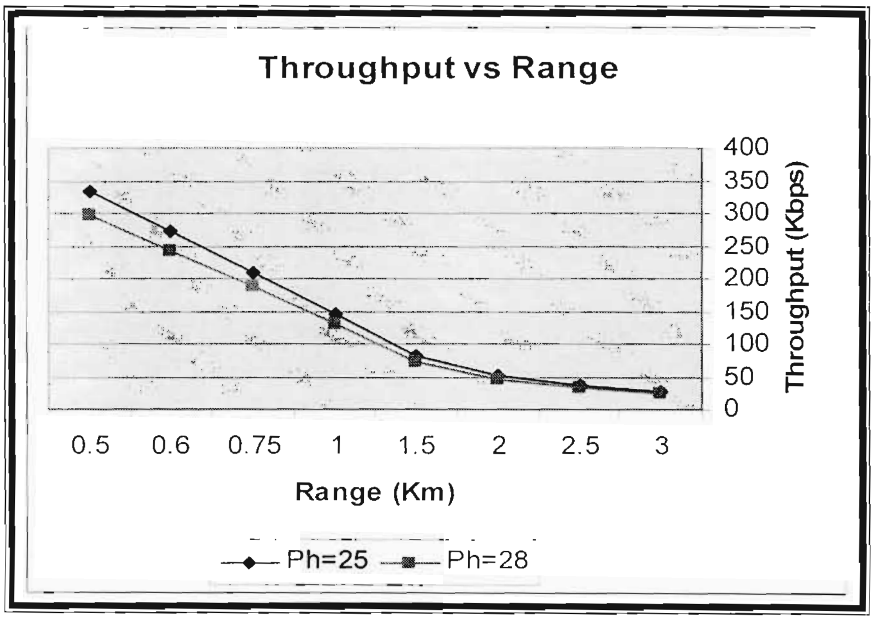

Figure 5.5: The relationship between ρ_h and G_{max}.

5.5 OPNET Simulation of Mesh Network

To investigate if this range is achievable with the estimated maximum throughput on each node as it increases exponentially over the years, simulations were run. An initial simple mesh scenario was set up using the OPNET MANET module.

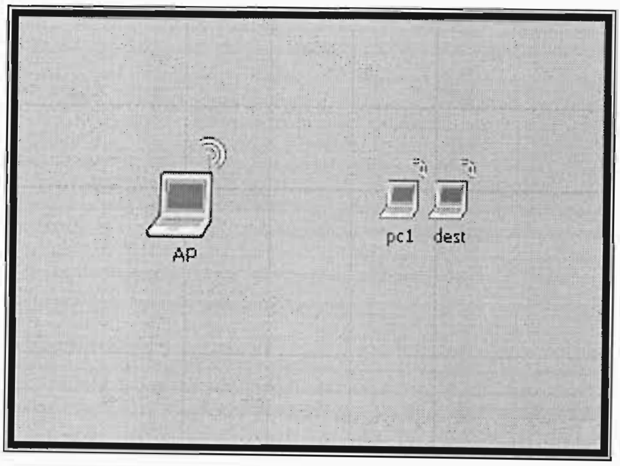

Figure 5.6: Simple Mesh Scenario in OPNET

The access point generates traffic to the destination node located a distance of R_{1max} +R (Km) away. Pc1 is the intermediate node between the AP and destination node. It is a simple two hop network. The first hop is R_{1max} which is the distance between the AP and pc1. The distance between pc1 and the destination node is R which is equal to 300 m. It is assumed, similar to [27], the use of 802.11g at all nodes with maximum transmit power of 100 mW and receiver sensitivity of -95 dBm at 54 Mbps. High gain omni directional antennas are used. The Ad hoc On Demand Distance Vector (AODV) routing protocol is used. The on demand algorithm builds routes between nodes only as desired by source nodes and maintains these routes for as long as they are needed by the sources. The AP is set to generate traffic according to the G_{max} values for the maximum offered load on a node. The results obtained are shown in Table 5.3.

Table 5.3: Simulation Results of a simple mesh network

Year	**R_{1max} (m)**	**G_{max} (Kbps)**	**Traffic Sent (Kbps)**	**Traffic received (Kbps)**	**Delay (ms)**
2007	1302	100	98.653	96.480	1.05
2008	1081	125	123.948	123.948	0.086
2009	945	150	150.038	150.019	0.111
2010	734	200	197.592	197.587	0.058
2011	608	250	247.985	247.983	0.057
2012	467	333	328.363	328.361	0.057
2013	390	400	396.006	396.004	0.0567
2014	290	500	493.020	493.020	0.056

