DEPLOYMENT OF A SUSTAINABLE, PRODUCTION-GRADE VoIP SYSTEM BASED ON OPEN SOURCE SOFTWARE COMPONENT IN AN EDUCATIONAL INSTITUTION: THE CASE OF THE UNIVERSITY OF NAMIBIA

A thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Information Technology of the University of Namibia

By

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Abstract

Governments and their agencies are often challenged by high cost and flexible telephonic and data services. Emerging technologies, such as those of Voice over Internet Protocol (VoIP) that allow convergent systems where voice and data networks can utilise the same network to provide both services, can be used to improve such services.

However, these convergent networks are based on the classical (best-effort) characteristics that come with some weaknesses in respect of quality of service and fair access to network resources. This is true for multimedia applications that need bounds on delay and minimum bandwidth. VoIP is an implementation of these convergence networks that are capable of transporting voice over Internet Protocol (IP) based networks.

In order to deploy a VoIP network capable of providing the traditional Public Switched Telephone Network–Private Branch Exchange (PSTN-PBX) scale solution, a number of issues such as services to be offered, end-user-terminal, quality of service, security, bandwidth, signalling, protocol and operating legislations must be addressed. The implementation also requires an application software, which can be an open source or a proprietary software.

This study examined how Asterisk, an open source VoIP software can be deployed to serve the needs of an educational institution. The educational institution in this case is the University of Namibia which is currently using a conventional PSTN system for voice and fax communication services, as well as the local area network connected to Internet for data services. Like any other open source software, Asterisk comes free of any proprietary costs.

The study investigated how this software could be deployed for a longer period at the University of Namibia. Asterisk was deployed on a pilot basis to provide for a larger scale model to cater for the entire university.

It was found out that the University of Namibia has a potential to implement the project although implementation can be scaled down so as to support sustainability. Since the software recommended for installation is open source, the project could be used as a source of information by students who specialize in real-time multi-media systems.

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Many thanks to all the people I have come to know as a result of this work. I owe my sincere appreciation to my wife, my family and relatives who have supported and encouraged me over the years. I especially want to thank Mr. L. Mwansa and Mr. E. Mulenga for their inspiration and continuous encouragement during my studies.

Finally, I want to extend my profound appreciation to my beloved parents for their love, affection and invaluable support during my life and studies.

Dedication

I dedicate this research work to my wife Linda and all my children.

Declarations

I declare that *Deployment Of A Sustainable, Production-Grade VoIP System Based On Open Source Software Component In An Educational Institution: The Case Of The University Of Namibia* is my own work, that it has not been submitted for any degree or examination in any other university, and that all the sources I have used or quoted have been indicated and acknowledged by complete references.

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Acronyms

ADSI	Analog Display Service Interface
API	Application Program Interface
CODEC	Coding and Decoding
Digium	Digium is the primary developer and sponsor of Asterisk™,
E1	E1 carries signals at 2 Mbps (32 channels at 64Kbps, with 2 channels reserved for signaling and controlling)
IAX	Inter-Asterisk Exchange
IP	Internet Protocol
ISDN	Integrated Services Digital Network, an international communications standard for sending voice, video, and data over digital telephone lines or normal telephone wires
IVR	Interactive Voice Response
OSS	Open Source Software
PBX	Private Branch eXchange
PRI	Primary-Rate Interface, a type of ISDN service designed for larger organizations
PSTN	Public Switched Telephone Network, which refers to the international telephone system based on copper wires carrying analog voice data.
SIP	Session Initiation Protocol
T1	A dedicated phone connection supporting data rates of 1.544Mbits per second
TDM	Time Division Multiplexing
VoIP	Voice over Internet Protocol

Chapter 1 Introduction

1.1 Background Information

A business challenge in today's Governments and their agencies is addressing the need to reduce costs, ensure continuity of government services, and improve effectiveness [Tanner 2006]. Traditional telephony solutions, however, fail to support these goals. As an example, outdated Private Branch eXchange (PBX) and the Public Switched Telephone Network (PSTN) systems require costly maintenance agreements and are difficult to integrate with other systems to improve services as they emerge with new technology [Carhee 2004].

The same drawbacks apply to the University of Namibia, where personnel moves, adds, and changes cost between N$500.00 and N$1,000.00 each, and adding basic productivity-enhancing features like voice mail, conferencing and calling increase the monthly charges. The university uses two communication networks: one for the voice and the other for the data and Internet access. It is expensive to maintain each of these networks separately. Convergent networks enable the combination of two networks in one that can provide both services of voice and data [Mueller 2002].

However, these convergent networks are based on the classical (best-effort) characteristics that come with some weaknesses in respect of quality of service and fair access to network resources especially for multimedia

applications that need bounds on delay and minimum bandwidth [Vagesna 2001]. Voice over Internet Protocol (VoIP) is one result of these converging networks that is capable of transporting voice over IP based networks. In order to deploy a VoIP network capable of providing the traditional PSTN-PBX solution, a number of issues such as services to be offered, end-user-terminal, quality of service, security, bandwidth, signalling, protocol and operating legislations must be addressed [Davidson et al. 2000]. Implementation of VoIP also requires application software, which can be open source or proprietary.

Much of the work already done on VoIP and its implementation differs according to the nature of deployment [Walker and Hicks 2004]. There is need that institutions undertake research in order to determine the appropriateness and viability of these converged networks in the context of their infrastructure. This is consistent with the issues raised by Mahler such as planning, analysis and assessment [Mahler 2005].

Recently, with available broadband services and the integration of voice and data at all levels, VoIP has begun to take off as an alternative to the traditional PSTN since its leverage as a single IP network to provide both data and voice services reduces on costs [Keagy 2000].

Further, VoIP applications have the potential to provide services that are difficult to implement in traditional phone systems such as deploying a unified

messaging system that would voice synthesize e-mails over a phone to the subscriber [Sulkin 2002].

In addition to cost advantages, VoIP services have compelling technical advantages over circuit switching. VoIP networks are based more on an open architecture than their circuit-switched contemporaries. This open, standards-based architecture means that VoIP services are more interchangeable than ones in a proprietary, monolithic voice switch [Ohrtman 2003]. It is possible to select the best product without being tied to one specific vendor.

As a result of this, components can be added and modified according to the requirement of the user. The open standards can also facilitate realization of new services that can rapidly be developed and added to the packages offered to the customers.

In terms of market growth, VoIP has taken a sharp growth. This is according to a market research that was conducted by In-stat. Analysts from Ins-stat also predicted that by the year 2011, VoIP will be used by around 66% U.S businesses [Ins-stat 2007]. In emerging market countries like South Africa, deregulation and related policies are fostering growth of VoIP infrastructures.

Global figures indicate that between 1997 and 2001, international VoIP traffic grew to a triple-digit every year. According to 2005 figures from telecoms market research firm Tele Geography, VoIP subscribers were to grow from 4 million in 2005 to 17.5 million by 2010 with annual revenues exceeding US$5

billion. Figure 1 shows projected VoIP subscribers and revenue for the period 2003-2010. This research also indicated that developing nations are the fastest growing destinations for international VoIP traffic. Brazil and Nigeria led the world in growth in 2004 with 112% and 103%, respectively. Bangladesh was third with 97% followed by the Dominican Republic at 81% [TeleGeography Research 2005].

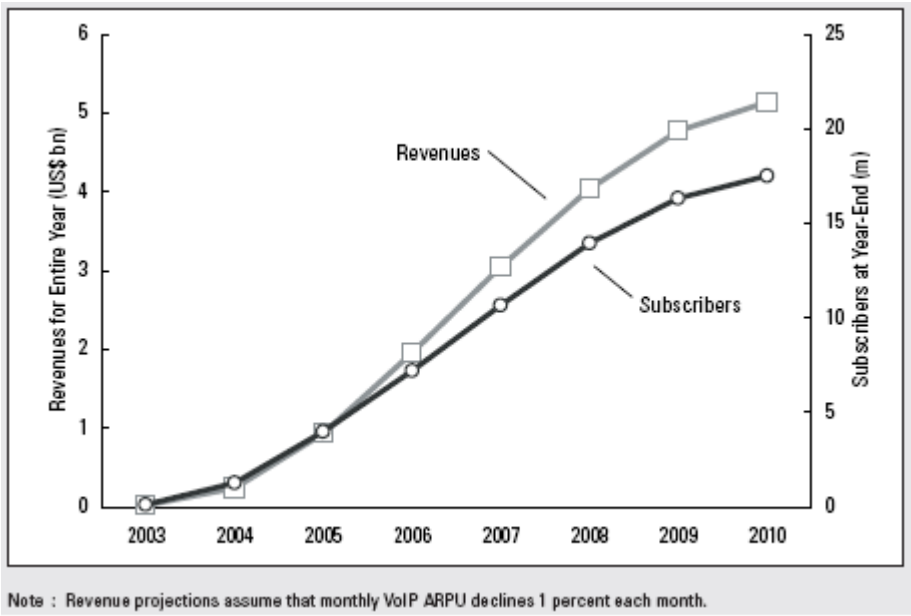


Figure 1 : Projected VoIP subscribers and revenue for the period 2003-2010. Source: TeleGeography Research 2005

Regulations of VoIP vary significantly around the world. In developing countries, regulators are mainly focussing on issues associated with licensing and entry into the market with regard to competition with existing Telecoms who are still providing the traditional PSTN telephony services. In developed countries, regulators are more concerned with post-entry issues like numbering resources, access to emergency services, regulatory fees and

quality of service [Lipman 2005]. In Namibia, regulations do not allow providers to use VoIP.

According to Wiegand, globally there were more countries that prohibited VoIP in 2005 than those that allowed it [Wiegand 2005]. Results from the annual ITU Telecommunication Regulatory Survey [ITU-D 2004] indicate that out of a total of 132 member states:

- 49 states representing 37% have unambiguously declared VoIP legal of which only 6% of these were from Africa.
- 11% states had partial competition where non-licensed Public telephone Operators (PTOs) were allowed to use either IP-based networks or the public Internet for the conveyance of voice calls.
- 11% states also had no policy regarding IP Telephony.
- In 24 states, IP telephony was prohibited. The majority of these were from Africa.
- 37 states restricted IP Telephony allowing only PTOs to use IP-based networks or the public Internet for conveyance of voice calls.

In this project VoIP was deployed and customised so as to provide communication services to students and lecturers at the University of Namibia. A VoIP model to address long term sustainability was also recommended for the university.

Similar projects have been conducted in South Africa and America. Rhodes University in South Africa put together the iLanga system from various

software components implemented by the open source community which included Asterisk, SIP Express Router (SER), and OpenH323 Gatekeeper [Penton and Terzoli 2004]. This system currently provides an environment that enables users who are staff and students of the university to communicate with each other using VoIP as well as legacy telephones including connection to the public Telkom's network.

In America, a study about comparative analysis of traditional telephone and VoIP systems was conducted and brought about successful deployment of VoIP to 1300 users by the United States Department of Education. Carnegie Mellon University and Colorado State University also had conducted research studies which begun with trial deployments before covering the entire campuses [Chong and Matthews 2004].

In today's business world, when implementing network solutions, especially for small-medium businesses, it is important that a proper design process is mastered by networking professionals [Rybaczk 2004]. These solutions involve issues that refer to implementation of IP telephones, security, unified communication and Customer Relations Management (CRM). This addresses the importance of VoIP deployment considerations based on IP components and related protocols.

The telephone network has revolutionized virtually in every aspect of human interaction. The shift to VoIP in corporations has brought about cost effectiveness, easy management and easier and quicker methods of adding

services than the traditional telephony system. During 2006, IBM and 3com unveiled a VoIP solution aimed at small and mid-sized business using 3Com software and the IBM System i mid-range computer [3Com 2006].

1.2 Problem Statement

This research project deals with two major problems. The problem of deploying the converged system and the problem of sustainability of the deployed system

1.2.1 Problem of deploying a converged system

Although VoIP is capable of offering cost effective solutions to voice and data services, users or customers do not generally accept voice quality or service that is inferior to what they are used to with the traditional systems. Deploying VoIP is also not as easy as installing a computer application on a server or personal computer. In deploying VoIP, there is need to analyse the network infrastructure in order to come up with a suitable logical and physical network topology that will support data and voice services. It is important that institutions consider an investigation of telephony usage within so as to determine correct functionality required from the VoIP network. This will address potential issues that go along with deploying a VoIP network [Salah 2006].

1.2.2 **Problem of sustainability of the deployed system**

Sustainability matters can be thought of as total cost of ownership such as cost of acquisition, implementation, maintenance and exit. In this project, an Open Source Software (OSS) product will used in the deployment. Therefore issues related to OSS sustainability need to be investigated and provide a solution.

Due to the fact that the VoIP technology development is currently highly dynamic, it is important to ascertain sustainability in terms of internal arrangement for supporting the system based on emerging technologies which need incorporation as well as offering support when new services are required. Therefore, a need to assess technology know-how of people expected to manage the system.

1.3 Working Hypothesis

Based on the problems identified it is therefore apparent to state the following as the working hypothesis “It is possible, using currently available VoIP Free OSS technology, to sustainably deploy integrated telephony and data services to staff and students at an academic institution”. This hypothesis will be articulated and tested in the case of The University of Namibia.

The general objective is to determine factors and conditions required for a production grade OSS system to be deployed in order to provide integrated telephony and data services to staff and students of the University of Namibia

Specific objectives for the project include the following:

- to determine current university physical infrastructure
- to investigate current behaviour in terms of data and telephony usage by staff and students
- to ascertain staff and student knowledge level of VoIP and OSS awareness
- to evaluate and recommend a suitable model for sustainability of the specified VoIP model.
- to deploy and customise an OSS VoIP system using OSS Asterisk PBX, so as to provide voice and data services to staff and students

1.4 Outline of the Thesis

The rest of the thesis is presented in five chapters.

Chapter 2 presents the literature review carried out for the purpose of this research project.

Chapter 3 discusses the research methodology.

Chapter 4 shows the research findings.

Chapter 5 illustrates the implementation of the VoIP prototype system using Asterisk PBX in the University of Namibia local area network (LAN).

Chapter 6 presents a conclusion and a discussion on the practical implication of research findings and suggestions for further research.

A list of appendixes is also included, as follows:

Appendix i – Staff and students situation at UNAM

Appendix ii – Data Network Diagram at UNAM

Appendix iii – Questionnaire

Chapter 2 Literature Review

2.1 Introduction

This chapter reviews the literature relevant to this research and presents some theory and operation of VoIP technology. OSS systems and how they are utilized to support VoIP technologies and implications are also reviewed in this chapter. A review on past projects in the area under the scope of this research has also been presented.

2.2 Definition and Background of Voice over Internet Protocol

VoIP is sometimes also referred to with the following names: IP Telephony or Internet telephony [Camp 2003]. It refers to some category of hardware and software that enables people to transmit voice and other forms of information or services over a packet switched network. These comprise of the associated protocols, mechanisms, and applications that allow voice that have been traditionally transmitted over dedicated circuit-switched networks to be transmitted via IP data networks [Walker and Hicks 2004, and Liesenborgs 2000].

The process of transmitting voice over the Internet involves steps such as sampling the voice from one end, compressing it, assembling it into IP packets, and transporting them across a data network to the destination of

the receiver. The receiving end disassembles the packets, decompresses the signals and audible signals are produced through devices such as speakers [Walker and Hicks 2004]. Figure 2 demonstrates this concept.

