# VoIP System Using Open Source Software Component in Tertiary Institutions: The Case of the University of Namibia

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## Abstract

Governments and their agencies are often challenged by high cost and flexible telephonic, Web based data services. Emerging technologies, such as those of Voice over Internet Protocol (VoIP) that allow convergent systems where voice and Web technologies can utilise the same network to provide both services, can be used to improve such services. The contribution of this paper is the outcome of a study from examining 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 PABX system for voice and fax communication services, as well as the local area network connected to Internet for Web and data services. Interesting findings include that the University of Namibia has a potential to implement the project. Since the software recommended for installation is open source, the project could be used as a source of valuable information by students who specialize in real-time multi-media systems in Southern African tertiary institutions at large.

## Introduction

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 As an example, outdated Private Branch eXchange (PBX) and the Public Switched Telephone Network (PSTN) systems require costly maintenance agreements and are difficult to integrate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

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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.

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).

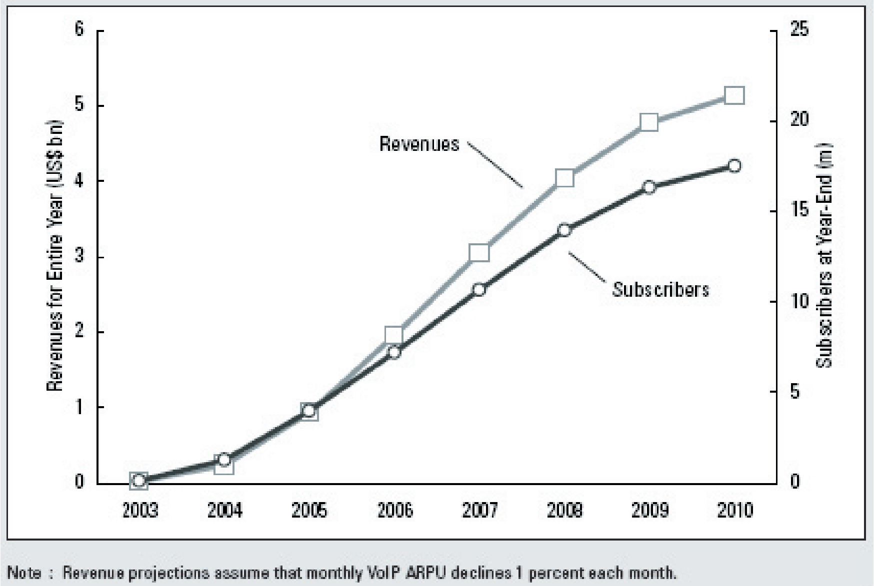



**Figure 1 :** Projected VoIP subscribers and revenue for the period 2003-2010.
Source: TeleGeography Research 2005

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.

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. Regulations in South Africa and America allow providers to use VoIP.

## Definition 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.

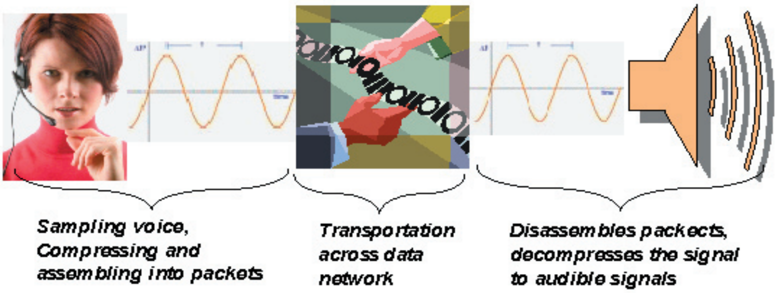



**Figure 2 :** Voice transmission from one end to another

## 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:

- o Codecs
- o Transmission Control Protocol/Internet Protocol (TCP/IP) or VoIP protocols
- o IP telephony servers or PBXs
- o VoIP gateways or routers
- o IP phones and softphones.

## 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).

**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). The suite consists of two important protocols namely the Transmission Control Protocol and the Internet Protocol.

**VoIP Protocols**

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.

**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.

### 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 Session Initiation Protocol (SIP) protocol to Integrated Services Digital Network (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.

### IP Phones and Soft phones

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.

### Planning for VoIP Implementing

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.

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.

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

**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.

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 Telecom 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.

## Data and Telephone usage

Investigations into the university revealed that currently there are about 654 staff members at the main campus and other campuses. 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 Telecom. A PABX software is installed that redistribute telephone bills as per caller identification.

