Extending WiFi Access for Rural Reach

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Abstract— **WiFi can be used for last-mile IP connectivity to rural users. In initial roll-out, hotspots can be positioned at community centres such as schools, clinics, hospitals or call-centres. The study will consider a typical South African rural region, with telecommunications services traffic estimates. The study focuses on IEEE 802.11G deployment option based on the requirements of the South African case. The research investigates the maximum number of VoIP calls with reasonable QoS that can be sustained on a WiFi network.**

Index Terms—WLAN, WiFi, VoIP, QoS

I. INTRODUCTION

South Africa has great disparity in rural versus urban telecommunication infrastructure provisioning. To bridge this digital divide appears to be a daunting task. The ECT act of 2005 is a government policy that governs the future development of telecommunications in South Africa and has become a major political issue. 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) 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 Internet is the tool to accessing wealth and knowledge and to empowering our nation. 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.

South Africa has an alarmingly low teledensity in some rural areas – less than 5% 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 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 value proposition in South Africa.

The aim of this research is to propose a sustainable method of providing basic VoIP telephony and Internet access to the government targeted rural communities identified in [1]; and Zululand is the targeted region. Many of the rural areas in Zululand do not have a locally accessible wire-line infrastructure to provide for a basic dial-up access to the Internet. Hence the research proposes a fixed wireless LAN (WiFi) implementation as well as ways of extending the WiFi range to get optimal coverage. This can be accomplished by means of WiFi backbone configuration and star mesh network topology. Backhaul networks were not considered but Worldwide Interoperability for Microwave Access (WiMAX) technology can potentially be utilized.

II. SCENARIO

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 draw backs 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 a substantial number of VoIP sessions, however the practical implementation shows a dramatically reduced number of calls that can be sustained. This is due to various protocol overheads. Thus the research is twofold:

1. To investigate the maximum number of quality VoIP calls that can be sustained on a WiFi network.
2. To investigate the performance of real-time traffic such as voice and video in 802.11 solutions during heavy load. Voice capacity and quality is observed as the bandwidth is shared with data applications.

The aim is to provide voice plus Internet access to telecentres in rural areas. The area of interest is limited to Ward 19 of Nongoma district in Zululand. There is only one hospital situated in Nongoma, namely the Benedictine Hospital in Nongoma town. 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. The telecentre profile is shown in Table 1. Jensen and Esterhuysen [3] 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 estimate is used for the selection of 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 1: Telecentre Profiles

No.	Telecentre	# of phones	# of pcs	%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%

The population for ward 19 in Nongoma is 3847 with an area of 21.84km^2. Since ward 19 hosts the town and only hospital for Nongoma it will be frequented by the population from surrounding areas. Hence the network is dimensioned to cater for these numbers. An estimated 15 951 people reside in Ward 19 and surrounding areas, which is approximately 6.72% of the Nongoma population. It is assumed that no other technology will penetrate the region during the 8 year period considered, and hence a market share of 100% is used.

III. TRAFFIC PROJECTIONS

A. VoIP Traffic Predictions

The access network is to be dimensioned for the eight years starting 2006 and ending 2014. 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. The layer 2 header, cRTP and Voice Activation Detection (VAD) have been accounted for in the bandwidth calculations.

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

$$\textit{Number of subscribers} = \rho\, P\, \gamma\, \sigma\, \alpha \qquad (1)$$

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.)

The yearly percentage increases is predicted using previous PSTN growth trends in South Africa [5].

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. 425 subscribers will frequent this telecentre making on average 10 calls per month each resulting in a total of 4250 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. Similarly calculations for the hospital and school telecentre were done. The growth trend for the VoIP traffic over a period of 8 years is shown in Figure 1.