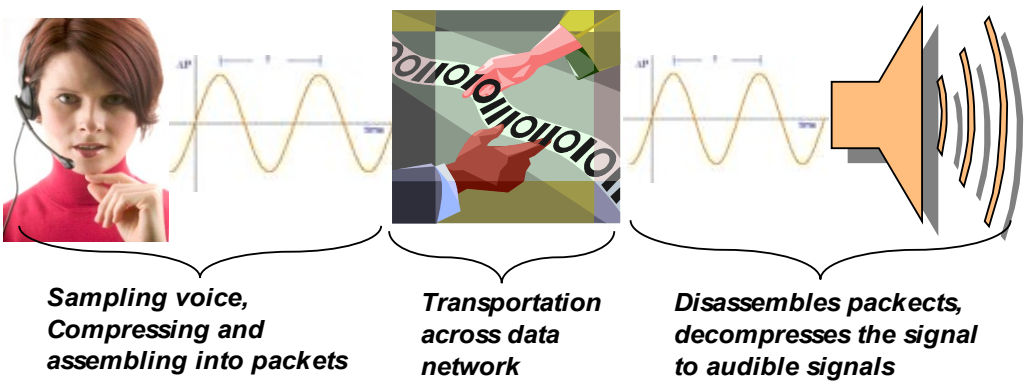


Figure 2 : Voice transmission from one end to another

There are a number of standards that are commonly used including the Internet Engineering Task Force's (IETF) Session Initiation Protocol (SIP) [Handley et al. 1999, Charter 2003b, Rosenberg and Shockey 2000] and the International Telecommunication Union's (ITU) H.323 [ITU-T 1998] protocols that have made the processes and mechanisms of VoIP and transporting voice over IP networks quite stable.

According to IEC Tutorials, the idea of voice communications using the Internet and not the PSTN became popular in 1995 when Vocaltec, Inc introduced its Internet phone software [IEC Tut. 2005]. It was designed to run on a 486/33-MHz personal computer with a sound card, speakers, microphone and a modem (see Figure 3). It was an example of a PC -to -PC

Internet phone set up which only works if both parties are using Internet phone software.

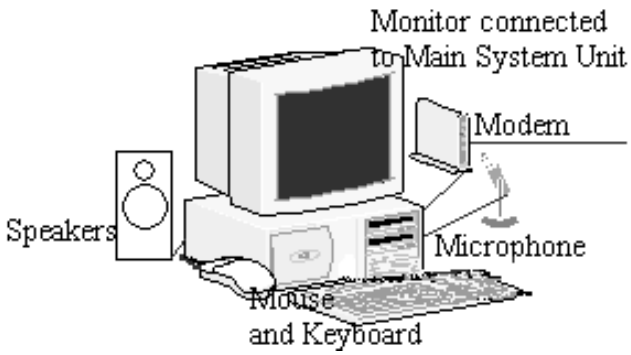


Figure 3: Basic PC configuration for VoIP

Within a short time span many software developers have provided PC telephony software with gateways to act as an interface between the Internet and the PSTN.

These gateway servers equipped with voice-processing cards enable users to communicate via standard telephones as depicted in figure 4 below.

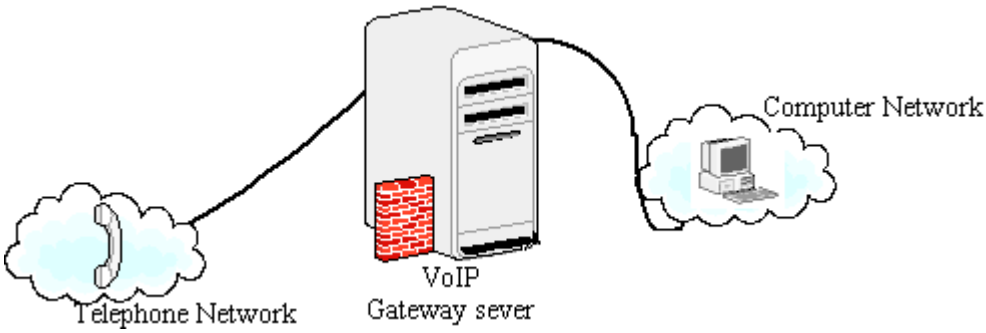


Figure 4: The Topology of PC –to-Phone

A caller is able to make a call from the traditional PSTN to the nearest gateway server, which samples the analog voice signal, compresses it into IP

packets, and moves it onto the Internet for transportation to a gateway at the receiving end. The gateway servers provide interconnection between different networks.

IP telephony or VoIP has attracted quite a number of users now because of the following reasons:

- Costs: Its ability to offer much lower tariffs compared to PSTN. Users can bypass the traditional PSTN by routing their voice traffic over the Internet for fixed monthly Internet-access fee plus local telephone connection charge.
- Integration/Convergence: It is easier to put together services and features for voice and data for the people connected to the VoIP network.
- Equipment/Hardware requirements: For new installations, there is less cabling and equipment required because all internal calls can be made using the LANs unlike a situation where there are two separate networks for data and telephony networks.

2.3 Components of Voice over Internet Protocol

The mechanism of VoIP requires basic components to be configured in order to enable its full functionality [Walker and Hicks 2004]. These components are categorised as follows:

- Codecs
- Transmission Control Protocol/Internet Protocol (TCP/IP) or VoIP protocols

- IP telephony servers or PBXs
- VoIP gateways or routers
- IP phones and softphones.

2.3.1 Codecs

A Codec can either mean compressor-decompressor or coder – decoder. This could be hardware or software with a purpose of performing transformations on data streams or signal from analog to digital and vice versa so that it can be transmitted over a networked interconnection. They are recommended by the ITU's Telecommunication Standardization Sector (ITU-T) and named by the letter G, a dot followed by a number (Say G.729). Essentially there are a number of codecs available with varying characteristics such as speed and quality of the output [ITU-T 2007]. Table 1 lists some of the most common Codecs [Walker and Hinks 2004]. The nominal data rate indicate the speed at which data is transformed and the packetization delay is the delay that the codec generates when performing the transformation of analog signals to digital and vice versa.

Codec Name	Nominal Data Rate	Packetization Delay
G.711μ	64.0 kbps	1.0 ms
G.711a	64.0 kbps	1.0 ms
G.726-32	32.0 kbps	1.0 ms
G.729	8.0 kbps	25.0 ms
G.723.1 MPMLQ	6.3 kbps	67.5 ms
G.723.1ACELP	5.3 kbps	67.5 ms

Table 1: Some common Codecs in VoIP

2.3.2 TCP/IP

TCP/IP is the name given for a suite of protocols that were developed by U.S. DoD during the late 1970s in order to facilitate communication between dissimilar computer systems at research institutions [Stevens 1993]. It is similar to the OSI model except that it represents more of the internet operations rather than the OSI which is mainly for reference purposes. The suite consists of two important protocols namely the TCP and the IP.

2.3.2.1 Internet Protocol

The IP Version 4, defined at the network layer (Layer 3) of the OSI model and in RFC 791 provides a connectionless, best effort packet switching. It is also performs fragmentation and reassembly of datagrams to support data links with different maximum-transmission unit (MTU) sizes [Malone and Murphy 2005]. This however helps in determining how datagrams are transferred across the IP network from the sending and receiving hosts. The fragmentation results into fragments called packets.

IP Version 4 addresses, IPv4 has inadequate address space. Due to this fact there has been a new development by introducing Version 6 which is currently yet practically implemented on the Internet [Malone and Murphy 2005].

2.3.2.2 Transmission Control Protocol

The TCP is a connection-oriented transport layer (Layer 4) protocol of the OSI reference model [ISO/IEC 1994]. TCP provides services such as stream data transfer, reliability, efficient flow control, full-duplex operation, and multiplexing. In performing stream data transfer, TCP delivers unstructured stream of bytes identified by sequence numbers, which are grouped into segments and passed on to the IP for delivery [Malone and Murphy 2005].

TCP offers reliability by providing connection-oriented, end-to-end reliable packet delivery through an Internetwork. This is accomplished by the process of sequencing of bytes with a forwarding acknowledgment number that indicates to the destination the next byte that the source expects to receive. In a case where some bytes are not acknowledged within a specified time period, they are retransmitted. This mechanism allows TCP enabled devices to deal with lost, delayed, duplicate, or misread packets. There is also a time-out mechanism that allows devices to detect lost packets and request retransmission.

Based on the operation of TCP, if an interface using TCP is to be used in making calls, it will mean that the sending application has to ensure that the receiving application has received everything that has been sent.

Together, the TCP and the IP represent the centre of the Internet protocols and therefore makes it a major component in the transmission of voice (assembled in packets) over the Internet.

2.3.2.3 User Datagram Protocol

Another important protocol in the Internet protocol family is the User Datagram Protocol (UDP) defined at transport-layer protocol (Layer 4) of OSI reference model. UDP protocol ports distinguish multiple applications running on a single device from one another. As compared to TCP, UDP provides no reliability, flow-control, or error-recovery functions to IP. This results in its simplicity where headers contain fewer bytes and consume less network overheads than TCP. Therefore this makes it more useful in situations where the reliability mechanisms of TCP are not necessary, such as in situations where a higher-layer protocol might provide error and flow control.

2.3.3 VoIP Protocols

A protocol is a convention or a standard that controls or enables the connection, communication, and data transfer between two computing end points and can be implemented in hardware, software or a combination of the two of which at the lowest level defines a hardware connection [Black 2000].

Most protocols specify one or more of the following:

- Detection of the underlying physical connection (wired or wireless), or the existence of the other endpoint or node
- Handshaking
- Negotiation of various connection characteristics
- How to start and end a message
- How to format a message

- What to do with corrupted or improperly formatted messages (error correction)
- How to detect unexpected loss of the connection, and what to do next
- Termination of the session or connection

Protocols that are used to transmit voice signals over the IP network are generally referred to as Voice over Internet Protocols or VoIP protocols. When making a call on the VoIP terminal, application programs that are based at the higher level are used. These programs have to interact with lower levels of the TCP/IP stack. For the purpose of providing telephony services, there is a need that a number of different standards and protocols come together. For instance, Real-time Transport Protocol (RTP) [Force 1996] to ensure transport, Remote Authentication Dial In User Service (RADIUS) to authenticate users, Lightweight Directory Access Protocol (LDAP) to provide directories and the , Resource Reservation Protocol (RSVP) to be able to guarantee voice quality and to inter-work with today's telephony network.

When initiating and completing a call on a VoIP terminal(s) into the network, protocols are required to facilitate call setup and streaming of voice. These protocols are classified in two categories namely call setup protocols and Voice streaming protocols.

2.3.3.1 Call setup Protocols.

When setting up a call using VoIP, application programs use protocols that utilize either TCP or UDP to transfer data during the initial phases of the telephone call. The call setup protocols perform functions that facilitate calls. Examples of these functions are mapping of phone numbers to IP addresses, generating dial tones and busy signals, ringing the called phone and hanging up. There are four common types of these protocols: The H.323 and Media Gateway Control Protocol (MGCP) developed by ITU community [Arango et al. 1999, Greene et al. 2000] and the SIP and Media Gateway Control (megaco) [Greene et al. 1999, Charter 2003a, ITU-T 2000] developed by IETF community.

H.323, defined by RFC 3508, sits on top of the basic Internet Protocols such as IP, TCP, UDP and RTP and is able to make integrated and differentiated services along with the resource reservation protocols. It provides audiovisual communication sessions on any packet network. Its implementation has involved real time applications such as NetMeeting and GnomeMeeting. It is part of the H.32x series of protocols that deal with communications over Integrated Services Digital Network (ISDN), PSTN or Signalling System Number 7 (SS7) and is commonly used in VoIP systems.

Media Gateway Control Protocol (MGCP) is used within the VoIP systems. It is used for controlling gateways from external call control elements called media gateway controllers or call agents. A telephony gateway is a network

element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks. MGCP assumes a call control architecture where the call control intelligence is outside the gateways and handled by external call control elements. The MGCP assumes that these call control elements, or Call Agents, will synchronize with each other to send coherent commands to the gateways under their control. MGCP is, in essence, a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents.

SIP has a standard RFC 3261 and was developed for initiating, modifying, and terminating interactive user sessions that involves multimedia elements such as video, voice, instant messaging, online games and virtual reality. Currently, SIP, which is still evolving, is one of the leading signalling protocols. SIP also implements advanced features that are in SS7. Due to the fact that it is a peer to peer protocol makes it possible for simple implementations because two SIP end points can communicate without any intervening SIP infrastructure. A session could be a simple two-way telephone call or it could be a collaborative multimedia conference session. SIP is a client-server protocol that closely resembles two other Internet protocols such as Hyper Text Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP) [the protocols that power the World Wide Web and email] and sits along side Internet protocols.

SIP also requires a Proxy and register network elements to work with. There are a number of these proxies or softswitches that are implemented with SIP. An example is the Asterisk softswitch.

Megaco is a signalling protocol used between a Media Gateway and a Media Gateway Controller. It is referred to as an IETF RFC 2885 and defines a centralized architecture for creating multimedia applications, which includes VoIP. In many ways, it is said to be an extension of MGCP.

2.3.3.2 Voice streaming protocols

The most widely used protocol for streaming is the RTP. It is the standardization of packet format for delivering audio and video over the Internet. It was originally designed as a multicast protocol by IETF and published in 1996 as RFC 1889. It is used in streaming media systems in conjunction with Real Time Streaming Protocol (RTSP), video conferencing and also in conjunction with signalling protocols such as SIP and H.323. In the TCP/IP communication protocol stack, RTP sits on the transport layer together with TCP and UDP. The VoIP packets are basically contained in the UDP due to the fact that UDP does not have a heavy pay load on the transmission of packets. Referring to RFC 1889, RTP provides services such as payload-type identification, sequence numbering, time stamping and delivery monitoring. It is thus instrumental in the solving of the problem of enabling the receiver to re assemble the packets from the source back into the correct order and without too much delay due to the fact that it does not have to wait for lost packets. This guarantees quality of service as long as

there is continuity and order. RTP was designed specifically for applications that sent data and need no acknowledgements. Hence, its datagram header structure has two most important attributes for identification which are a time stamp and sequence number for orderly reconstruction of packets and dealing with duplicates, missing or out of order datagrams respectively. Figure 5 below shows the two attributes.

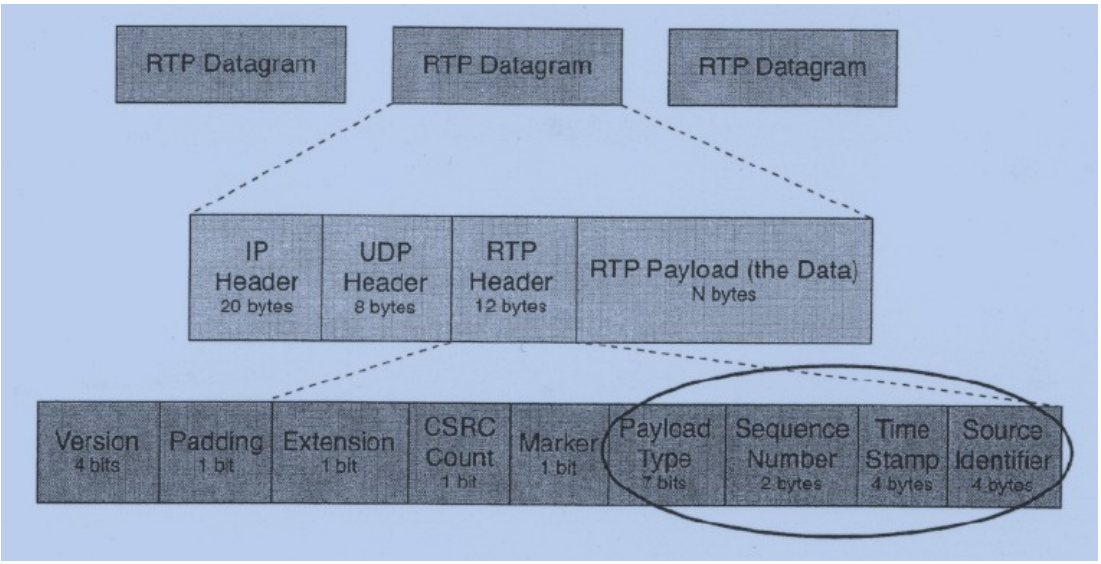


Figure 5: Time stamp and sequence number attributes in the RTP datagram. (Walker and Hicks 2004)

2.3.4 IP Telephony Servers and PBXs

A server is usually a computer running an application that manages the setup or connection of telephone calls between terminals. It registers terminal's IP addresses and stores them for the purpose of connecting calls. The server will receive call setup request messages, determine the status of destination devices, check the authorization of users to originate and/or receive calls, and create and send the necessary messages to process the call requests.