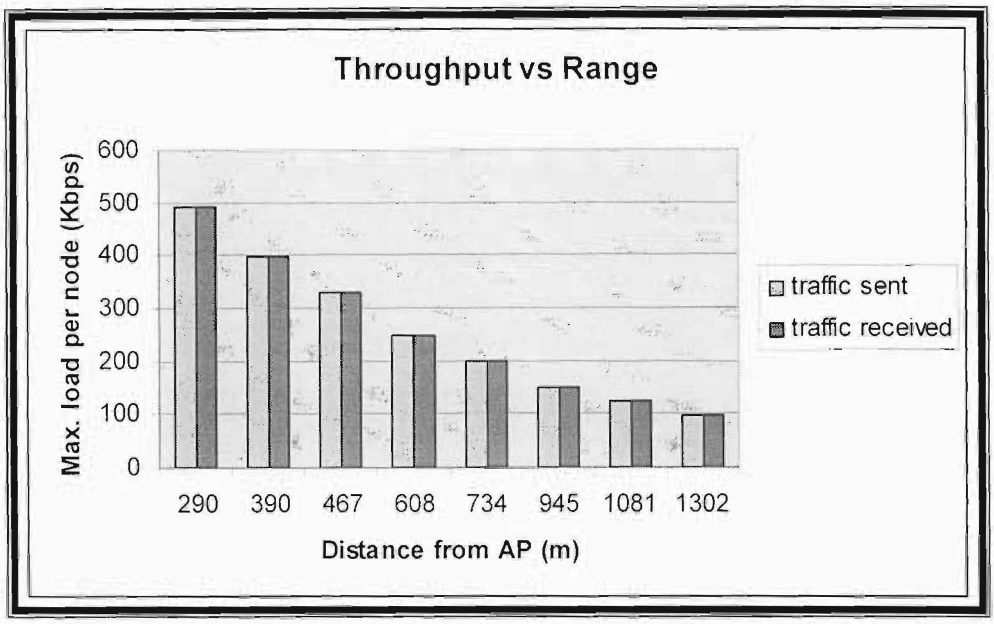

Figure 5.7: Throughput analysis as the range of the first hop is extended in a mesh network.

It can be clearly observed that as the distance of the first hop is increased, the maximum load offered on a node decreases. Figure 5.7 exhibits no significant loss on a mesh network whilst extending the first hop range.

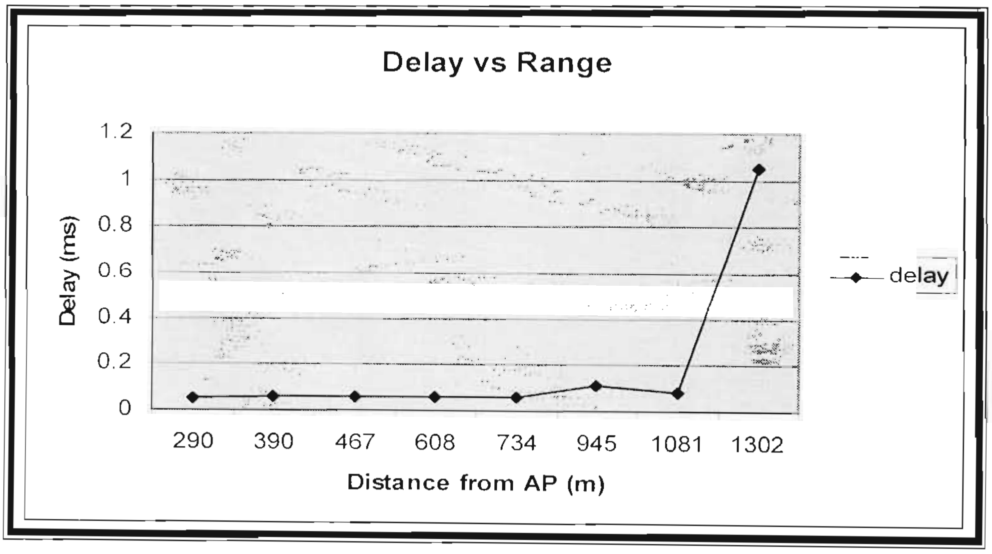

Figure 5.8: Delay vs Range for a simple mesh network

Figure 5.8 shows a sharp increase in the delay of the MANET network when there is a substantial increase in the range of the first hop.

Further investigations were done with a more complex scenario to compare to the analytic model.