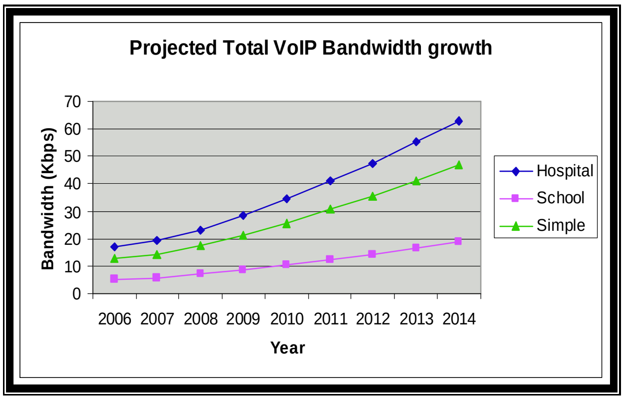

FIGURE 1: Graph of projected VoIP traffic

B. 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 [6]. The data and email percentage increase is estimated from Internet trend patterns. 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 16 Kbps and for file transfer it is about 400 Kbps. The 2006 telemedicine bandwidth requirement is the minimum bandwidth required to meet diagnostic needs which is 1.5Mbps [7]. 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. The minimum average web-browsing bandwidth is 500 kbps.

To satisfy user requirements, the acceptable average tolerance of users to delays or download times is 10 seconds with a maximum of 15 seconds [7]. 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, web-browsing requires 500Kbps, it will only require 500Kbps/10s = 50Kbps. Similarly the bandwidth requirements for Email and HTTP are calculated.
The total bandwidth required for data applications can be calculated using the following formula:

$$\text{Bandwidth} = P * N * O + R \qquad (2)$$

Where:
P = number of PCs 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 1
O = Overbooking factor. (Assumes that not everyone is online at the same time therefore 'peak' bandwidth has not been allocated to every PC/user.)
R = Data Transfer Rate of Real-time Applications such as; Videoconferencing, e-Learning, Telemedicine, streaming audio/video, VoIP.
N = Data Transfer Rate of Non-Real Time Applications such as; email, web-browsing, documents (downloads), limited audio/video.

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 x 50 Kbps x 0.4 = 120 Kbps
School: Data BW = (8 x 50 Kbps x 0.4) + 1600 Kbps = 1760 Kbps
Hospital: Data BW = (8 x 50 Kbps x 0.4) + 1500 Kbps = 1660 Kbps

Table 2: Data Bandwidth Requirements for each telecentre

Year	**Simple telecentre Data BW (Kbps)**	**School telecentre Data BW (Kbps)**	**Hospital telecentre Data BW (Kbps)**
2006	120	1760	1660
2007	132	1856	1751
2008	171.6	2076.8	1961.3
2009	257.4	2486.88	2352.9
2010	437.58	3155.856	2995.08
2011	722.007	4203.92	4001.342
2012	1083.011	5527.982	5272.734
2013	1516.215	7330.778	6998.955
2014	1971.079	9530.011	9098.642

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 for each telecentre.

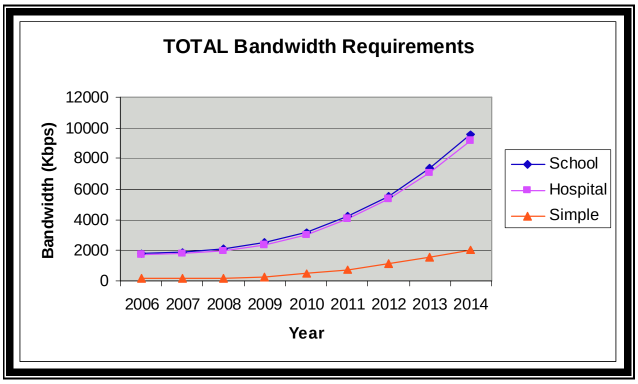

FIGURE 2: Total Bandwidth required for the telecentres

IV. SIMULATION RESULTS

A. 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. DCF can be used in both, ad-hoc or infrastructure modes [10]. Research has shown that for the Nongoma rural scenario 802.11g is the preferred protocol since 802.11g offers a higher throughput and lower delays as compared to 802.11b for higher loads [8]. Networks built to the 802.11g standard boast a data rate of 54Mbps yet have 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. 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.

The work done in this section is closely related to the work of Hole and Tobagi [11], where the capacity of an IEEE 802.11b network carrying voice calls in a wide range of scenarios were evaluated. Hole & Tobagi 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, given in equation 3.

$$N = \{1 / R [2(T_{VOICE} + SIFS + T_{ACK} + DIFS) + (T_{SLOT} X CW_{MIN} / 2)]\} \quad \text{\_\_\_\_\_\_} \quad (3)$$

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. Values for the above parameters are found in Table 3 and Table 4.