The VoIP network requires a client - server topology where in this case IP PBX server is the main telephony server. An IP PBX is a private branch exchange (telephony switching system within an enterprise) that switches calls between VoIP users on local lines while allowing all users to share a certain number of external phone lines. The typical IP PBX can also switch calls between a VoIP user and a traditional telephone user, or between two traditional telephone users in the same way that a conventional PBX does. The abbreviation may appear in various texts as IP-PBX, IP/PBX, or IPPBX.

With a conventional PBX, separate networks are necessary for voice and data communications. One of the main advantages of an IP PBX is the fact that it employs converged data and voice networks. This means that Internet access, as well as VoIP communications and traditional telephony communications, are all possible using a single line to each user. This provides flexibility as an enterprise grows, and can also reduce long-term operation and maintenance costs. Like a traditional PBX, an IP PBX is owned by the enterprise. In VoIP systems, IP PBXs are normally built on a PC platform running on any operating system. An example of an IP PBX is the Asterisk which is built and runs on Linux operating system. These IP PBXs provide functions and features equivalent to the traditional PBXs of the PSTN.

These IP telephony servers can be clustered in a group and managed as a unit in order to increase scalability, reliability and redundancy. H.323 protocol

uses the Gatekeeper to provide call admission control (CAC) and other management functions such as address lookup for multimedia services.

2.3.5 VoIP Gateways, Routers and Switches

Gateways are end points that make it possible to connect call between end points that would normally not inter operate. They usually translate from one signalling protocol to another such as from SIP protocol to ISDN protocol and also translating of network addresses between different network addressing schemes. The gateways make it possible to interface VoIP and the traditional PBX.

In order to move RTP voice datagrams, you need to have VoIP gateways set. VoIP gateways provide a link between the VoIP network and the traditional PSTN network making it possible to make a call to telecoms lines. The VoIP gateways use SS7 protocol to signal switches in the PSTN network when a call originates from the VoIP network and the called is in the PSTN network. VoIP gateways also provide what is called transcoding, a process of converting different codecs.

Usually a router, configured to route IP packets are connected to at least two networks, commonly two LAN or Wide Area Networks (WAN) or a LAN and its Internet Service Provider's network. Routers are located at gateways, the places where two or more networks connect. Routers use headers and forwarding tables to determine the best path for forwarding the packets, and

they use protocols to communicate with each other and configure the best route between any two hosts. Switches are also used in VoIP networks in order to filter and forward packets between LAN segments. Switches operate at the data link layer (layer 2) and sometimes the network layer (layer 3) of the OSI Reference Model and therefore support any packet protocol.

2.3.6 IP Phones and Softphones

This is the end point of communication which is usually in form of hard phone or a soft phone. These are referred to as answering machines and they are identified by an IP address which is capable of handling many terminals for the same purpose. The one that is enabled first completes the call and others become disabled.

From inception of VoIP, computers have been used as terminals although currently telephone adaptors and or VoIP telephones are available. These are also compatible with cordless and wireless configuration.

2.4 Quality of Service

According to Hardy, a principal analyst for quality measurement and analyses at WorldCom, Quality of service (QoS) is fundamental to the operation of VoIP network [Hardy 2003]. In VoIP, quality of service refers to the probability of the telecommunication network meeting a given traffic contract, or in many cases is used informally to refer to the probability of a packet succeeding in passing between two points in the network.

Initially when the Internet was created, a need for the quality of service was not emphasised due to the fact that the entire network ran on the “best effort system”. However, there can be many factors that can affect packets as they move from the source to destination on the network [Vagesna 2001]. These can arise as results of dropped packets by the routers, delays in reaching the destination, jitter which is a variation of delays as packets move to the destination, out-of-order delivery caused by packets arriving out order due to jitter, and error which results in some corrupted packets.

There are essentially two ways to provide quality of service guarantees. The first is to provide lots of resources, enough to meet the expected peak demand with some safety margin. The second is to require people to make reservations for use of the services.

In VoIP, differentiated quality of service is used. IP precedence utilizes the 3 precedence bits in the IPv4 header's Type of Service (ToS) field to specify the class of service for each packet. You can partition traffic in up to six classes of service using IP precedence (two others are reserved for internal network use) [Mueller 2002]. The queuing technologies throughout the network can then use this signal to provide the appropriate expedited handling. The 3 most significant bits (correlating to binary settings 32, 64, and 128) of the Type of Service (ToS) field in the IP header constitute the bits used for IP precedence. These bits are used to provide a priority from 0 to 7

(settings of 6 and 7 are reserved and are not to be set by a network administrator) for the IP packet.

Because only 3 bits of the ToS byte are used for IP precedence, you need to differentiate these bits from the rest of the ToS byte.

RFC 2475 extends the number of bits used in the ToS byte from 3 to 6. The 6 MSBs will be used for precedence settings (known as DS codepoints), with the 2 least significant bits (the right-most 2 bits) reserved for future use. This specification is commonly referred to as DiffServ.

2.5 Internet connection requirement for VoIP

VoIP offers tremendous opportunities and challenges for telecommunications services, but two key performance factors that need to be considered for successful implementation are bandwidth utilization and packet handling.

Ideally, an institution should have a broadband connection (> 128 kpbs) [Unuth 2007]. The reasons for this requirement are as follows:

- You need at least 64 kbps download plus 64 kbps upload bandwidth to make an uncompressed VoIP call. Otherwise you will need to compress your VoIP traffic using compression codecs such as G.723 or G.729 which can decrease the bandwidth to approximately 20 kbps. However, such methods of compression are usually used in situations where

multiple calls are going over the same connection, not when bandwidth is restricted to less than 128 kbps.

- A broadband connection is usually 'always-on' thereby ensuring that you can receive and make calls with minimum interruption.

- Depending on the usage of the Internet connection, there is need also to make sure there is enough bandwidth leftover for typical Internet traffic (eg emails, web browsing, file downloads).

2.6 Firewall and Traversal Techniques

Due to limitations of the IPv4 32 bit address space, Network Address Translation (NAT) is another important aspect to consider during implementation of VoIP systems in institutions like the University of Namibia where users are behind the NAT firewall.

The following four are the different types of NATs that are implemented:-

- full-cone
- address restricted cone
- port restricted cone and
- symmetric NAT

VoIP however presents a problem for NAT because when IP Phones or routers establish VoIP call signaling, call control, and media communications,

the IP address and port number are embedded within the data payload of the IP packets. This causes end-to-end routing problems between the end points if the embedded IP addresses are private IP addresses.

There are several methods to traverse these NATs which include Application Layer Gateways (ALGs), media tunnels, third party proxies, or simple transversal of UDP through NAT (STUN).

2.7 Planning for VoIP Deployment

Deployment of new applications and technologies in an enterprise network is always a challenge for all the network administrators and managers. Before deployment, there is need to carry out good planning, analysis and assessment of the current communication and network environment. Then an evaluation and purchase of hardware and software is conducted based of the assessment of the current environment [Walker and Hicks 2004].

In cases where consideration is being made to put together voice and data on the same network, a need to ensure that the existing networks can take on this additional load is necessary. This will require an analysis of the current network for congestion and a plan for bandwidth in the case of WAN links.

The planning stage of VoIP deployment requires information about existing telephone usage, reliability indicators, call quality determination and bandwidth calculation.

2.7.1 Telephone usage

Telephone usage refers to the main characteristics of telephone calls that pass through the existing phone system. This includes determination of current call volumes, profile of these calls, frequency, duration, location and call flow. In traffic theory, you measure traffic load which is the ratio of call arrivals in a specified period of time to the average amount of time taken to service each call during that period. These measurement units are based on Average Hold Time (AHT). AHT is the total time of all calls in a specified period divided by the number of calls in that period, as shown in the following example:

(3976 total call seconds)/(23 calls) = 172.87 sec per call = AHT of 172.87 seconds

A busy hour of any given day in a standard business environment, accounts for approximately fifteen to twenty percent of the traffic for that day. An approximation of 17 percent of the total daily traffic is normally used in computations to represent the peak hour traffic. In many business environments, an acceptable AHT is generally assumed to be 180 to 210 seconds. Network capacity measurements are made in conjunction with an AHT to derive a BHT that can be used for traffic analysis [Cisco 2007].

In the telephony industry, the busy hour traffic is often calculated in erlangs [Boucher 1992]. An Erlang is a unit of telecommunications traffic measurement that represents the continuous use of one voice path. In practice, it is used to describe the total traffic volume of one hour.

For example, 60 calls in one hour, each lasting 5 minutes, results in the following number of erlangs:

Minutes of traffic in the hour = Number of calls x Duration

Minutes of traffic in the hour = 60 x 5

Minutes of traffic in the hour = 300

Hours of traffic in the hour = 300/60

Hours of traffic in the hour = 5

Traffic figure = **5 erlangs**

Traffic measurements made in erlangs assist telecommunication network engineers to understand traffic patterns in voice networks. They can also be made to estimate how many lines are required between the telephone system and the Telecom's PSTN or between multiple network locations. Erlang calculations can further be broken down as follows:

- Erlang B - the most commonly used traffic model. Erlang B is used to work out how many lines are required if the traffic figure during the busiest hour is known. This model assumes that all blocked calls are cleared immediately.
- Extended Erlang B - similar to Erlang B, this model can be used to factor in the number of calls that are blocked and immediately tried again.

- Erlang C - this model assumes that all blocked calls are queued in the system until they can be handled. Call centers can use this calculation to determine how many call agents to staff, based on the number of calls per hour, the average duration of calls and the amount of time calls are left in the queue

An erlang value of 1 means the telephone line is 100% busy. Simple calculators are available on Internet [Erlang 2007] that implement erlang B calculation to allow modeling scenarios of different statistics. Erlang b can be used to tabulate one of the following factors, given the other two:

- Busy Hour Traffic (BHT) - the number of hours of call traffic during the busiest hour of operation
- Blocking – the percentage of calls that are blocked because not enough lines are available
- Lines – the number of lines in a trunk group.

Determining telephone usage also involves call flow analysis that provides useful statistics on distribution of calls within an organization as well as calls that travel to and from the PSTN. The source of information about call statistics can be taken from the Call Detail Records (CDRs) that can easily be accessed at most PBXs as softcopies.

2.7.2 Reliability

Users expect highly available telecommunications services with high-quality voice as they have been experiencing with PSTN services. Networks with VoIP need reliable high-performance networks to meet user expectations, and must be able to guarantee performance and reliability to their customers. In converged voice and data networks, the network infrastructure must deliver very high quality and availability for some customer needs, while also providing low-cost high-capacity bandwidth for other needs.

Individual components that make up the network should be examined. Cisco identifies the following as major availability factors [Lambert 2001]:

- Hardware reliability;
- Software reliability and features;
- Network link and carrier reliability;
- Environment and electrical power;
- Network design; and
- User errors and process management.

The use of quality of service mechanisms to provide prioritization for various traffic types is a key element needed for voice and data network convergence. However, it is not sufficient if the underlying networks are unreliable

2.7.3 Call Quality

Traditional methods of monitoring network traffic which focus on isolated traffic statistics cannot be used on VoIP. The reason being that these statistics give little insight into call quality as they disregard burst losses, miss jitter buffer discards and do not incorporate the perceptual effects of network impairments. VoIP quality depends on the clarity of speech or the end user's perception [Szigeti 2004].

Network congestion brings about packet loss that tends to be a major course of voice signal. When codecs are used to encode/decode digitally sampled audio signals, they try to mask packet loss by replaying the last packet, interpolating from previous packets or adding noise. These packet-loss-concealment techniques suffice when packets are lost individually or at random. But they are fairly ineffective with burst loss, in which much more signal is lost. Packet loss can also occur at different rates through out the course of a call.

Data networks before VoIP are customarily tuned to make network applications, such as web transactions, e-mail, and ERP, run really well. They are based on characteristics such as sending data using TCP protocol and transaction oriented way of sending requests and responses. The key performance measurements of these types of networks are throughput and response time [Szigeti 2004].

However, when VoIP is incorporated, additional requirements to provide good quality, voice traffic places a new set of demands on data networks.

Voice has real-time characteristics, which have very strict requirements for network performance. Voice applications have two characteristics that require real-time network performance. These are sending data using RTP and interactive conversations that cannot tolerate large delays.

When a converged network is tuned correctly, many types of applications can coexist and perform well. Otherwise the converse is also true. The fundamental network performance measurements for voice traffic are delay, jitter, and packet loss.

Ever since the telephone was invented, call-quality testing has usually been subjective. The leading subjective measurement of voice quality is the mean opinion score, or MOS, as described in the International Telecommunications Union (ITU) recommendation P.800 [ITU-T 1996]. The MOS can range from 5 down to 1, using the rating scale in Table 2 below.

MOS	**Quality Rating**
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

Table 2: Mean Opinion Score Scale
Source: [ITU-T 1996]

A MOS of 4 or higher is generally considered toll quality while a MOS below 3.6 results in many users who are not satisfied with the call quality. Although

MOS is a subjective measurement, considerable progress has been made in establishing objective measurements of call quality. Various standards have been developed. A new type of intelligent call-quality monitor based on the International Telecommunication Union's E-Model [ITU 2004] for estimating voice quality is designed to accurately model packet-loss distribution and end-user perception, and correlate this with codec type and delay, to give a single score. This information is important for network managers monitoring a VoIP service-level agreement. This addressed shortcomings of the ITU recommendation G.107 [ITU-T 2003]. The output of an E-model calculation is a single scalar, called an *R-value*, derived from delays and equipment impairment factors. Once an R-value is obtained, it can be mapped to an estimated MOS. The R-value ranges from 100 down to 0, where 100 is excellent and 0 is poor. Figure 6 below shows the mapping between R-values and estimated MOS.

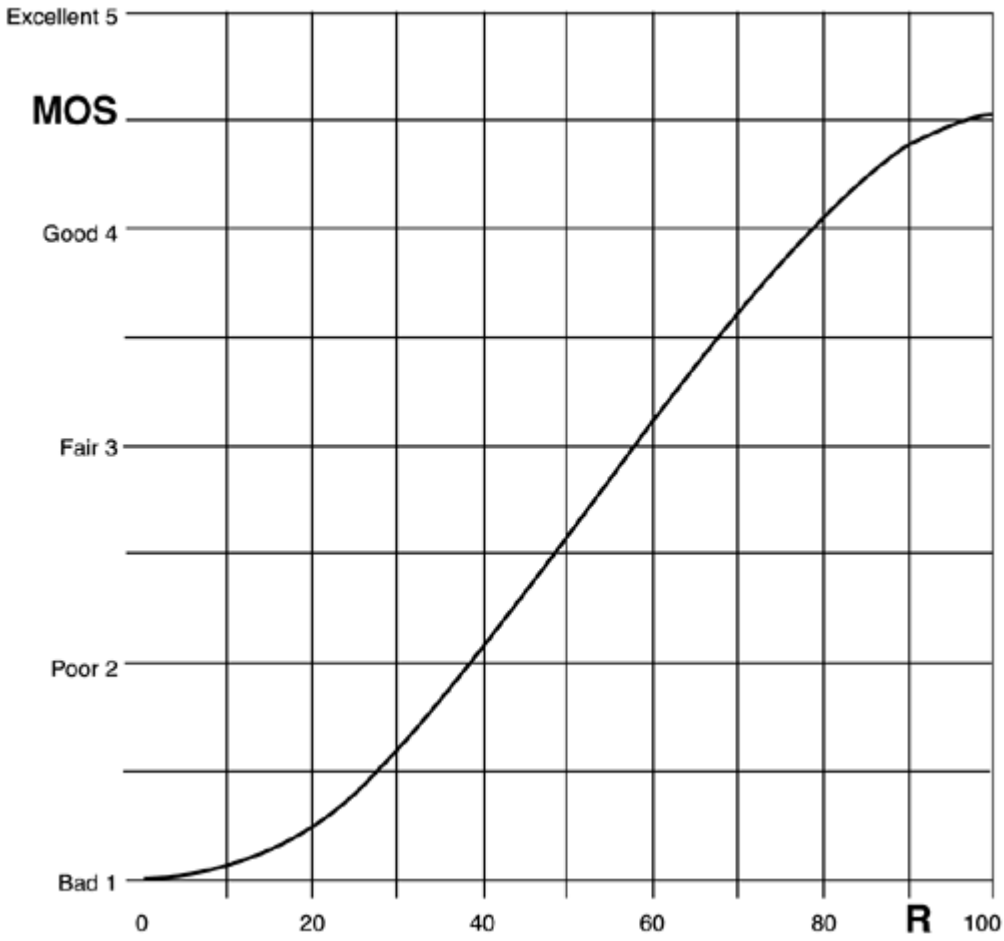


Figure 6: Mapping between R-values and estimated MOS
Source: ITU-T G.107 Recommendation

However, the intrinsic degradation that occurs when converting an actual voice conversation to a network signal and back reduces the theoretical maximum R-value (a value with no impairments) to 93.2, so the highest possible MOS is 4.4. Therefore, the R-value range from 0 to 93.2 maps to a MOS range of 1.0 to 4.4.