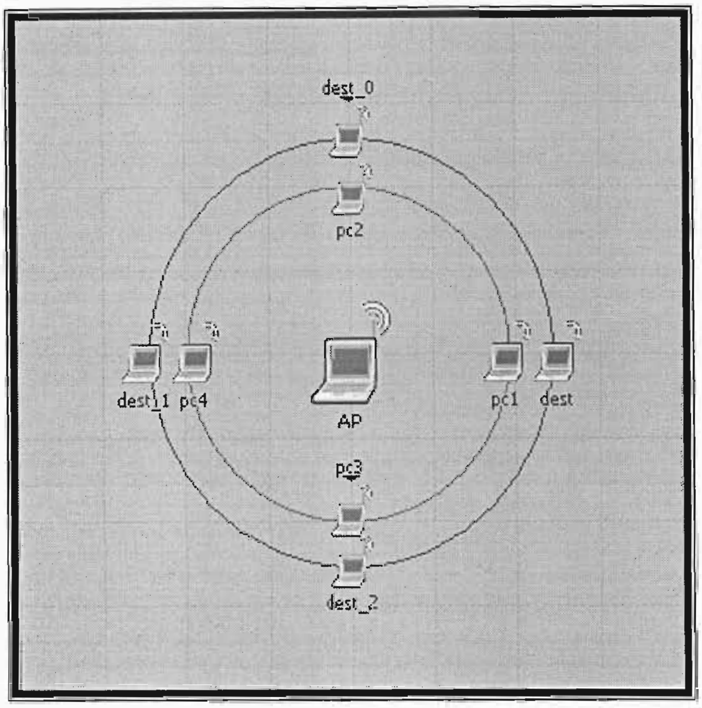

Figure 5.9: Star Mesh Network in OPNET

Figure 5.9 illustrates the star mesh network as explained above, where the area of region D_1 lies within the red circle and the area of region D_2 lies within the blue circle. The nodes are placed on the boundaries of the regions to investigate if the maximum offered load is received by the nodes at those maximum ranges. If this condition is met, it is assumed that all the nodes that lie within those regions will successfully receive its data load. The parameter settings remain the same as the simple mesh network scenario. The results obtained are shown in Table 5.4.

Table 5.4 OPNET Simulation Results for a Star Mesh Network

Year	R_{1max}	G_{max}	Total Traffic	Total Traffic	Traffic Received	Delay
2007	1302	100	393.433	385.111	96.742	5.24
2008	1081	125	488.165	488.165	121.860	0.591
2009	945	150	598.823	598.816	149.996	0.523
2010	734	200	786.729	786.703	196.496	0.516
2011	608	250	984.527	984.508	246.961	0.514
2012	467	333	1313.334	1313.316	329.950	0.512
2013	390	400	1575.949	1575.931	395.153	0.510
2014	290	500	1968.036	1968.036	492.960	0.509

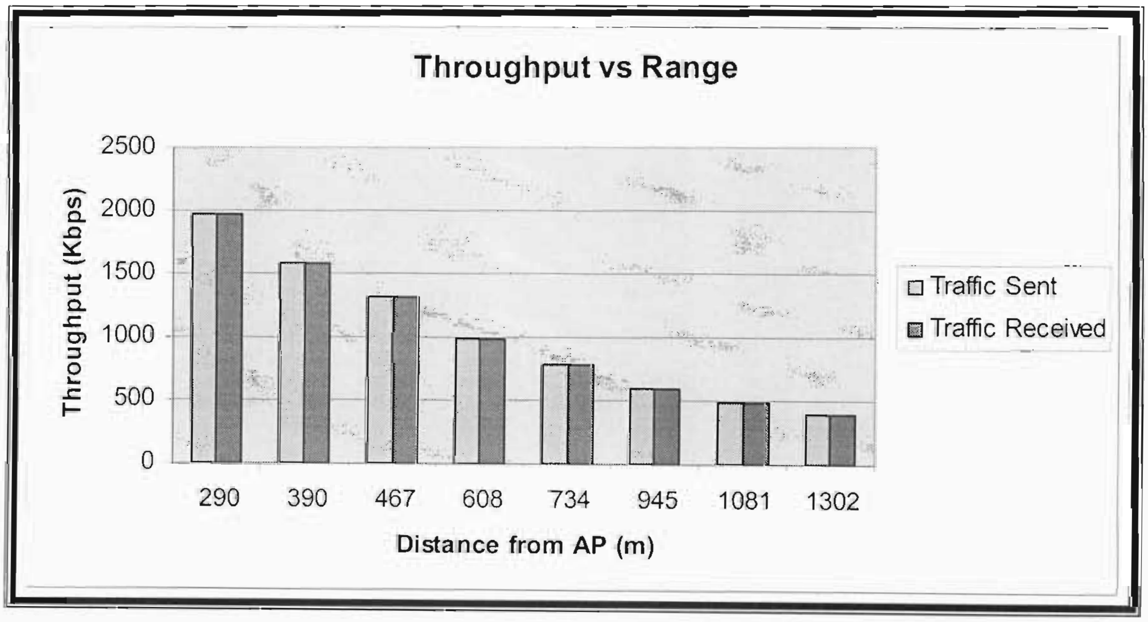

Figure 5.10: Throughput vs Range for a Star Mesh Network

In this sample scenario the access point generates traffic to four destination nodes lying on the border of the maximum range limit for a specific maximum throughput. The simulation results exhibit a high throughput with minimal to zero data loss.