Table 3: 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: Component Times of T_{VOICE} & T_{ACK} for 802.11b at 11Mbps & 802.11g at 27Mbps

		802.11b (11Mbps)	**802.11g (27Mbps)**
T_{voice}	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
T_{ack}	PLCP Preamble & Header	192µs	26µs
	ACK Frame	10.2µs	4.15µs

In this research, equation 3 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. The parameter T_{CTS} which is the CTS duration is included in the calculation of the total transaction time in equation 4.

$$N= \{1 / R [2(T_{VOICE} + SIFS + T_{ACK} + DIFS + T_{CTS}) + (T_{SLOT} X CW_{MIN} / 2)]\} \quad \text{\_\_\_\_\_\_}(4)$$

The values for the parameters in equation 4 can be found in Table 3 (802.11g/ 802.11b compatible) and Table 4 (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.

Equation 4 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 5 tabulates the maximum number of VoIP connections for the three codecs.

Table 5: Maximum number of VoIP connections for various codecs

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

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. For example for a 10ms payload size, the packet size will be *x* and for a 20ms frame size the packet size will be 2*x*. 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 [12], that 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 [12]. The G.729 codec 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. 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 6: 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 close agreement with those obtained via simulation using the OPNET modeler. The tightness of the analytical upper bound is demonstrated in the simulations due to the delay constraints [8]. 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.

B. WiFi WLANs Modeled with Data Applications and VoIP

The objective of the simulation studies is to characterize some of the proposed WLANs mentioned earlier. The projected traffic for the time period between 2006 and 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 OPNET Modeler version 11.5.

The hospital telecentre simulation scenario is configured to support five applications, namely; FTP, email, web-browsing, voice and videoconferencing. 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 [8]. In order to evaluate the end-to-end 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.

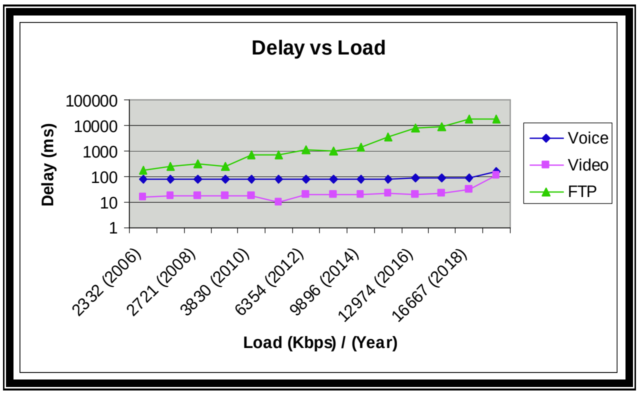

FIGURE 3: End-to-End Delay vs. Load for the Hospital telecentre

Figure 3 shows the end-to-end delay of the various applications as the load is increased on the network. The voice packet end-to-end 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. 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. 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 is 98.86% correlated to the work of Gast [9] 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. 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.
The school telecentre scenario is very similar to the hospital telecentre scenario with regard to the set-up and applications offered. Two of the 8 computers are assigned to run e-learning applications only. The 802.11g protocol at a data rate of 24 Mbps is applied on all the nodes. At 2014 the total load on the school telecentre network is 10.030Mbps. 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.
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. The 802.11g (extended physical rate) protocol at a data rate of 2 Mbps is applied to all nodes. 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 150ms 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.

V. CONCLUSION

In an effort to assist South Africa in the drive to diminish the digital divide, a scalable solution is proposed in this research. The implementation of WiFi WLANs at local public venues supported by a WiMax backhaul to the Internet provides a sustainable method of rolling out telecommunications to rural SA. To accommodate the growing demand of VoIP applications in rural regions the research proposes a solution of providing telecentres to provide mainly VoIP applications. This research was done based on the work of Hole & Tobagi [9], to calculate the maximum capacity of an 802.11g with CTS-to-self VoIP network using a G.729 codec with various frame sizes. 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. The research also looks at the sustainability of telecentres offering data services with VoIP services. Three telecentres have been dimensioned for an 8 year period to cater for the population residing in and around ward 19 of Nongoma. 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 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. The maximum number of VoIP calls that can be sustained in the presence of data traffic is left for further research.

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