E-model makes particular sense for use in the VoIP readiness assessment of a data network. Given data-network statistics an R-value can be calculated as follows:

$R = R_0$

This represents an R-value which starts with the unadulterated signal. With no network and no equipment, quality is perfect. Otherwise, with simultaneous impairments to the signal, delays introduced from end to end and the equipment impairment, the signal quality reduces as it travels from end to end. The formula below illustrates how an R value can be calculated based on these factors including the advantage factor.

$R = R_0 – I_s – I_d – I_e + A$

where:

I_s— Simultaneous impairments to the signal.

I_d— Delays introduced from end to end.

I_e— Impairment introduced by the equipment, including packet loss.

A— The advantage factor. For example, mobile users may tolerate lower quality because of the convenience. Set to 0 in most models and assessments.

(Source: ITU-T G.107 Recommendation)

Codecs also play a major role in VoIP call quality. Table 3 shows the codec default values for the six most commonly used codecs in VoIP.

Codec	Nominal Data Rate (kbps)	Typical Datagram Size (ms)	Codec Impairment Factor [Range 0 - 50]	Bandwidth Required (kbps)	Theoretical Maximum MOS [Range 1 - 5]
G.711u	64.0	20	0	87.2	4.41
G.711a	64.0	20	0	87.2	4.41
G.726-32	32.0	20	7	55.2	4.22
G.729	8.0	20	11	31.2	4.07
G.723.1-MPMLQ	6.3	30	15	21.9	3.87
G.723.1-ACELP	5.3	30	19	20.8	3.69

Table 3: Impairment Factors for Commonly Used Codecs
Source: ITU-T G.107 Recommendation

The G.711 codec also called pulse code modulation (PCM) gives the best voice quality, since it does no compression, introduces the least delay, and is less sensitive than other codecs to datagram loss. Other codecs, like G.729 and the G.723 family, consume less bandwidth by doing compression. The fact that they use less bandwidth is good, since you can get more concurrent calls, but the compression results in reducing clarity, introduces delay, and makes the voice quality very sensitive to lost datagrams.

The codec impairments factor, shown in Table 3 above, can reduce the R factor significantly. They are added directly into the "I_e" portion of the R-factor equation. For example, using the G.723.1-ACELP codec causes 19 points to be subtracted directly from the 93.2 points available in the R factor.

Codec selection is a function of the call setup routine, handled by SIP, H.323, or a proprietary signaling protocol, but there are varying degrees to which you can tune the process. Depending on vendor, the soft PBX has some control over codec selection on a phone-by-phone basis. It is also possible to set preferences for each SIP channel as to which codecs should be used, or allowed. This can be a necessity if a particular channel is connected to the server using a low-bandwidth link. It would be prudent to limit that SIP channel to codecs that preserve bandwidth. Figure 7 shows how calls across a WAN might use G.729A, a low-bandwidth codec, while local Ethernet-based calls might use G.711.

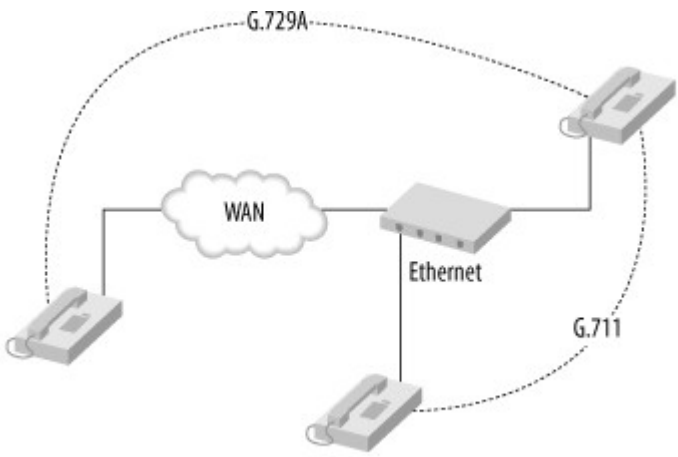


Figure 7: Possible Use of codecs on a WAN / LAN links

2.7.4 Bandwidth Calculation

The required bandwidth can be calculated using the current telephone usage statistics and hardware configuration information. The aim is to look at existing telephone usage and existing data-network utilization and determine if the current network infrastructure can support the future VoIP traffic. The simplest case is to use the projected call volumes using erlangs and codec selections using the following formula:

Bandwidth Requirement = Number of erlangs X Codec bandwidth requirement.

2.8 Open Source Software

The government of Namibia, through the Ministry of Education has recently developed an ICT policy for education, which advocates implementation of ICT to address the educational needs of the country [ME 2005]. At the same

time, there has also been a calling, world wide, for adoption of OSS by individuals, small and large enterprises, schools, universities and other public and private institutions. This has been due to advantages such as lower total cost of ownership, reduced dependence on software vendors, easier to customize and higher level of security [Scavo 2005]. This reflects a growing acceptance of and confidence in OSS and open standards that contributes to further development. Even major brands such as IBM have expressed confidence and commitment in OSS [Wladawsky-Berger 2001].

Following this trend, South Africa has officially recognized the opportunities which OSS can offer, not only as a way of saving money, but also as a stimulus to an autonomous domestic ICT sector and the development of a local ICT skills base [Bruggink 2003]. The Namibian government through Schoolnet Namiba supports the use of free, OSS to support ICT training in schools around the country [Schoolnet 2007]. The establishment of the Meraka institute in South Africa, a vision of President Thabo Mbeki, is also partially underpinned by this recognition [Meraka 2005].

According to a study report from Bridges.com [Bridges 2005], Free Open Source Software (FOSS) is software for which the underlying source code is available to the users so that they may read it, make changes to it, and build new versions of the software incorporating their changes. There are many types of OSS, mainly differing in the licensing term under which (altered) copies of the source code may (or must be) redistributed. The main idea behind open source is that programmers can read, redistribute, and modify

the source code for a piece of software, and the software evolves. OSS has for twenty years been building momentum in the technical cultures that built the Internet and the World Wide Web. Now it is penetrating into the commercial world, and that's changing all the rules of developing software systems. Following this trend an institution namely Free Software and Open Source Foundation for Africa *http://www.fossfa.net/fossfa* has been founded in Africa with a view to promote its use and maximize benefits arising from it.

OSS benefits programmers in that they experience shorter times in the development of their systems eliminating the economic loss, which results from duplicated work. This brings about the possibility of having complete products easy to localize and customize. Unlike commercial software products, which do not give access to source programs, FOSS can provide a total production-grade solution and total control in localization. Another important benefit is cooperation development that eliminates competition within the developer community. It in fact facilitates legitimate cooperation between developers and institutions. For example, the recent visit by Rhodes University computer science faculty signifies the beginning of a process of formalising partnership with University of Namibia, which will involve sustaining the ILanga open source VoIP software system.

Explored statistics in the United States by D'Antoni and Wheeler, on the use of OSS, they indicate that a significant number of companies use OSS/FS products, and some are still expecting to use it in the near future [D'Antoni 2004 and Wheeler 2003]. In Africa, van Reijswoud and Topi, indicates that

the impetus of both private organisations and governments has recently been to the promotion of open source although there are still some obstacles such as lack of skills needed to make modifications to it [van Reijswoud and Topi 2005].

Another interesting point to take note is of Weerawarana and Weeratunga which prescribes prerequisites to implementing open source systems in developing countries are advocacy and education focussed on FOSS, bearing in mind the current dominancy of proprietary software [Weerawarana and Weeratunga 2004]. With deliberate policies on ICT, initiatives at institutions such as the University of Namibia to offer education in open source will generate real-world software solutions using the open source development community. Hence, the decision to adopt OSS to support VoIP deployment at University Namibia.

To sum up, it is clear that there are a number of reasons why most of the institutions are turning to open source products although we may pick scalability and flexibility as particularly important. Scalability because these open source environments allow institutions to have as many users as they like without incurring bigger license fees; and flexibility because institutions can choose to develop the open source environment to meet their particular needs.

2.9 Sustainability of Free Open Source Software

Farmer of Georgetown University refers to sustainability of OSS as one with features of reliability, long-term support, training and documentation, active user community, enhancement synchronized with needs, reasonable costs, integration with other software, availability of trained staff and freedom to choose suppliers [Farmer 2006]. David [David 2006] also explains "sustainability" in the concept of FLOSS software development as one that focuses on three (non-funding) dimensions of;

- Sustaining commitment in FLOSS projects' development communities,
- Sustaining rate of founding of new projects
- Sustaining maintainability as functionality (and size) grows in established projects.

These two authors have brought about an economic perspective in their discussion of sustainability. David indicates that economists are intrigued by the sustainability of OSS communities, particularly the mobilization and efficient co-ordination of disparate developers working on long-term projects whilst Farmer provides a different viewpoint, stating that economic sustainability was about focusing on constancy and permanence and providing medium to long-term resource planning capability. He said that a lot of the discussion around what sustainability means can be centred on the user and their needs, as well as demands such as reliable software, long-term product support, good quality documentation and training, integration

with other systems and reasonable costs. OSS that can deliver this is much more likely to be economically sustainable.

Taylor of Open Source Consortium believes that sustainability should be considered as inter-linked on several levels: technical (level 0), commercial (level 1) through to issues of government and educational policy (level 3) and finally, widespread public acceptance at the highest level. In his view, only when the latter stage is reached can FLOSS be truly considered sustainable [Taylor 2006].

Sustainability affects every level of organization from the local neighbourhood to the entire planet. It is sometimes an evolving topic that can help us shape our future. The Brundtland Report puts it in simpler terms as "Meeting the needs of the present generation without compromising the ability of future generations to meet their needs" [Brundtland 1987].

We may summarize sustainability of open source projects by answering a question focused on community commitment, the rate of innovation through discovering new projects and managing maintainability of these projects as they add functionality. In similar projects about open educational resources, we may deal with three models of sustainability namely Funding, Technical, operational, and Staffing. Hence, it should be noted that what involves 'sustainable' will eventually depend on the economies and the objectives of the provider [Downes 2006].

2.9.1 Economic or Funding Models

These are models that originally come from governments, from foundations and organizations, and from groups and individuals. Projects involving these models must be financially supported in some kind; there are a number of them as indicated below:

*Endowment Mode*l – This model involves a project that obtains base funding then a fund administrator manages this base funding and the project is sustained from interest earned on that fund.

Donations Model – With this model, a project that is seen as worthy of support by the wider community, requests and receives donations. Donations are in turn managed by a non-profit foundation, which may apply them to operating expenses or, if amounts are sufficient, seek to establish an endowment. There are a number of open source and open content projects that are funded using this model. These include Wikipedia [Foote 2005] and the Apache Foundation [Apache 2005].

Conversion Model – as summarized by Sterne and Herring (2005) "In the Conversion model, you give something away for free and then convert the consumer of the freebie to a paying customer" [Sterne and Herring 2005] This approach has worked well in educational communities as they have been adopted by Elgg and LAMS, an e-learning software.

Contributor-Pay Model – This model consists of a system whereby contributors pay for the cost of maintaining the contribution, and where the provider thereafter makes the contribution available for free.

Sponsorship Model – This model involves some open access to stake holders through radio, television etc which can range from commercial messages to sponsorship lobbying.

Institutional Model – This is a variation based on the sponsorship model where in this case an institution assumes responsibility itself for an open source project initiative.

Governmental Model – This is comparable to the institutional model. In this case the governmental model represents direct funding for the projects through government agencies.

2.9.2 Technical Models

Has it has been seen, securing funding alone does not necessarily entail achievement of sustainability. This involves the organizations ability through its development group to develop a framework of possible target hardware, software and operating environment as well as the systems size. These technical frameworks are associated with risks such as failure to attain benefits from the project, inaccurate project cost estimates, inaccurate project duration estimates, failure to achieve adequate system performance levels and failure to adequately integrate the new system with existing hardware, software, or organization procedures [Hoffer et el 2005]. However,

for assessment purposes, these risks can also be categorised as project size, project structure, development group and user group. Table 4 indicates effects of degree of project structure, project size, and familiarity with application area on project implementation risk.

		Low Structure	**High Structure**
High familiarity with technology or application area	Large project	(1) Low risk (very susceptible to mismanagement)	(2) Low risk
	Small project	(3) Very low risk (very susceptible to mismanagement)	(4) Very low risk
Low familiarity with technology or application area	Large project	(5) Very high risk	(6) Medium risk
	Small project	(7) High risk	(8) Medium-low risk

Table 4: Effects of degree of project structure, project size, and familiarity with application area on project implementation risk
Source: Hoffer et el, 2005

2.9.3 Operational Models

Operational model entails a frame work that shows that the proposed system will solve business problems or takes advantage of business opportunities [Hoffer et el 2005]. These also address scalability issues on the project such as modification of system and adding extra modules to it.

2.9.4 Staffing Models

These are models that involve selection and hiring of staff (typically professional staff) to install and maintain the system.

2.10 Asterisk

According to Asterisk website [Asterisk 2006], Asterisk is a complete softswitch IP PBX and that was originally written by Mark Spencer of Digium. The code has been contributed from open source coders around the world, and testing and bug-patches from the community have provided invaluable aid to the development of this software.

The Asterisk website also indicates that Asterisk, is a Linux based open source and free software, which provides all of the features you would expect from a traditional PBX. Asterisk implements four VoIP protocols namely H.323, SIP, IAX, and MGCP (both gateways and phones) as well as traditional TDM technologies like ISDN (PRI and BRI) and PSTN. At the same time, it gives consistent interface to various telephone applications such as call bridging, Voicemail services with Directory, Call Conferencing, Interactive Voice Response (IVR), etc. Figure 8 [Spencer et al. 2003] shows the architecture of Asterisk which is sometimes referred to as “middleware”.