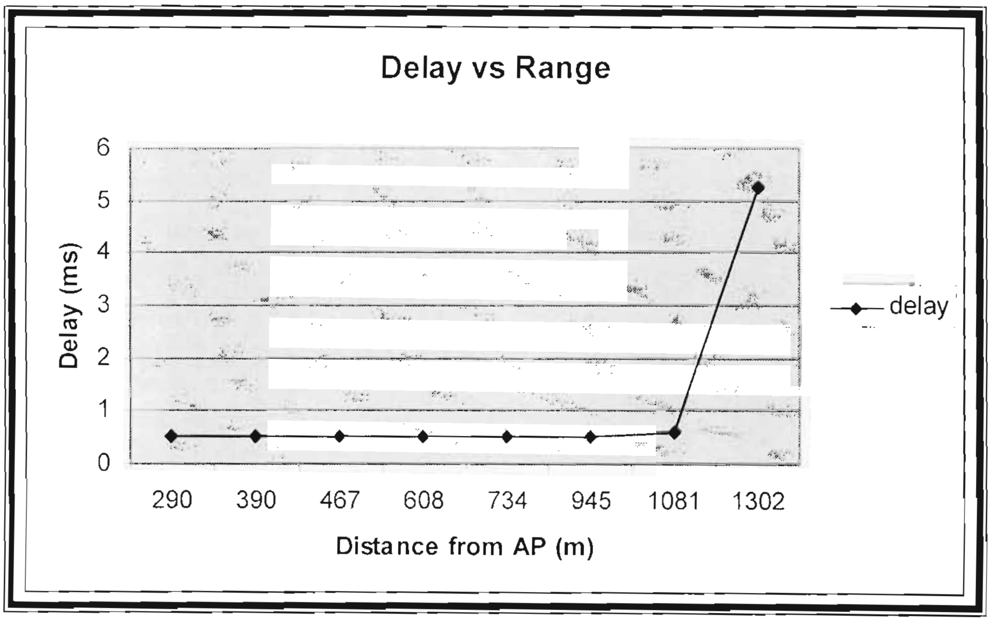

Figure 5.11: Delay vs Range for a Star Mesh Network

Compared to Figure 5.8, it is observed that as the number of nodes in the network is increased the overall delay of the network increases. There is a slow but gradual increase in delay within the microsecond range as the distance is increased to about 1Km, beyond 1Km there is a steep rise in delay to the millisecond range.

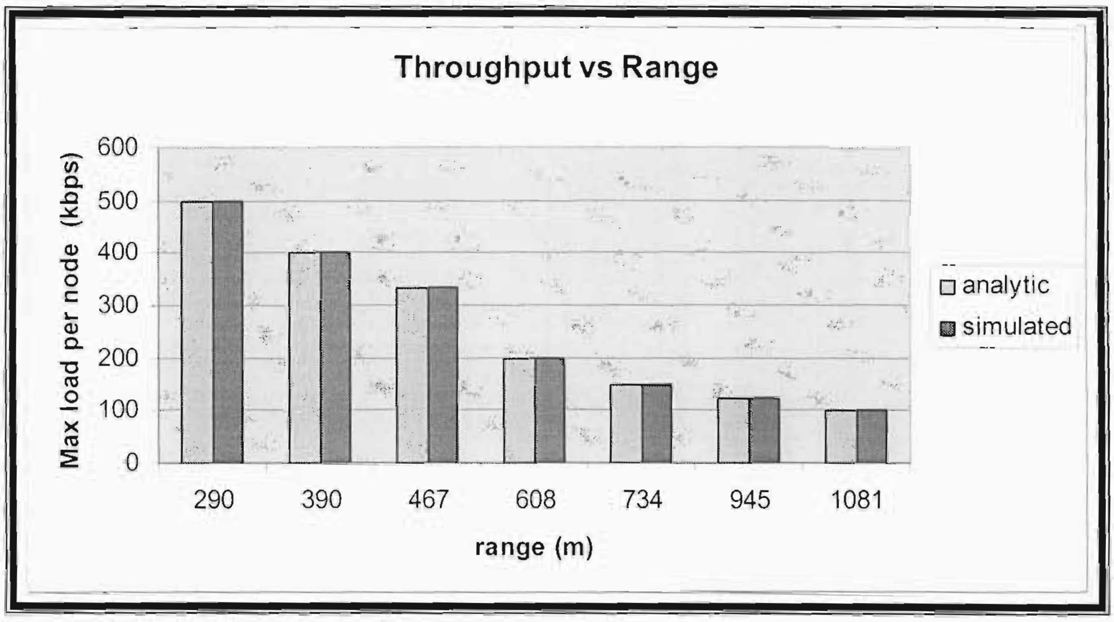

Figure 5.12: Comparison of the analytic results to the simulated results.

Figure 5.12 echoes the conclusion that the simulation results verify the analytic model that for a specific maximum load offered per node there is a corresponding maximum first hop range, beyond which QoS will degrade or the network will fail.