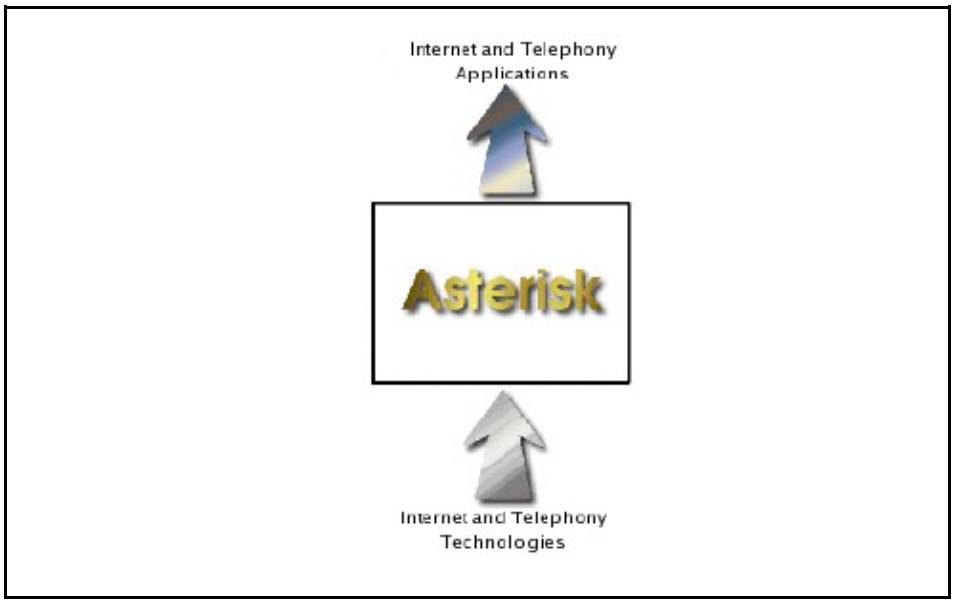


Figure 8: Architecture of Asterisk
Source: [Spencer et al. 2003]

Asterisk can interoperate with nearly all standards-based telephone equipment using comparatively inexpensive hardware. In order to interconnect with digital and analog telephone equipment, Asterisk supports a number of hardware devices, most of which are manufactured by Asterisk's sponsors, Digium. Asterisk performs multiplexing functionalities more in software and very little in hardware as referenced from the same Asterisk website.

In a slightly more detailed way, the architecture of Asterisk can be broken down into four Application Programming Interfaces (APIs): the Application API, the Codec Transition API, the File Format API, and the Channel API as illustrated in Figure 9 [Spencer et al. 2003]. These APIs are defined around an advanced, central PBX core system which handles the internal interconnection of the PBX.

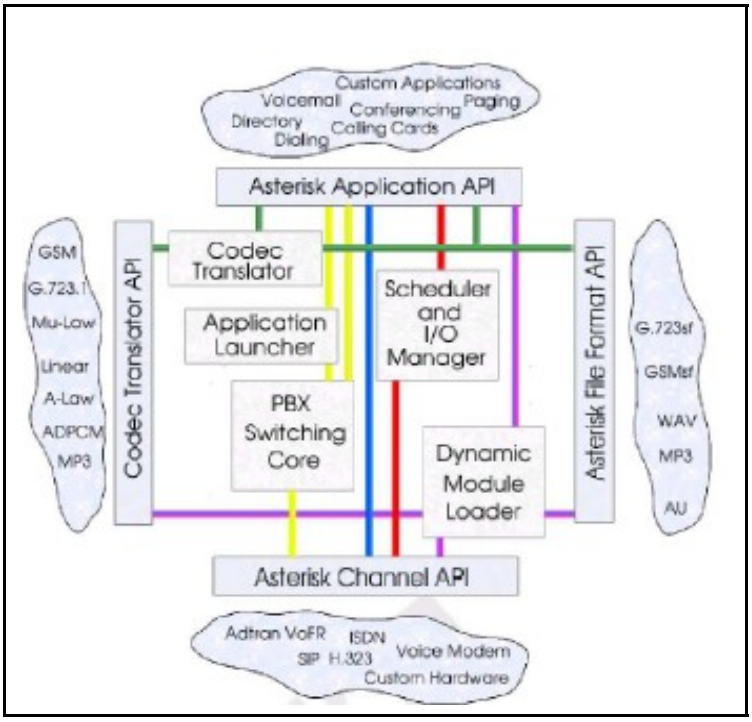


Figure 9: A Detailed Representation of Asterisk's Architecture
Source: [Spencer et al. 2003]

The Application API provides for flexible use of application modules to perform any function flexibly on demand, and allows for open development of new applications to suit unique needs and situations, for example, voicemail, conferencing, etc. The Channel API allows the Asterisk switching core to interface with different TDM or packet voice sources. VoIP protocols used include SIP, H.323, etc. The File Format API allows Asterisk to be able to read and play sound in different formats including WAV, AU, and MP3. This gives Asterisk based applications more flexibility in dealing with ring tones, DTMF, etc. The Codec Translator API provides a flexible way for the core to deal with encoded voice no matter where it is coming from. Formats like GSM, G.723, ADPCM, and MP3 are supported.

These APIs allows Asterisk to achieve a complete abstraction between its core functions as a PBX server system and the varied technologies existing (or in development) in the telephony arena. Their modularity means that every aspect of the PBX is sealed off from the rest of the system, and modifying one (for instance, adding a new telephony technology) does not require any modification of the rest.

2.10.1 Interfaces

Asterisk supports world wide networks; that is, the legacy PSTN and the Internet. An Asterisk server is capable of servicing calls to and from PSTN, or a LAN, or the Internet. It operates with traditional analog and digital telephone technologies that are supported by various hardware devices. All calls are routed through interfaces that are associated with different hardware and protocols. Each interface has its own configuration file, for example, the SIP interface is configured in the file sip.conf. Below is a list of configuration files for a standard distribution.

- SIP – Session Initiation Protocol IETF
- IAX – Inter-Asterisk eXchange protocol – v1 and v2
- MGCP – Media Gateway Control Protocol / Megaco IETF
- ZAP – Zapata channels
- Modem – Modem channels (Including ISDN)
- Skinny – Skinny channels (Cisco phones)
- Voice over Frame Relay – Adtran style
- Console – Linux OSS console client driver for sound cards /dev/dsp

- vbp – VoiceTronix Interface driver
- H.323 – H.323 ITU
- Phone – Linux Telephony channel
- Agent – ACD Agent channel

2.10.2 Channels

A channel is a connection that brings a call to the Asterisk PBX. Individual calls use a channel within an interface. It could be a connection to an ordinary telephone handset or ordinary telephone line, or to a logical call (like an Internet phone). Every call is placed or received on a distinct channel. Asterisk uses a channel driver (typically named `chan_xxx.so`) to support each type of channel. It is specified by a *technology* and a *dialstring*:

<technology>/<dialstring>

The technology is one of the installed channel modules such as SIP, IAX2, or Zap. The syntax of the dialstring varies depending on the type of channel selected.

2.10.3 The Dialplan

Asterisk switches calls between interfaces under the control of the dialplan. When a call arrives on the channel, the dialplan determines what should be done with the call, for example answering the call, connecting to another telephone, forwarding the call, or directing the call to VoiceMail, among other

choices. Ringing a telephone, dialing out over a phone line, connecting to a VoiceMail are all controlled by the dialplan.

The dialplan is configured in the flat file /etc/asterisk/extensions.conf or alternatively in a relational database. The content of "extensions.conf" is organized in sections, which can be either for static settings and definitions or for executable dialplan components in which case they are referred to as **contexts**. The settings sections are **general** and **globals** and the names of contexts are entirely defined by the system administrator. A special type of contexts is macros, labelled by a userdefined name prefixed with **macro-**. These are reusable execution patterns, like procedures in a programming language. Every section in extensions.conf starts with the name of the sections contained within square brackets. This gives the extensions.conf file a similar structure to the traditional .ini file format of the Windows world.

2.11 Implementing VoIP using Asterisk

VoIP can be implemented using several methods. One way is to find a vendor who supplies a commercial software product and equipment. Another way is to implement it internally through the use of OSS such as Asterisk. Various configurations and design can then be made to it in order to customize the system.

Planning is an important aspect to consider for successful implementation of VoIP. Further, Walker and Hicks emphasises on planning, analysis and

assessment of current data and voice networks in order to make projections for the VoIP model [Walker and Hicks 2004]. This assists in determining the kind of management of hardware and software resources that would be involved for the purpose of continuity and enhancement.

One example of where VoIP has been implemented successfully is at Rhodes University Computer Science Department, where Asterisk has been successfully deployed and customised [Penton and Terzoli 2003] [Penton and Terzoli 2004]. Following its deployment, a number of research projects focused on the customization, enhancement and performance testing of iLanga had been conducted by staff and students. These research projects have enabled students to contribute to the existing environment on a daily basis and currently, the department has been involved in extension of this facility to other departments and partners. Readers can refer the following web site for further details:
http://hobbes.ict.ru.ac.za/csae/cs405/projprops/at/VoIPprojects.html, (Accessed 2005).

Another case in reference is the decision by Sam Houston State University (SHSU) to migrate from the Cisco VoIP to open source Asterisk [SHSU 2006]. The university dumped the proprietary Cisco VoIP and moved close to 6000 students, faculty and staff to Asterisk system due to the reason that it would be more cost effective to run the open source solution.

Asterisk like any other software, has a number of areas for improvement [Manesh 2004]. Some features needing improvement are as follows:

- Asterisk is still undergoing development that will incorporate Fax services.
- The user interface needs to be made more user friendly as it is currently a command line interface. Naturally, people find command line interfaces not easy to use.
- There is more work to determine performance of Asterisk since not much has been done yet.
- Asterisk gives a platform for APIs but these can only be utilized by software people and not any ordinary telecommunication expert.
- It is difficult to write SIP applications that interacts with asteriask because Asterisk does not support high level SIP APIs.

Chapter 3 Research Methodology

Scientific method or scientific process is fundamental to scientific investigation and acquisition of new knowledge based upon physical evidence by the scientific community [Bridgman 1955].

Observations and reasoning are conducted using scientific techniques in order to propose tentative explanations of observable phenomena, which can be used to answer research questions, which can follow through by accepting or rejecting. The design of this project follows this model, with experiments, and the collection and analysis of data. Its plan and structure have been deliberated so as to obtain answers to the research question and fulfil the study objectives.

The plan is the overall scheme or programme of the research while the structure is the framework, organisation or configuration of the research [Cooper and Schindler 2003]. These definitions are consistent with the one given by Trochim which refers to the strategy to integrate the different components of the research project in a cohesive and coherent way in order to try and address the central research question [Trochim 2001].

This research project follows a two stage hybrid design method where the current situation at the university has been studied using questionnaires, observations and interviews and has been described in chapter 4 and then a follow up with a prototype implementation in chapter 5.

3.1 Determination of the current situation

The current situation at the university was determined by investigating the current network infrastructure, data and telephones usage by staff and students, VoIP technology and OSS awareness and sustainability models.

3.1.1 Current university network infrastructure.

A descriptive method of approach [Lauer and Asher 1988] was used in order to obtain information concerning the current status so as to describe "what exists" with respect to the current network infrastructure at the university. The strategy was to use 'how' or 'why' questions although the investigator had little control over events but still focused on a contemporary phenomenon within some real-life context.

Unstructured interviews are usually employed in explorative research in order to identify important variables in a particular area; to formulate penetrating questions on them; and to generate hypothesis for further investigation for the sole reason that the area being entered is so unfamiliar, therefore, it is impossible to compile a schedule for interviews in such instances.

The researcher used unstructured interview approach by asking relevant questions to the staff of the Computer Centre. Mainly, the interview questions where based on the current network infrastructure and performance.

3.1.2 Data and telephone usage by staff and students

Quantitative method of approach [Creswell 2002] was used. This involved selecting a population, sampling procedure, methods of measurement and plan for data collection and analysis for the purpose of ascertaining current behaviour in terms of data and telephone usage by staff and students. Telephony and computer services available to staff and students were investigated. Details about the population, sample and instrument of data collection is as explained in section 3.3 of this chapter.

3.1.2.1 Staff and student services

Statistics of staff and student numbers were retrieved from the university's annual report of 2005. Unstructured interview approach was used in determining telephony and computer services available to staff and students.

3.1.2.2 Data and Telephone Usage

Current data and telephony traffic analysis was conducted to establish readiness of implementing VoIP. Data in form of diagrams, graphs and telephone call records were collected from the university and analysed.

The data on telephone calls was sampled from a period of one year, in which monthly averages were calculated and BHT. To get the most accurate results within the same period, 32 samples were collected to represent the BHT. This was in accordance with the ITU-T recommendations on how to accurately sample a network to dimension it properly. To avoid skewedness in the yearly data, holidays and weekends were eliminated from the analysis. Read-out periods of 60 seconds on the PSTN connections were taken as measurement for traffic. These intervals were necessary in order to

summarise the traffic intensity over a period of time. A Daily Peak Period (DPP) which records the highest traffic volume measured during a day, was used. Erlang C traffic model was used in the calculations for voice. Therefore an erlang was calculated as follows [Cisco 2007]:

Erlang of traffic = (number of calls X AHT)/3600

3.1.3 VoIP Technology and OSS Awareness

A study population can be defined as the entire congregation or a collection of all objects, cases, elements, or people that are the focus of the research, that meet the criteria the researcher is interested in studying and are accessible to the researcher as a pool of subjects for study to which the results obtained by testing a sample are generalised.

It is the complete set of units or elements from which information is to be obtained and about which inferences are to be made. An element or unit here refers to the most basic component from which data is collected and about which conclusions are made, for example the person, object or event [Bland 1999]; [Bless and Hugson-smith 2000]; [Mason et al. 1999].

For this study, the population was accessible since the researcher already had a connection to the university. It was also possible to identify enough subjects to yield useful conclusions, as the subjects were willing to participate in the research.

Katzenellenbogen et al. indicate that studying the whole population or all the individuals in the study population is impractical, unnecessary, costly and presents logistical problems [Katzenellenbogen et al. 1999].

The researcher defines a sample as a part, portion or a subset of the study's defined population that consists of a group of people selected by the researcher to represent the total population from which they are drawn for the purpose of obtaining information about the population [Bland 1999], [Bless and Hugson-smith 2000], [Cooper and Schindler 2003] and [Mason et al. 1999],.

Findings from this sample were generalised to the total study population. Generalisation here refers to the researcher's ability to relate the results from the sample to the population from which it is drawn and that the research findings can be credibly applied to a wider setting than the research setting [Bickman and Rog 1998] and [Mertens 1998].

Advantages of using samples are that they are practical, and collection of data is quick and less expensive providing good quality data. In this study a sample was selected from the staff and student population of the computer science department of the University of Namibia because they are the ones to spearhead management and sustainability of the system.

A quantitative research design approach was used in describing the level of VoIP technology and OSS awareness. A simple random sampling was used in selecting the participating elements of the population for the sample. A

total of 97 questionnaires were distributed at random to staff and students. 68 of these questionnaires were collected representing 70% sample size which was representative to carry out data analysis.

The questionnaire, which was the main instrument for data collection was designed in a closed-end style and was used to collect information about biographical data such as year of study and also to collect opinions on subjects that were identified to be aspects of consideration such as awareness etc. The reason for adopting a closed-end style was to allow the researcher to have control over the conditions under which the questionnaires were completed.

The questionnaire covered a population of sixty percent male students and forty percent female students who are mostly in the age group of between twenty years and thirty years spread through first to forth years of study.

Categorisation of some variables from the questionnaire was also done to make the analysis easier. Statistical methods were identified that enabled the researcher to summarise, organise, evaluate, interpret and communicate quantitative data. A statistical application package (SPSS Version 8) was used to analyse the collected data. This data was expressed in numbers, percentages and presented in frequency tables.

3.1.4 Sustainability models for university's VoIP model

Qualitative method [Creswell 2002] of evaluation was used. Sustainability factors such as funding, technical, operational and staffing were evaluated for the specified project.

3.2 Deployment and customizing a prototype VoIP system

A computer science method of systems development called rapid prototyping development was used [Hamblen et al. 2005]. This involved a small-scale deployment of a VoIP system to test out certain key features for a large scale design. VoIP components were reviewed and specified for the purpose of specifying a larger scale system for the university. Despite a wealth of literature that exists, it was important also to learn from the experts who have experiences in the field. The experts came from Rhodes University and facilitated the initial set up.

Before deploying VoIP, Asterisk was set up on a linux server. In this project, an implementation of Asterisk is used. Other necessary packages such as the zaptel telephone drivers and the PRI libraries (libpri) were downloaded from the Asterisk website. Necessary kernel sources were created and modified to suit the installation environment.

Configuration files were also modified to represent the dialling structure at the university. To do this, the dialling structure for the university had to be formulated.

Chapter 4 Research Findings

4.1 Introduction

This chapter has been structured in such way that shows the research findings and results from questionnaires, observations and interviews conducted for the purpose of addressing the current situation at the university. The following below are the specific objectives as specified in chapter 1 that are discussed based on the findings:

- to determine the current network infrastructure
- to investigate current behaviour in terms of data and telephones usage by staff and students
- to ascertain staff and student knowledge of VoIP technology and OSS awareness.

The Computer Centre provided information on data network statistics and the Estates Department provided data on telephone usage. The university's annual reports of up to 2004 provided information on its infrastructure, staff establishment and student enrolment statistics.

4.2 University network infrastructure

The University of Namibia consists of the main Campus in Windhoek and the Northern Campus in Oshakati. It also has centres in other regions that cater for other university services such as provision of open learning, distance and continuing education programmes.

The diagram in Appendix ii shows the physical topology of backbone data network of the main university campus. The diagram indicates that all the buildings at the campus are connected to the data network. It was found that the university does not have detailed documentation on the design of its communication system that include the backbones, routers, switches access methods and protocols used. Access to logical and physical network information was also restricted. As a result, the researcher could not clearly analyse and discuss university logical and physical configurations as factors in the context of applying a solution to the problem.

However, all university computers at main campus LAN connect to the Internet through the main router in the main distribution facility situated in the Computer Centre. The Northern campus has also the same arrangement. University regional centres have dial up connections to the Internet through Telecoms and Internet Service Providers. In all these network connections, network performance problems were observed. These problems included general Internet congestion in web browsing, downloads and data streaming.

The main university runs a traditional circuit switched PABX which connects to the Telecoms PSTN. It has a capability of managing up to 1500 users through PBXs situated in various buildings on campus and the northern campus. Other centres connect to this PABX through PSTN lines.

4.2.1 Data and Telephone usage

Investigations into the university revealed that currently there are about 654 staff members at the main campus and other campuses. Appendix i indicates distribution of these staff members. All staff members have access to the traditional telephony services that can be used to make internal and external calls. External calls are made through use of external PSTN lines and are charged by Telecomm. A PABX software is installed that redistribute telephone bills as per caller identification.

4.2.2 Staff and Student Services

According to Mr. Isaacs of Human Resources, access to phones has never been a problem even when new staff is received. However, with more expansion in the physical infrastructure, staff establishment and student population is expected rise in the near future with introduction of new faculties such as medicine and engineering.

Currently, the university enrols 6000 students across its faculties. Students do not have access to the university telephony network apart from calling from outside into the network. An investigation indicated that students have access to computers of which are mostly university computers in the computer labs and the library. However, library computers can not be used for voice services. This limits the majority of students who have no access to the computer labs.

4.2.3 Telephone Usage

A soft copy of the call detail Records was collected and telephone data for a period of one year (June, 2005 – June 2006) was analysed. It was found out that the university currently has seven PABXes based in regions as indicated in Table 5.

PABX ID	Town
HENTIS	Windhoek
KEETMA	Keetmanshoop
OGONGO	Oshakati
OSHAKA	Oshakati
PABX1	Windhoek
RUNDU	Rundu
SWAKOP	Swakopmund

Table 5: Location of PABXes of the University

4.2.3.1 Busy Hour Traffic (BHT)

Daily analysis of a monthly data collected from the sample data indicated the BHT (Above 10% traffic volume) between 08:00hrs and 12:00hrs and between 14:00hrs and 15:00hrs. Table 6 below indicates the analysis based on hourly time bands. A sample number of 224448 (84%) out of a total 268044 average monthly observations or telephone calls were considered from a sample based on the period under observation.

Time	Number of Calls	Total Duration	AHT	Traffic in erlangs
00	512	42193	82.408	11.720
01	314	20988	66.841	5.830
02	257	7809	30.385	2.169
03	259	8318	32.116	2.311
04	273	9110	33.370	2.531
05	364	13770	37.830	3.825
06	616	31669	51.411	8.797
07	5769	324351	56.223	90.098
08	25266	1639989	64.909	455.553
09	29550	2044025	69.172	567.785
10	28376	1964503	69.231	545.695
11	27794	2015556	72.518	559.877
12	24844	1841798	74.135	511.611
13	8344	695249	83.323	193.125
14	26349	1624429	61.650	451.230
15	25498	1798487	70.534	499.580
16	14748	1183829	80.270	328.841
17	3671	301917	82.244	83.866
18	111	11102	100.018	3.084
19	120	14028	116.900	3.897
20	63	9693	153.857	2.693
21	107	6119	57.187	1.700
22	1211	127241	105.071	35.345
23	32	3767	117.719	1.046
Total	**224448**	**15739940**	**70.127**	**4372.206**
Max	**29550**	**2044025**	**69.172**	**567.785**
Min	**32**	**3767**	**117.719**	**1.046**

Table 6: Analysis of Call based on 24 hourly band

Based on the BHT found from Table 6, daily observations were done on 32 days as summarised in Table 7. Average of 22 erlangs for the BHT was found and 2% blocking factor.

Time	Number of Calls	Total Duration	AHT	Traffic in erlangs	Blocked Calls
08	1191	73988	62	21	24
09	1267	85775	68	24	25
10	1285	86264	67	24	30
11	1279	89431	70	25	32
12	1083	72116	67	20	28
14	1221	69688	57	19	33
15	1123	71535	64	20	35
Average	**1207**	**78400**	**65**	**22**	**29**

Table 7: Summary of Calls Based on BHT

4.2.3.2 Call Flow Analysis

Forty seven percent (47%) of calls made were found to be internal within the university as shown in Table 8. This indicates a slightly larger percentage on calls outside the university.

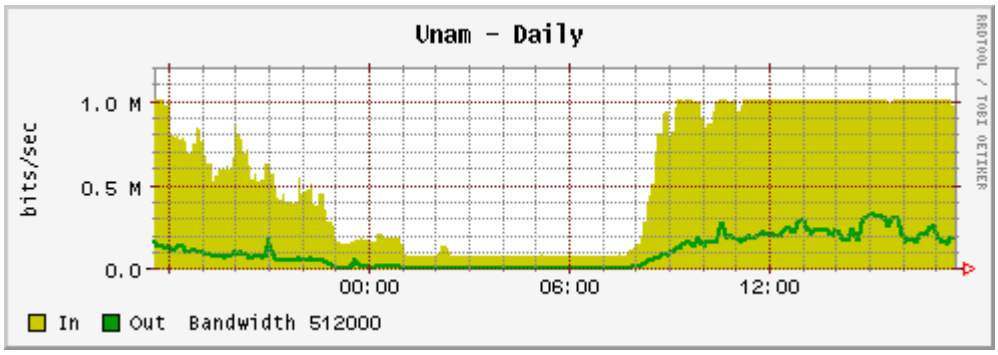
Month	Total Calls	Internal Calls	% Internal Calls
June 2005	268046	114613	43%
August 2005	29089	13243	46%
September 2005	265365	123110	46%
October 2005	272358	127411	47%
November 2005	261457	123105	47%
December 2005	118952	48394	41%
January 2006	229869	96547	42%
February 2006	300719	146428	49%
March 2006	299859	146970	49%
April 2006	228971	106507	47%
May 2006	266396	134025	50%
June 2006	243890	125829	52%
TOTAL	**2784971**	**1306182**	47%

Table 8: Call Flow Analysis

4.2.4 Data network and usage

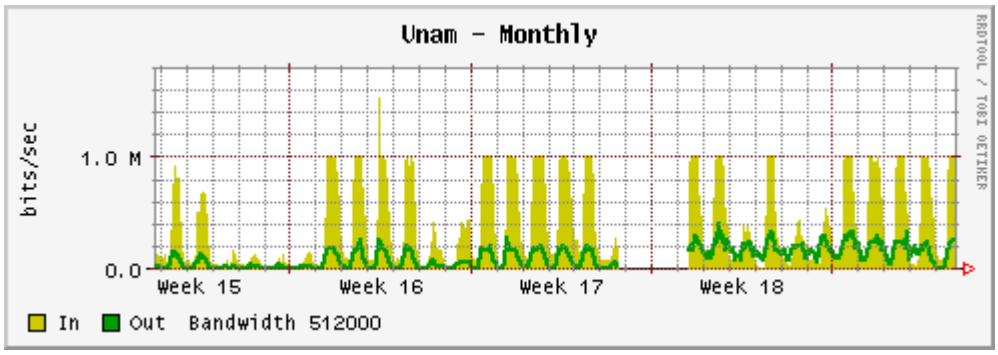
Currently, the university is provided with bandwidth of 1.536Mbps from the service provider.

The overall university daily, monthly and annual data network usage is depicted in Figure 10, Figure 11 and Figure 12 respectively.



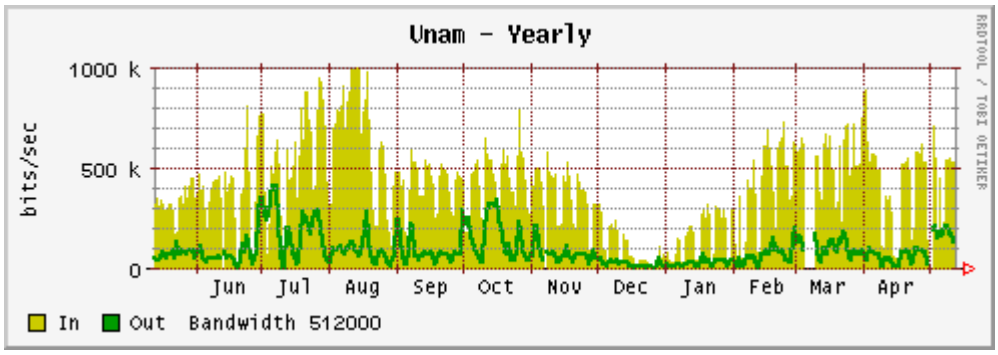
Legend	Min	Max	Avg	Last
In	71865	1013337	554525	969743
Out	5335	332336	98232	181903

Figure 10: Unam Daily Data Network Statistics



Legend	Min	Max	Avg	Last
In	4503	1539017	386874	1008770
Out	1371	392218	96895	259857

Figure 11: Unam Monthly Data Network Statistics



Legend	Min	Max	Avg	Last
In	724	998515	401228	534005
Out	411	407871	86489	133878

Figure 12: Unam Annual Data Network Statistics

The daily graph indicates peak hours starting from 09:00hrs to 17:00hrs of which is mostly dominated by the in ward traffic. Maximum usage in terms of in bound and out bound comes at 15:00hrs when there is a consumption of 1345673 bit per second (1.3Mbps). This represents 88% utilization. Monthly and yearly usage trend confirms periods when there is activity at the university.

4.3 VoIP Technology and OSS Awareness

An investigation into the computer science department's curriculum coupled with random interview with students revealed that the department does not offer a course in Real-Time Multi-Media (RTMM) and hence student demonstrated little knowledge about the VoIP technology although a few indicated knowledge of commercial products such as skype and skyphonic VoIP systems available on the Namibian market.

In 2005, during a visit on 15 August and 19 August by Professor Terzoli from Rhodes University, a one-day workshop was conducted to introduce students to the technology involved in the packetization of voice over the Internet protocol. This workshop was targeted to the third and fourth year students in the computer science department.

The study observed that about 60% students are already aware of these systems, which clearly indicates that open source software is not new to the students. Table 9 below shows results from the study.

Category	Number	Percent
Not Aware of OSS	28	41.2
Aware of OSS	40	58.8
Total	**68**	**100.0**

Table 9: Open Source Software Awareness

Out of the total number of students observed, 99% know how to program using any one of the programming languages such as C++, Java or Visual Basic. Over 60% are capable using open source programming language for various programming tasks. Statistics also show that about half of the students start developing systems using open source software from second year onwards. However, the forth year students did not show any systems development using open source software. This could be explained due to the

fact that most final projects do not involve development of systems using open source software.

The system concept has already been introduced to staff and students of the University of Namibia who eventually would be active contributors to system modifications and support in order to meet future needs. Linux which will implement the system is currently regarded as the emerging technology and hence many professionals in IT and students of the colleges and universities in computer science are already studying and learning how to use it.

4.4 Addressing issue of sustainability

The sustainability of this project will depend on models such as Funding, Technical, Operational, and Staffing.

4.4.1 Funding

After finding out facts about the university such as the infrastructure, it is evident that the university would require a large capital investment to be ploughed in to the project if it has to cover all the established faculties, campuses and centres.

By using the endowment model through which a scaled down budget would be allocated at the beginning of the project in anticipation that the savings that would stem from the project would be ploughed back into it. It is envisaged that these savings will support maintenance to the project

financially. Management of this scaled down project should be a responsibility of the computer science department through one of its specialised areas or sections.

4.4.2 Technical

The technical situation as found from the university has been analysed through a use of a matrix as indicated in Table 2 in Chapter 2 of this document. The following were the results of findings of technical factors considered based on the scaled down project:

1. The project is a relatively small project for the university. There is only one department involved which is specialised in this area and capable of acquiring necessary hardware and software needed for the project with no problems. Therefore, deployment of the system will not be a very big undertaking.
2. The requirements for the project are highly structured but can easily be obtained initially without major problems.
3. The targeted development group is not familiar with the project but have the skills to implement because they are capable of handling software and hardware problems
4. The user group is familiar with the technology as evidenced from the study and they are already using the telephones and computers.

Following the above, it can be deducted that the project can be viewed as a medium – low risk in terms of technical feasibly.

4.4.3 Operational

As indicated already, it is true that users of the project who are staff and students are familiar with telephones and the use of computers. The VoIP system will also use handsets similar to the traditional phone sets and computers will be installed with telephone applications that simulate real handsets. This indicates feasibility in terms of operations of the project.

4.4.4 Staffing

Currently, the university does not have permanent staff, or volunteers attached to VoIP projects apart from two members of staff who are involved in research. Rhodes University Computer Science Department has done quite a number of VoIP deployments and research as evidenced from their presentation.

Partnering or corroboration arrangements can be made between university and Rhodes university in which case staff and student exchanges can be made to help in the starting and continued running of the project.

The arrangement will likely stimulate research interest amongst staff and students and hence promote the project to a campus wide stage that will cover the entire institution.

4.5 Study Limitations and Constraints

The following were the limitations and constraints experienced by the author in the course of the study albeit having completed the project.

- The university Computer Centre lacks necessary telephone equipment to facilitate installation of a successfully VoIP system even for experimental purposes.
- Access to installed equipment in the computer room is restrictive and prohibits students' access. As a result of this, the author did not have full access to the installed systems to perform further tests on the systems.
- Data collection was also another big difficulty. The university data network documentation is non existent.

Chapter 5 Deployment of VoIP Prototype

The Computer Science Department of the University of Namibia does not have its own PABX; neither does it have an ISDN telephone line. Therefore, the project catered only for the LAN implementation.

The system was setup on a single 2.4GHz computer with 1GB of RAM. A 100Mbit Ethernet interface provides access to the department's data network. Linux Ubuntu server 6.10 operating system was used as a platform for the Asterisk server.