5.6 Star Mesh Network with Sectorized Antennas

Another solution proposed in [27] is to use the Star Mesh Network with sectorized antennas (SMNS) if the network becomes capacity limited. In the SMNS model, the single AP is replaced by 3 APs, each with a sector antenna that covers 120 degree horizontally. Each sector antenna at each AP has a gain of 20dBi. Each AP operates at a different frequency and static routing is assumed as in [27].

The maximum average offered load G_{max} for an SMNS model is:

$$G_{max} \leq \frac{B}{\left(\frac{2N_2 + N_1}{3}\right)} \tag{5.8}$$

We use the assumed parameters from the above SMN model and calculate the maximum first hop range.

A sample calculation for R_{1max} at 2007 is shown below. R = 300m; B = 27Mbps;

$G_{max} = 100Kbps; \rho_h = 25$.

From (8), the values are plugged into the equation and it is simplified to:

$$\tfrac{2}{3}(\rho_h \pi (R^2 + 2RR_1)) + \tfrac{1}{3}\rho_h \pi R_1^2 \leq \frac{B}{G_{max}}$$

Let $K = \frac{R1}{R}$:

$$\tfrac{2}{3}(\rho_h \pi (R^2 + 2R^2K)) + \tfrac{1}{3}\rho_h \pi R^2K^2 \leq \frac{B}{G_{max}}$$

$$\left(\frac{7.07}{3}\right) * \left(2 + 4K + K^2\right) \leq 270$$

$$(K + 2)^2 - 2 \leq 114.41$$

$$K + 2 \leq 10.79$$

$$K \leq 8.79$$

$$R_{1max} = K_{max} R = 4.34 * (0.3) = 2.64Km$$

Table 5.5: The calculated maximum first hop range for the SMNS model

Year	ρ_b	G_{max}(Kbps)	R_{1max}(Km)
2007	25	100	2.640
2008	26	125	2.249
2009	26	150	2.006
2010	27	200	1.632
2011	27	250	1.405
2012	27	333	1.149
2013	27	400	1.007
2014	28	500	0.822

A comparison of the two network models are shown below.

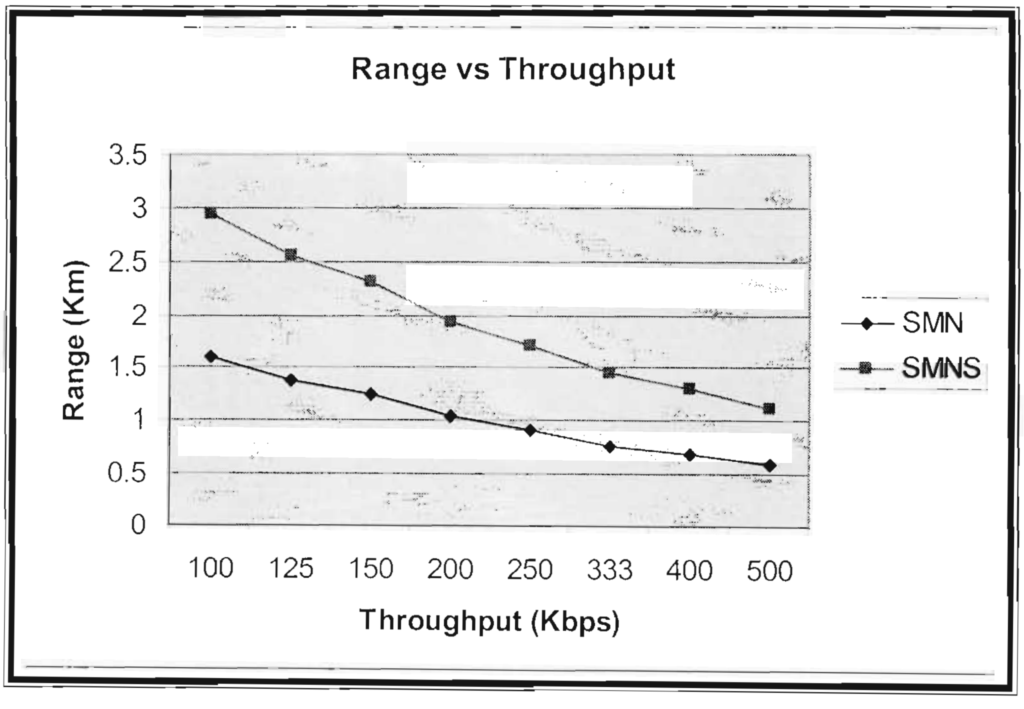

Figure 5.13: Comparison of the SMN and SMNS models

It can be seen clearly from the graph that the Star Mesh Network with sectorized antennas (SMNS) offers a greater range and a total overall area coverage as compared to the SMN. According to [27] SMNS has even better economic performance than SMN. This substantiates the choice of SMNS as a feasible solution to a low density, coverage limited rural area. For a G_{max} value of 100Kbps the SMNS 2-Hop model has a total coverage area of 27.15Km^2 which will provide network coverage to the entire ward 19 of Nongoma and surrounding areas. However, since the SMNS network requires three access points instead of one, it is more costly. If one is willing to trade cost for coverage then the SMNS network is a feasible option. Further study and simulation of the SMNS network is left for future work.