The installation involved getting Linux sources from #deb http server and installing them on the server. The following libraries were installed:

- ncurses – provides functions for creating windows in text mode
- openssl – opens source toolkit for implementing secures sockets layer protocol
- zlib – read only compressed file system emulation
- bison – generates a program for analysing the structure of text files
- autoconf and automake – gnu tools
- build-essential – list of packages for installing debian packages
- libtool - generic library support script providing interface to shared libraries
- flex – generates programs which recognise patterns in text

Asterisk packages were also retrieved and installed. These packages are:

- Asterisk – base files for the Asterisk Open Source PBX
- Asterisk add-ons – source code to various and addons to the Asterisk Open Source PBX
- Libpri – primary rate libraries to T1/E1 ISDN interfaces
- Zaptel – kernel drivers for interface cards

These sources were compiled. However, libpri and zaptel are not really functional at the moment due to the fact that the current set up does not support hosting PSTN gateway to allow SIP-PSTN calls or PSTN-SIP calls

5.1 System Architecture

The telephony system that has been installed runs the Asterisk as the main switching component. However the installed system will require an extension of SIP Express Router (SER) and OpenGK servers to cater for the SIP network and H.323 network environment respectively at a larger scale. Figure 13 illustrates the current pilot system.

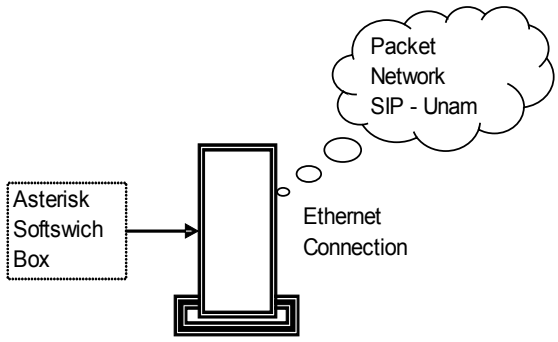


Figure 13: Asterisk main components of the pilot project.

Its design allows users to utilize its services independent of the network where they are and provides interoperability of different communication protocols and networks. Asterisk allows translation at call signalling layer as well as the media layer. Due to the fact that the university is wide and involves many departments and campuses, the Asterisk softswitch can be

decomposed into multiple distributed servers by the Inter-Asterisk Exchange (IAX) protocol in order to decongest a single server installation.

As indicated earlier, the current pilot project does not implement the SER which is a more advanced SIP proxy in terms of functionality. Later SER will have to be incorporated in order to have flexibility in the features that SIP provides and improve in speed of registrations. With a fast growing number of users, SER has features that can easily support network monitoring and address operational problems such as network attacks on the system. By way of implementation of SER with all its modules, the system will support digest authentication, Call Processing Language (CPL) scripts, instant messaging, MySQL support, a presence agent, Radius authentication, record routing, an SMS gateway, a SIP-SIMPLE integration, a transaction module, a registrar, and user location with forking.

The SER and Asterisk complement each other as peer SIP proxies and will forward calls to Asterisk according to its dial plan. The SER's dial plan will also be capable of forwarding calls to users on different networks (for example, H.323, IAX, TDM) as well as service calls (like voicemail, call conferencing, directory services) to Asterisk.

The current pilot project does not provide for the H.323 environment. Since Asterisk has supports for H.323 protocol and acts as its gateway, an integration of an OpenGK server to the pilot project will be necessary as there will be need to integrate the H.323 environment at the university. This

will facilitate management, authentication, authorization and alias address mapping in the H.323 network.

5.2 Pre-paid Billing

Asterisk does not have in-built billing, however, a pre-paid billing system written by Rhodes University computer science department has been incorporated in the current system and reconfigured to meet local settings. This can be populated with current billing rates from the Telecom Namibia. Otherwise, different billing systems as preferred can also be written by university students as research projects using the Asterisk Gateway Interface (AGI) or the C API.

5.3 Customisation and Enhancement

5.3.1 Dial Plans

Since dial plans for Asterisk are defined in text files, it makes them accessible to be edited easily using standard text editors or via customised front-ends.

A dial map was created based on the groups of communication at the university. These groups of communication were identified as internal, other campuses and centres.

Selecting an outside local and international telephone line not in the university structure was done but, not incorporated in this plan as it is

currently illegal in Namibia to interface VoIP systems with the public. However, this can be incorporated when legalized. This applies to other special numbers such as for emergency services.

Figure 14 shows the dial map implemented in the current project. It shows that there are several dial plan rules that are used each time a number is dialled. The first thing in our dial map is to determine if the first digit is 0, 2, or 8. This first rule allows the system to determine if the caller desires to reach an attendant (0), or to call an internal number (2+), to call other campuses (8). This rule changes how the next digit is processed. If the first digit is a 2, it is an internal call (4 digits for this system) and the system will wait for 3 more digits before attempting to connect to call another unit in the system such as voice mail. If the digit is 8, the system will have to capture multiple digits and analyse the call to other university campuses based on the campus gateway settings.

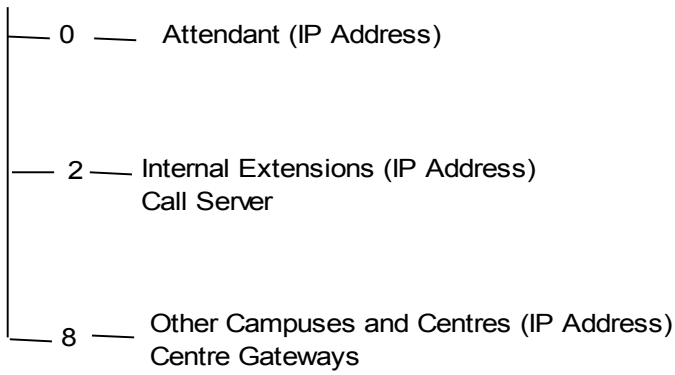


Figure 14: Dial Map Operation

The dial plans offer an intuitive means of defining dial plan logic. User information is stored in a MySQL database and can be managed directly from the Asterisk box or remotely via a simple CGI-based web interface.

The Asterisk manager interface can also be used to customise the deployed Asterisk and to build innovative user interfaces such as like the one that has been developed at Rhodes University. This can be done by use of commonly available open source tools and programming languages. Currently, as found out in this study from the department of computer sciences curriculum, the university teaches students various programming languages that can be used in this project to enhance the system locally.

5.4 IP phones

The current system has a number of IP hard phones and soft phones installed that switchs from the Asterisk server. These are internal phones with numbers that can range from 2000 up to 7000 depending on the university's internal IP addressing scheme. Call set up on the phones was performed using SIP.

5.5 Tests on the Pilot Project

This pilot project has currently been tested with users able to use the system to call from the local area network using hard phones and soft phones from the personal computers connected to the university network.

The current configured users each have an account that allows them to access the Asterisk system from SIP networks. Current users also have voicemail boxes that they can be customised to suit their needs in future.

The main goal of deploying Asterisk PBX in order to provide voice and data services to staff and students was achieved.

Chapter 6 Conclusions and Recommendations

As set out in the objectives of this study in Chapter 1, the main thrust of this work was to show factors and conditions required for a production grade VoIP system based on OSS system to be deployed in order to provide integrated telephony and data services to staff and students of the University of Namibia. This chapter concludes the study and makes some recommendations on this subject.

To determine factors and conditions required for deployment of the production grade VoIP system with open source component, the existing equipment and current status of the internet connection, data and telephone usage, current VoIP technology know-how and OSS awareness at the university were examined and an analysis was done based on the ideal requirement for the university.

6.1 Network Infrastructure

In the study, it was found out that all the buildings at the university and other campuses are connected to the data network. The current data network infrastructure is conducive to cater for all points within the university. Due to the fact that the number of internal calls are almost the same as the number of external calls, no specific area of preference in terms of deployment can be chosen although preference would be to start with internal deployment (main campus) and then to the northern campus. The call volume at the main campus is far more than any other site.

The majority of students lack access to computers. Implementing VoIP systems will have to call for an investment in hardware such as computers and VoIP handsets at some points within the university.

6.2 Data and telephone usage

The condition of VoIP readiness in the context of this study depends on factors such as telephone usage, reliability, and call quality requirements.

Further, in order to address this study's objective, a plan for deployment has been recommended.

6.2.1 Telephone usage

Currently, 654 staff members of the university are using 22 erlangs at peak periods. Assuming that students get access to telephones through provision of 1000 telephones and that students usage is half that of staff, then we expect to have the following:

Total number of erlangs required = (22/654)*500 +22 = 39 erlangs

6.2.2 Reliability

The current 88% data network utilization is high. Therefore, inclusion of voice traffic in the same network will not offer the needed reliability especially of voice services as compared to PSTN which offers a 99.999% of time.

Focus should be made on reliability of network equipment and its components and reliability of the VoIP components such as VoIP servers, gateways, IP PBXs etc.

6.2.3 Call quality and bandwidth

Call quality measures on the system should be made to be higher than Mean Opinion Score (MOS) of 3.6 as a standard. Depending on the availability of bandwidth, a codec can be selected to offer a loftier MOS. Table 10 below illustrates a guide in selecting these specifications including the required bandwidth. The bandwidth requirements can further be reduced in cases where the router supports RTP header compression.

Codec	Data Rate (kbps)	Typical Datagram Size (ms)	Packetization Delay (ms)	Codec Bandwidth Requirement (kbps)	Theorectical Maximum MOS [Range 1 - 5]	University Bandwidth Requirement for Voice [Mbps]	88% of Current Bandwidth of 1.536Mbps	Total University Bandwidth Requirement [Mbps]
G.711u	64	20	1	87.2	4.41	3.3	1.3	4.6
G.711a	64	20	1	87.2	4.41	3.3	1.3	4.6
G.726-32	32	20	1	55.2	4.22	2.1	1.3	3.4
G.729	8	20	25	31.2	1.07	1.2	1.3	2.5
G.723.1 MPMLQ	6.3	30	67.5	21.9	3.87	0.8	1.3	2.2
G.723.1 ACELP	5.3	30	67.5	20.8	3.69	0.8	1.3	2.1

Table 10: Specification on Required Bandwidth

The formula for calculation of university bandwidth requirement for voice (BWV) is as follows:

BWV = number of erlangs x CODEC Bandwidth requirement

6.2.4 Plan for deployment

Implementation can be done in two stages as follows.

<u>Stage 1</u>

The study findings indicated that although most computer science students (who are targeted for implementation and take up through projects) are aware of open source products and are capable of programming in at least one of the programming languages, they had limited knowledge of VoIP technology. However, the author believes that the students lack these technological concepts due to non existence of a course in RTMM systems.

Introduction of this course in the curriculum coupled with already programming skills of the students will give sustainability to the project. Hence, undergraduate and postgraduate students will take advantage of the project environment especially that it is open source, to come up with research projects that will enhance and expand the project to cater for the whole university including its outside campuses. In doing so, open source products will be promoted and contribution to the open source community will be possible from the university staff and students.

The project can start with the following equipment and software which would cost approximately N$80,000.00.

- At least one ISDN port for connection to the PSTN
- 100BaseT Ethernet environment using BayStack 450 switches
- At least two PCs: Asus A7V, 900MHz Athalon, 256 MB Ram, 60GB hard disk .
- At least fifty hard phones
- Digium T100P
- Ubuntu Linux operating system and the iLanga installation kit.
- APC Smart-Ups 700

Bandwidth requirements for stage 1 can be calculated based on number of erlangs and codec required as indicated in Table 8.

<u>Stage 2</u>

This will involve an extension to other faculties, administration and campuses. Stage 2 greatly depends on the success of stage 1.

6.3 VoIP Technology and OSS Awareness

As derived from the observations, the introduction of a course in RTMM systems would provide students with VoIP technology knowledge. The majority of staff and students (66%) are familiar with OSS system and 99% are well versed system development that includes programming using various programming languages. As a result, it should not be difficult in incorporating them in the development and maintaining of the project.

6.4 Sustainability

The project is capable of sustaining itself by following a model based on funding, technical, operational and staffing as articulated in section 4.4 of this document.

6.5 Deployment of VoIP Prototype System

The pilot project installed in the Computer Centre involved use of OSS Asterisk. This system is able to provide voice services to staff and students with limited functionalities. However, with necessary purchase of hardware, the project can be extended beyond its current operation.

As illustrated in chapter 5, the prototype VoIP system was configured using OSS Asterisk PBX and provided integrated data and telephony services to staff and students.

6.6 Summary of conclusions

In chapter 1 of this document, a working hypothesis was set as "*It is possible, using current available VoIP Free OSS technology, to sustainably deploy integrated telephony and data services to staff and students at an academic institution*". This hypothesis was to be articulated and tested in the case of The University of Namibia.

The researcher accepts this hypothesis as true based on recommendations articulated in section 6.1, 6.2, 6.3, 6.4, and 6.5 of this document.

This study shows a significant opportunity for computer science lecturers and students at the university to pre-package or bundle open source VoIP solutions. Though there are numerous open source tools available for Asterisk, none were integrated into this trial environment due to constraints as mentioned in the implementation section.

Implementation of this project will not only serve the purpose of cost saving but will most importantly serve to increment knowledge of computer science staff and students of the university due to the fact that this project will attract new ideas in the area of RTMM technology. Open source products also come with a package cheaper for support and maintenance.

The university's infrastructure is however large and such a project with new knowledge can not be implemented in one stage but phased out starting from the Computer Department and depending on the successes deployed to other sections of the university.

Lastly, the project may remain for a while as an academic research project and not commercialised because VoIP is currently not commercially allowed in Namibia.

6.7 Suggestions for Future Work

The following are the suggestions by the author for future work based on this research.

It was observed that the current university data network performance has problems in the provision of client services. A study based on the QoS need to be done in the university network so as to determine congestion problems with the current logical and physical network topology. This study should aim to evaluate capability of existing network devices such as routers and switches and the effectiveness of QoS services for the applications used by the university community.

The prototype VoIP system introduced in the university LAN will need to be extended as recommended in chapter 5. This could possibly be done as research projects for final year students.

References

[3Com 2006] 3Com (2006). *3Com® VCX™ IP Telephony Module.* <http://www.3com.com/other/pdfs/products/en_US/400865.pdf>

[Apache 2005] Apache Software Foundation. *Contributing to The Apache Software Foundation*. 2005. <http://www.apache.org/foundation/contributing.html>

[Arango et al. 1999] Arango, M., A. Dugan, I. Elliott, C. Huitema and S. Pickett (1999). *RFC 2705: Media Gateway Control Protocol (MGCP),* Oct. 1999.

[Asterisk 2006] Asterisk (2006). *The asterisk pbx,* 2006, <http://www.asterisk.org>

[Bickman and Rog 1998] Bickman, L. and Rog, D.J. (1998). *Handbook of applied social research methods*. Thousand Oaks: Sage Publications.

[Black 2000] Black, U (2000). *IP routing Protocols*. Upper Saddle River, New Jersey 07458: Prentice-Hall, Inc., ISBN 0–13–014248–4

[Bland 1999] Bland, M. (1999). *An introduction to medical statistics*. Second edition. Oxford University Press. Great Britain.

[Bless and Hugson-smith 2000] Bless, C. and Higson-Smith, C. (2000). *Fundamentals of social research methods: an African perspective*. 3rd ed. Kenwyn: Juta.

[Boucher 1992] Boucher, J.R. (1992). *Traffic System Design Handbook: Timesaving Telecommunication Traffic Tables and Programs.* IEEE Telecommunications Handbook Series. Wiley-IEEE Press 1992. ISBN-10: 0780304284

[Bridges 2005] Bridges.org (2005). *Comparison Study of Free/Open Source and Proprietary Software in an African Context*. <http://www.bridges.org>

[Bridgman 1955] Bridgman P.W. (1955). *On Scientific Method*. Reflections of a Physicist, 1955.

[Bruggink 2003] Bruggink, M. (2003*). Open Source in Africa: Towards Informed Decision-Making*. IICD Research Brief – No 7, August 2003.

[Brundtland 1987] Brundtland Report. (2005). <http://en.wikipedia.org/wiki/Brundtland_Report>

[Camp 2003] Champ, K. (2003). *IP telephony Demystified*. New York: MacGraw-Hill. ISBN 0071406700

[Carhee 2004] Carhee, S. (2004). *Road to IP Telephony, The: How Cisco Systems Migrated from PBX to IP Telephony*. Cisco Press. ISBN-10: 1-58720-088-0; ISBN-13: 978-1-58720-088-5;

[Charter 2003a] Charter, Media Gateway Control (2003a). The media gateway control charter working group at the ietf, 2003, <http://www.ietf.org/html.charters/megaco-charter.html>.

[Charter 2003b] Charter, SIP (2003b). *The sip-charter working group at the ietf*. 2003,<http://www.ietf.org/html.charters/sip-charter.html>.

[Cisco 2007] Cisco Documentation (2007). *Traffic Analysis for Voice over IP*. Version 3, 2007. <http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/ta_isd.htm#xtocid0>

[Cooper and Schindler 2003] Cooper, D.R. and Schindler, P.S. (2003). *Business research methods*. 8th ed. McGraw Hill Company:New Yolk.

[Creswell 2002] Creswell, J.N. (2002) *Research Design: Qualitative, Quantitative and Mixed method Approaches.* SAGE Publication Ltd, 2Rev Ed edition (2002), ISBN-10:0761924418

[D'Antoni 2004] D'Antoni, H. (2004). *Open Source Software Use Joins The Mix*. InformationWeek Publication, Nov. 1, 2004, <http://www.informationweek.com>

[David 2006] David, P.A. (2006). *A Multi-dimensional View of the "Sustainability" of Free & Open Source Software Development-Sustaining Commitment, Innovation and Maintainability with Growth*. Oxford Internet Institute & Stanford University. England: OSS Watch Conference: 7 June 2006

[Davidson et al. 2000] Davidson J., Peters, J. and Gracely, B (2000). *Voice over IP Fundamentals*. Cisco Press. ISBN-10: 1-57870168-6

[Downes 2006] Downes, S. (2006). Models for Sustainable Open Educational Resources. National Research Council Canada, available at: <www.oecd.org/document/32/0,2340,en_2649_33723_36224352_1_1_1_1,00.html>

[Erlang 2007] Erlang Caculator <http://www.erlang.com/calculator> Accessed July, 2007.

[Farmer 2006] Farmer, J. (2006). *The Commercialization of Open Source*. Oxford, England: OSS Watch, University of Oxford, Open Source and Sustainability Conference, 12 April 2006.

[Foote 2005] Foote, T. (2005) *Wikipedia*. Utah: 2005 Open Education Conference. <http://cosl.usu.edu/media/presentations/opened2005/OpenEd2005-Foote.ppt>

[Force 1996] Force, Internet Engineering Task (1996). Rtp: A transport protocol for real-time applications, Jan. 1996, <http://www.ietf.org/rfc/rfc1899.txt>.

[Greene et al. 1999] Greene, N., A. Rayhan, C. Huitema, B. Rosen and J. Segers (1999). *RFC 3015: Megaco Protocol* Version 1.0, Oct. 1999.

[Hamblen et al. 2005] Hamblen, J.O., Hall, S.T. and Furman, M.D. (2005). *Rapid Prototyping of Digital Systems*. Springer-Verlag, New York Inc. 2nd Edition 2005, ISBN-10:0387277285

[Handley et al. 1999] Handley, M.,Schulzrinne, H., Schooler, E. and Rosenberg, J. (1999). *IETF RFC 2327: SIP: Session initiation protocol*, Mar. 1999.

[Hardy 2003] Hardy, W.C. (2003). *VoIP Service Quality : Measuring and Evaluating Packet-Switched Voice*. McGraw-Hill Professional; 1 edition. ISBN 0071410767

[Hoffer et el 2005] Hoffer, A. J., Goerge, F. J. and Valacich, S. J. (2005). *Modern Systems Analysis and Design*, Pearson Education Inc. USA, Fourth Edition. ISBM 0-13-127391-4

[IEC Tut. 2005] *IEC Web ProForum Tutorials*. (2005). <http://www.iec.org>

[Ins-stat 2007] Ins-stat (2007). *Europe Leads the Booming Consumer VoIP Market.* <http://www.instat.com>

[ISO/IEC 1994] ISO/IEC 7498-1 (1994) *Information technology - Open Systems Interconnection - Basic Reference Model: The Basic Model*: JTC 1; ISO Standards, ICS: 35.100.01

[ITU 2004] ITU-T Recommendation G.108, *Application of the E-Model: A Planning Guide*. <http://www.itu.int/publications/main_publ/itut.html>

[ITU-D 2004] The International Telecommunication Union (UIT).(2004). *Telecommunication Regulatory Survey*. <http://www.itu.int/ITU-D/treg/Events/Survey/survey.asp>

[ITU-T 1996] ITU-T Recommendation P.800, *Methods for Subjective Determination of Transmission Quality*. <http://www.itu.int/publications/main_publ/itut.html>

[ITU-T 1998] ITU-T (1998*). Itu-t recommendation h.323: Packet-based multimedia communications systems*. Tech. rep., Telecommunications Standardization Sector, 1998.

[ITU-T 2000] ITU-T (2000). Itu-t recommendation h.248: Gateway control protocol. Tech rep., Telecommunications Standardization Sector, 2000.

[ITU-T 2003] ITU-T Recommendation G.107, *The E-Model, a Computational Model for Use in Transmission Planning.* <http://www.itu.int/publications/main_publ/itut.html>

[ITU-T 2007] ITU-T (2007*). Itu-t recommendation: Transmission systems and media, digital systems and networks. Tech. rep.,* Telecommunications Standardization Sector, 2007

[Katzenellenbogen et al. 1999] Katzenellenbogen, J.M., Joubert, G. and Abdool K.S.S. (1999). Epidemiology: a manual for South Africa. Second impression. Cape Town: Oxford University Press.

[Keagy 2000] Keagy, S. (2000). *Integrating Voice and Data networks*. Cisco Press. ISBN-10: 1-57870-196-1

[Lambert 2001] Lambert, H (2001). *Getting Ready for Voice over Data*, Cisco Systems, VoiceCon 2001, Washington, D.C., February 26, 2001.

[Lauer and Asher 1988] Lauer, J.M and Asher, J.W. (1988). *Composition Research: Empirical Designs*. Oxford: Oxford University Press, Inc. ISBN-10: 0195041720

[Liesenborgs 2000] Liesenborgs, J. (2000). *Voice over IP in networked virtual environment*. Masters Degree Thesis. University of Maastricht.

[Lipman 2005] Lipman, A.D (2005). *Global Trends in VoIP Regulations*. Berlin: Swidler. <http://www.ptc.org/events/ptc06/program/public/proceedings/Andrew_Lipman_chair_s_intro1_remarks_t14.pdf>

[Mahler 2005] Mahler, P. (2005). *VoIP Telephony with Asterisk – A Systematic Approach to Learning the Open Source PBX*, Signate, Second Edition, ISBN: 0-9759992-2-2

[Malone and Murphy 2005] Malone, D. and Murphy, N. (2005). IPv6 Network Administration. USA: O'Reilly Media, Inc ISBN: 0-596-00934-8

[Manesh 2004] Manesh, K.N. (2004). Asterisk: A Non-Technical Overview. Newport Beach, CA 92660: Millenigence, Inc. 2004

[Mason et al. 1999] Mason, R.D., Lind, D.A.and Marchad, W.G. (1999). *Statistical techniques in business and economics.* 10th edition. McGraw Hill Company: New Yolk.

[ME 2005] Ministry of Basic Education, Sport and Culture, Ministry of Higher Education, Training and Employment Creation. *ICT Policy For Education*. Polytechnic Press 2005 Namibia. ISBN 0-86976-666-X

[Meraka 2005] Meraka (2005). *Meraka Institute*. <http://floss.meraka.org.za/postnukeII/>

[Mertens 1998] Mertens, D.M. (1998). *Research methods in education and psychology: Integrating diversity with quantitative and qualitative approaches*. Thousand Oaks:

[Mueller 2002] Mueller, S. (2002). *APIs and Protocols For Convergent Network Services.* New York: MacGraw-Hill. ISBN 007138880-X

[Ohrtman 2003] Ohrtman, F. (2003). *Softswitch : Architecture for VoIP (Professional Telecom).* New York: MacGraw-Hill. ISBN 0071409777

[Penton and Terzoli 2003] Penton, J. and Terzoli, A. (2003). *Asterisk: A Converged TDM and Packet-based Communications System*. South African Telecommunications Networks and Applications Conference, September 2003, Fancourt

[Penton and Terzoli 2004] Penton, J. and Terzoli, A. (2004). *iLanga: A Next Generation VoIP-based, TDM enabled PBX*. South African Telecommunications Networks and Applications Conference, September 2004, Spiers

[Rosenberg and Shockey 2000] Rosenberg, J. and Shockey, R. (2000). *The session initiation protocol: A key component for internet telephony*. 2000, <http://www.cconvergence.com/shared/article/showArticle.jhtml?articleID=8700868>

[Rybaczk 2004] Rybaczk, P. (2004*). Cisco Network Design Solutions For Small-medium Businesses*. Cisco Press. ISBN 1587051435

[Salah 2006] Salah, K. (2006). *On the deployment of VoIP in Ethernet networks: methodology and case study.* Computer Communications, Volume 29, Issue 8, 15 May 2006

[Scavo 2005] Scavo, F (2005*). Key Advantage of Open Source is Not Cost Savings* . Research Byte. Computer Economics. <http://www.computereconomics.com/article.cfm?id=1043>

[Schoolnet 2007] Schoolnet Namibia (2007). <www.schoolnet.na>

[SHSU 2006] Sam Houston State University (2006) *University Dumps Cisco VoIP for Open-Source Asterisk*. Texas, USA: September 2006 <http://www.networkworld.com/news/2006/091206-von-sam-houston.html>

[Spencer et al. 2003] Spencer, Mark, M. Allison and C. Rhodes (2003). *The asterisk handbook*, Mar. 2003.

[Sterne and Herring 2005] Sterne, P. and Herring, N. (2005). *LinuxWorld: The Conversion Model*. 2005. Linux World, Dec. 30, 2005. <http://de.sys-con.com/read/158863.htm>

[Stevens 1993] Stevens, T.W. (1993). *TCP/IP Illustrated, Volume 1: Protocols*. Addison Wesley Longman, Inc. ISBN 0-201-63346-9 (v.1)

[Sulkin 2002] Sulkin, A. (2002). *PBX Systems for IP Telephony.* New York: MacGraw-Hill. ISBN 0071375686

[Sziget 2004] Sziget, T. (2004). *End-to-End QoS Network Design*. Cisco Press 2004. ISBN-10: 1-58705-176-1

[Tanner 2006] Tanner, S. (2006). *Handbook on Cost Reduction in Digitisation.* September 2006, <http://www.minervaeurope.org/publications/costreduction.htm>

[Taylor 2006].Taylor, M. (2006). *OSS Watch conference,* Open Source and Sustainability*, held at the Said Business School*, Oxford, 10-12th April 2006

[TeleGeography Research 2005] TeleGeography Research (2005). International Voice Market Improving. 2005 <http://www.telegeography.com/press/releases/2005-11-10.php>

[Trochim 2001] Trochim, W.M.K. (2001). *Research methods knowledge base*. [Online]. Available: <http://trochim.cornell.edu/kb/guideelements.htm >

[Unuth (2007)] Unuth, N., (2007). *Your Guide to Voice over Internet Protocol*. <http://voip.about.com/od/requirements/a/bandwidth.htm >

[Vagesna 2001] Vagesna, S. (2001). *IP Quality of Service*. Cisco Press. ISBN-10: 1-57870-116-3

[van Reijswoud and Topi 2005] van Reijswoud, V. and Topi, C. (2005). *Alternative Routes in the Digital World - Open Source Software in Africa*. <http://opensource.mit.edu/papers/reijswoudtopi.pdf>

[Walker and Hicks 2004] Walker, J.Q. and Hicks, J. T. (2004). *Taking Charge of your VoIP Project*. Cisco Press. ISBN 1587200929.

[Weerawarana and Weeratunga 2004] Weerawarana, S. and Weeratunga, J. (2004). *Open Source in Developing Countries.* SIDA.

[Wheeler 2003] Wheeler, D.A. (2003). *Why Open Source Software / Free Software (OSS/FS)? Look at the Numbers!,* May 2003. <http://www.dwheeler.com/oss_fs_why.html>

[Wiegand 2005] Wiegand, H. (2005). *VoIP: To regulate or not to regulate?*. Siemens Venture Capital. Investment Associate. <http://www.itu.int/ITU-D/treg/index.html>

[Wladawsky-Berger 2001] Wladawsky-Berger. (2001). IBM, <http://ibm.com/news/us/2001/08/15.html>

Appendix i

Faculties and Centres Academic Staff for 2004

Faculty/Centre	Number of Staff
Agriculture & Natural Resources	35
Economics & Managemnet Science	34
Education	39
Humanities & Social Science	71
Law	9
Medical & Health Sciences	37
Sciences	63
Centre for Ecternal Studies	14
Language Centre	15
MRCC	7
Library	14
Northern Campus	5
Total	**343**

Faculties and Centres Support and Administartive Staff for 2004

Faculty/Centre	Number of Staff
Agriculture & Natural Resources	11
Economics & Managemnet Science	5
Education	6
Humanities & Social Science	8
Law	6
Medical & Health Sciences	6
Sciences	17
Centre for Ecternal Studies	28
Language Centre	3
MRCC	3
Total	**93**

Central Administration Staff for 2004

Faculty/Centre	Number of Staff
Office of the Vice Chancellor	5
Office of the PVC (AA & R)	6
Office of the PVC (A & F)	5
Office of the Registrar	19
Computer Center	12
Dean of Students	45
Estate Services	44
Finance	20
Human Resources	13
Library	30
Northern Campus	10
UNAM Foundation	3
Henties Bay Marine and Coastaln Resources Research Centre	6
Total	**218**
Gradnd Total	**654**

Appendix ii

Appendix iii

PART A: PERSONAL INFORMATION

This information is strictly confidential and will only be used for this research
(Please tick or fill the correct answer where necessary.)

1. Gender

Male	Female

2. Age Group

<20	20-30	30-40	40-50	>50

3. Study Year

1st Year	2nd Year	3rd Year	4th Year	Postgrad

Lecturer	Other(Please Specify)

PART B: LEVEL OF AWARENESS ON OPEN SOURCE

In this section, we would like to know if you have any prior experience with open source software. *(Please tick or fill the correct answer where necessary.)*

1. Have You Heard About Open Source Software

Yes	No

2. If Yes, State Type of Open Source (System/Application/Both)

System	Application	Both

3. Are You Involved in System Development?

Yes	No

4. If Yes, Does Your System Development Involve Open Source?

Yes	No

5. Indicate Programming Languages That You Know

PART C: DEMOGRAPHIC FACTORS

In this section, we would like to know your views on the following questions.
(Please tick or fill correct answer where necessary)

1. Do you have access to a phone?

Yes	No

2. If yes state the type of phone

Land Phone	Cell Phone

3. What do you often use the telephone for? Use the scale below; please circle the closest match for each of the following.

		Often	Medium	Rarely
A	Academic enquiries (e.g. getting results)	3	2	1
B	Consultation with lecturers	3	2	1
C	Personal/private discussions	3	2	1

4. State how much you spend on phone per month (Approximately) N$

5. Do you have direct access to computer system?

Yes	No

6. How often do you use a computer system?

On a daily basis	
On weekly basis	
On a monthly basis	
Can't tell (i.e. infrequent)	

7. Where do you often access Internet?

Home	
University	
Internet Café	
Other *(Please specify)*	

8. What type of Internet connection do you use (i.e. at the place most frequent access to computer)?

Dial up connection	
ISDN/streamyx connection	
Other *(Please specify)*	
Not sure	

9. How do you communicate with UNAM (e.g. Check for marks / make appointments etc) from outside campus?

Email	
Telephone	
Fax	
Other *(Please specify)*	