5.7 Summary

This chapter discussed the feasibility of utilising mesh technology to extend WiFi coverage in a rural area. The applicability of a proposed star mesh network solution by Zhang et al. [27] was evaluated for the Nongoma case study. Simulations assisted in drawing the conclusion that the star mesh network is a feasible solution. Further study of a more costly solution, a star mesh network using sectorized antennas is proposed which appears to be promising by extending the coverage area even further.

CHAPTER SIX: CONCLUSION

6.1 Conclusion

In an effort to assist South Africa in the drive to diminish the digital divide, a scalable solution is proposed in this research/dissertation. The implementation of WiFi WLANs at local public venues supported by a WiMax backhaul to the Internet provides a low-cost, sustainable method of introducing telecommunications to rural SA. The region selected for study is Nongoma, a municipal area of Zululand situated in the north-eastern part of Kwa-Zulu Natal. Three telecentres have been dimensioned for an 8 year period to cater for the population residing in and around ward 19 of Nongoma which hosts the Benedictine Hospital, a secondary school and the town. The number of possible subscribers based in each region mentioned above was estimated and a traffic estimate was done for each telecentre for the time period 2006 - 2014. The telecentres cater for the basic communication needs of the people by providing FTP, email, web-browsing, video-conferencing and VoIP services. The total traffic predicted for the hospital telecentre by 2014 is approximately 9 Mbps, for the school telecentre approximately 9.5Mbps and for the simple telecentre approximately 2Mbps.

The abovementioned telecentre scenarios were evaluated via simulation to investigate whether voice applications can be sustained with good QoS in an IP network. The simulations were done using the OPNET modeler 11.5. The simulation results indicate that voice applications will be sustained with good QoS in all the telecentres for the 8 year period. The network was further loaded to establish the threshold for this particular design. It was found for the hospital telecentre that the threshold is reached as the load increases beyond 14Mbps in 2017. By 2019 the network becomes completely saturated and the delays for the real-time applications increase to above their delay budgets. The school telecentre reaches its threshold at 14.283Mbps in 2015, since beyond this point there is an observable deterioration in the QoS and the delays increase rapidly. 2017 marks the point at which voice and video cannot be sustained with reasonable QoS since both applications exceed their delay budgets of 150ms and 100ms respectively.

For all the scenarios it is observed that the FTP delay increases until it reaches a threshold point where the amount of dropped FTP data is so large that the FTP load is now reduced forming shorter queues and lesser delays.

A comparison of the 802.11b and 802.11g protocol was done via simulation which revealed that due to 802.11g having a smaller CWmin and preamble duration in the DCF mode of the MAC layer it produces higher throughput and lower delays for high loads

In most rural regions of South Africa there is little or no infrastructure for PSTN networks. Hence there is an extremely large demand for VoIP applications in these areas. This large demand may lead to a dimensioning problem. In order to accommodate for this growing need the research delves into the solution of providing a larger telecentre to provide mainly VoIP applications. The research investigates the maximum number of simultaneous VoIP sessions that can be maintained by a single AP with reasonable QoS. A previous study that calculated the maximum capacity of a VoIP network operating with 802.11b protocol evaluating various codecs was used to investigate the maximum capacity of a VoIP network using the 802.11g protocol for the rural region. The 802.11g protocol in DCF mode with CTS-to-self protection mechanism was used in the three Nongoma telecentres. Thus an analytic solution was used to calculate the maximum capacity of a VoIP network with these specifications. Simulations verified the analytic results. It was found that the use of a G.729 codec with a voice frame size of 20ms gives a maximum capacity of 40 VoIP simultaneous sessions with good QoS. Thus no more than 40 VoIP phones may be connected in a telecentre. One should bear in mind however that when data traffic is added to the network this value will drop.

A proposed solution was to use Mesh technology to extend the coverage area creating larger WiFi hotspots or even a hotzone. The research proposes to use a combination of the Infrastructure and Client Mesh networks called a Hybrid WMN. For a rural scenario the infrastructure network provides the connectivity to the internet, WiMax and WiFi whilst the routing capabilities of the nodes provide improved connectivity and coverage around the WMN. The applicability of a proposed star mesh network solution by Zhang et al. [27] was evaluated for a rural South African scenario. Simulations assisted in drawing the conclusion that the star mesh network is a feasible, low-cost, robust solution. For a maximum throughput of 100Kbps on each node in the network the distance of the first hop from the access point is 1.302Km resulting in a total coverage area of 8.06Km^2.

6.2 Recommendation for Future Work

The sites for telecommunication trends in rural areas and data collection should be expanded for South Africa; more detailed maps with specific areas and population statistics should be more easily available. Further, telecommunication data and trends for South African rural regions need to be published. This will facilitate accurate dimensioning of the WiFi-WiMax networks for rural regions.

The maximum capacity of VoIP calls in an 802.11g, CTS-to-self enabled network has been calculated. This maximum capacity will however be reduced in the presence of data traffic. The extent of the reduction in capacity in the presence of data traffic is left for further research.

A more costly solution, a star mesh network using sectorized antennas is proposed to extend the coverage even further than a simple SMN network, but this model requires further investigation for the rural scenario.

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APPENDIX A

The settlement types can be summarised as follows:

Table A1: Settlement Types of Rural Areas [3]

Item	Type	Characteristics
1	Urban	Built-up areas
2	Rural Village	1 200- 1 500 people per km^2
3	Rural Scattered	55- 1 200 people per km^2
4	Scattered	0- 500 people per km^2
5	Rural Farms	Communities living outside tribal areas on farms

Table A2. shows Nongoma's population distribution according to types of settlement. In 2000 Nongoma had a population of 230 672 people.

Table A2: Population distribution (2001) [3]

Urban	Rural Village	Rural Scattered	Scattered	Rural Farms	Total
3 842	183 525	43 305	0	0	230 672
1.66%	79.56%	18.78%	0.00%	0.00%	100.00%

Table A3: Shows the call rates and inter-arrival times for the various telecentres

Table A3-1: Hospital Telecentre (12 hour day)

Year	calls/mth	calls/phone	calls/hr	IRT (sec)
2006	8528	1421.333	3.948148	911.8199
2007	9636.64	1606.107	4.461407	806.9203
2008	11563.97	1927.328	5.353689	672.4335
2009	14223.68	2370.613	6.585037	546.6939
2010	17210.65	2868.442	7.967895	451.8132
2011	20652.78	3442.131	9.561474	376.511
2012	23750.7	3958.45	10.9957	327.4008
2013	27550.81	4591.802	12.75501	282.2421
2014	31407.93	5234.655	14.54071	247.5808
2015	36193.95	6032.325	16.75646	214.8425
2016	43432.74	7238.79	20.10775	179.0354

Table A3-2: School Telecentre (8 hour day)

Year	calls/mth	calls/phone	calls/hr	IRT (sec)
2006	2560	640	2.666667	1350
2007	2892.8	723.2	3.013333	1194.69
2008	3471.36	867.84	3.616	995.5752
2009	4269.773	1067.443	4.44768	809.4107
2010	5166.425	1291.606	5.381693	668.9345
2011	6199.71	1549.928	6.458031	557.4454
2012	7129.667	1782.417	7.426736	484.7351
2013	8270.413	2067.603	8.615014	417.8751
2014	9428.271	2357.068	9.821116	366.5571

Table A3-3: Simple Telecentre (8 hour day)

Year	calls/mth	calls/phone	calls/hr	IRT (sec)
2006	4250	1062.5	4.427083	813.1765
2007	4802.5	1200.625	5.002604	719.6252
2008	5763	1440.75	6.003125	599.6877
2009	7088.49	1772.123	7.383844	487.5509
2010	8577.073	2144.268	8.934451	402.9347
2011	10292.49	2573.122	10.72134	335.7789
2012	11836.36	2959.09	12.32954	291.9816
2013	13730.18	3432.545	14.30227	251.7083
2014	15652.4	3913.101	16.30459	220.7968

*IRT = Inter Request Time

APPENDIX B

B.1 OPNET Parameter Settings

For the wireless workstations in the scenario the following parameters were defined and set.

BSS Identifier: Identifies the BSS to which a WLAN MAC belongs.

Access Point Functionality: It enables or disables access point operation in the node. It is used to configure Basic Service Set (BSS) and Extended Service Set (ESS) topologies. It is required to be enabled for PCF operation or to support roaming.

Transmit Power: A node's fixed transmission power in watts.

Packet Reception Power Threshold: Defines the receiver sensitivity in dBm (Vendor specific). A packet whose reception power is less than the threshold will not be sensed by the MAC. Such packets may still cause interference noise at the receiver.

RTS Threshold (bytes): Sets the packet size threshold for which the request-to-send (RTS) / clear-to-send (CTS) mechanism will be used. It is a solution to the hidden terminal problem. Overhead is due to the RTS/CTS frame exchange.

Short and Long Retry Limits: Specifies the maximum number of transmission attempts for a data frame. Two independent counters – Long retry count incremented only if a data transmission fails despite a successful RTS/CTS exchange. High retry limits may perform better in noisy networks. Low retry limits can be suitable for networks with high mobility.

Fragmentation Threshold (bytes): When the MSDU is greater than the threshold, fragmentation occurs. Smaller packet size reduces packet loss but increases overhead.

Large Packet Processing: Action taken in the case where a higher layer packet size is greater than the maximum allowed data size. Based on this, a packet will be dropped or fragmented. It is outside the scope of the standard.

Max Receive Lifetime (seconds): The maximum time for a packet to wait to be reassembled at the receiver's reassembly buffer.

Buffer Size (bits): Maximum length of higher layer data arrival buffer

AP Beacon Interval: Specifies how often beacons will be transmitted by an AP. A low value provides faster roaming but consumes more time of the shared medium.

Roaming Capability: Enables the MAC to perform scanning procedures to associate with another AP when the communication is lost with the current one. It requires configuration of regular WLAN operational channels and cannot be turned on for access points.

CTS-to-self Option: 802.11g specific. It is used as a protection mechanism in mixed 11b/11g networks. It is an alternative to RTS/CTS frame exchange.

B.2 OPNET Queuing System Concepts

The definition of a queuing system is a data network where packets arrive, wait in various queues, receive service at various points, and exit after some time.

- The arrival rate is the long-term number of arrivals per unit time.
- Occupancy is the number of packets in the system (averaged over a long time).
- Time (delay) in the system is the time from packet entry to exit (averaged over many packets).
- A single queue system is stable if the packet arrival rate is less than the system transmission capacity.
- For a single queue, the ratio: **packet arrival rate / system transmission capacity** is called the utilization factor which describes the loading of a queue.
- In an unstable system packets accumulate in various queues and/or get dropped.
- For unstable systems with large buffers some packet delays become very large.

- Flow/admission control may be used to limit the packet arrival rate whilst prioritization of flows keeps delays bounded for the important traffic. Stable systems with time-stationary arrival traffic approach a steady state.

Delay is caused by interference. If arrivals are regular or sufficiently spaced apart, no queuing delay occurs. Interference is caused by burstiness and packet length variation. It should also be noted that high utilization exacerbates interference. Bottlenecks typically occur at the access points or at points within the network core as a result of overloads caused by high load sessions or convergence of a number of moderate load sessions at the same queue.

Table B.1: Source Type Properties [18]

Traffic	Characteristics	QoS Requirements (one way)
Voice	-Alternating talk-spurts & silence intervals -Talk-spurts produce constant packet rate traffic	Delay < ~150ms Jitter < ~ 30ms Packet Loss < ~ 1%
Video	-Highly bursty traffic (when encoded)	Delay < ~ 100ms Jitter < ~ 30ms Packet Loss < ~ 1%
Data FTP Web telnet	-Poisson type -Sometimes batch arrivals or bursty or sometimes on-off	Zero or near-zero packet loss Delay tolerant

For priority servers and queuing, packets are classified into separate queues. There is a separate FIFO queue for each priority class. All packets in a higher priority queue are served first. Packets of lower priority start transmission only if no higher priority packet is waiting. Typically in routers, if a higher priority packet arrives while a lower priority packet is being transmitted, it waits until the lower priority packet completes.

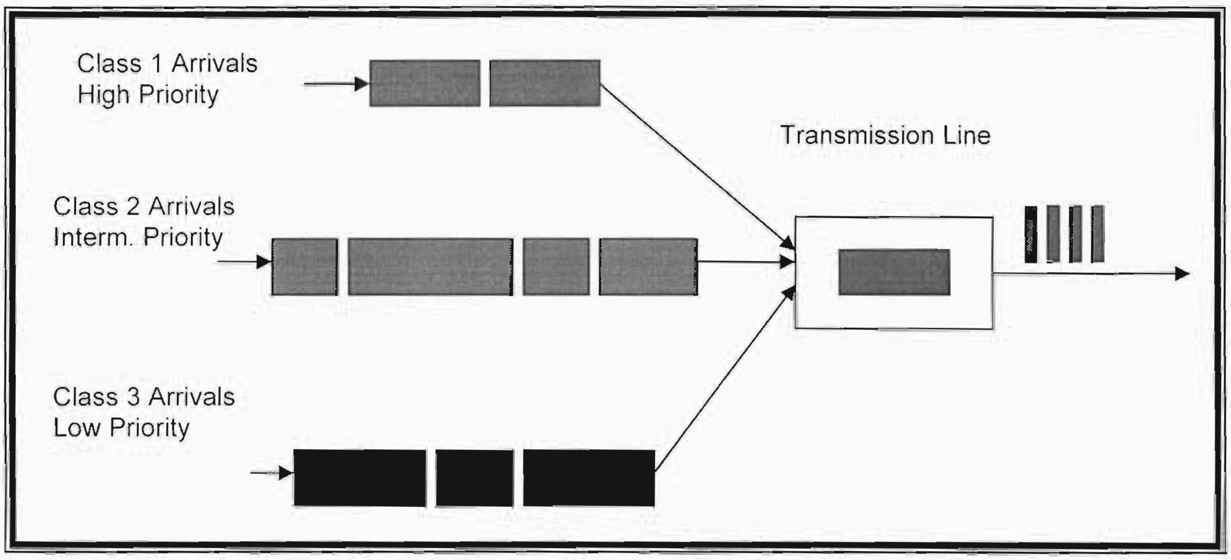

Figure B.1: Priority Queuing