## 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.

Daily analysis of a monthly data collected from the sample data indicated the Busy Hour Traffic (BHT) (Above 10% traffic volume) between 08:00hrs and 12:00hrs and between 14:00hrs and 15:00hrs. Table 1 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 1: Analysis of Call based on 24 hourly band**

Based on the BHT found from Table 1, daily observations were done on 32 days as summarised in Table 2. An 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 2: Summary of Calls Based on BHT**

## Call Flow Analysis

Forty seven percent (47%) of calls made were found to be internal within the university as shown in Table 3. 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% |

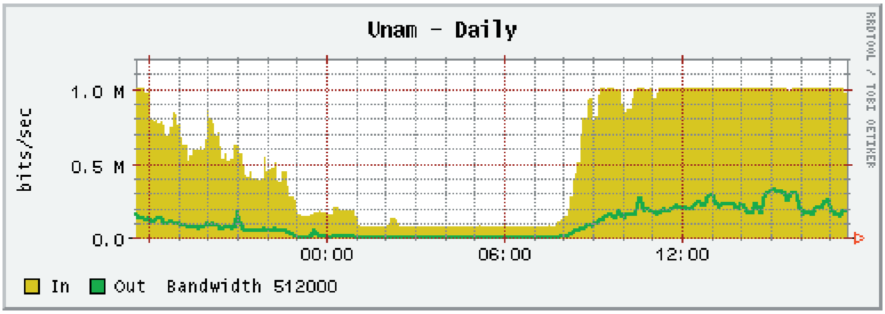
| | | | |
|---|---|---|---|
| March 2006 | 299859 | 146970 | 49% |
| April 2006 | 228971 | 106507 | 47% |
| May 2006 | 266396 | 134025 | 50% |
| June 2006 | 243890 | 125829 | 52% |
| TOTAL | 2784971 | 1306182 | 47% |

**Table 3: Call Flow Analysis**

## Data network and usage

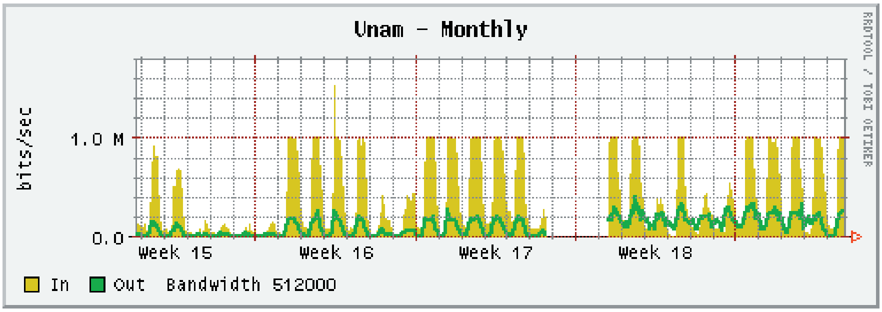
Currently, the university is provided with bandwidth of 1.536Mbps from the service provider.

The overall university daily, monthly and annual data network usage is depicted in Figure 3, Figure 4 and Figure 5 respectively.




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

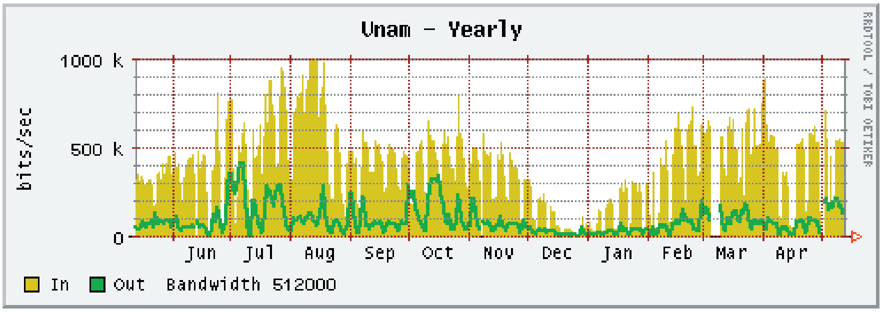
**Figure 3: Unam Daily Data Network Statistics**
**Source: Unam Computer Center**




| Legend | Min | Max | Avg | Last |
|---|---|---|---|---|

| | | | | |
|---|---|---|---|---|
| **In** | 4503 | 1539017 | 386874 | 1008770 |
| **Out** | 1371 | 392218 | 96895 | 259857 |

**Figure 4:** Unam Monthly Data Network Statistics
*Source: Unam Computer Center*




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

**Figure 5:** Unam Annual Data Network Statistics
*Source: Unam Computer Center*

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.

## 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.

## 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 6 illustrates the current pilot system.

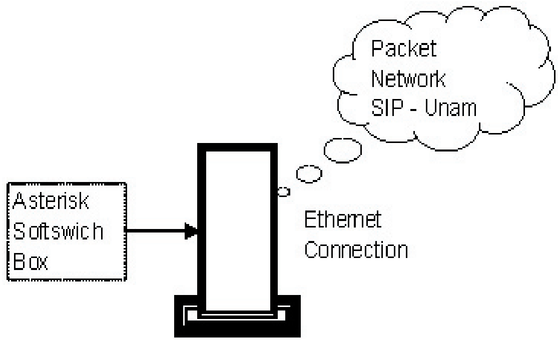



**Figure 6:** Asterisk main components of the pilot project.

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.

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.

## Customisation and Enhancement

*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 7 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.

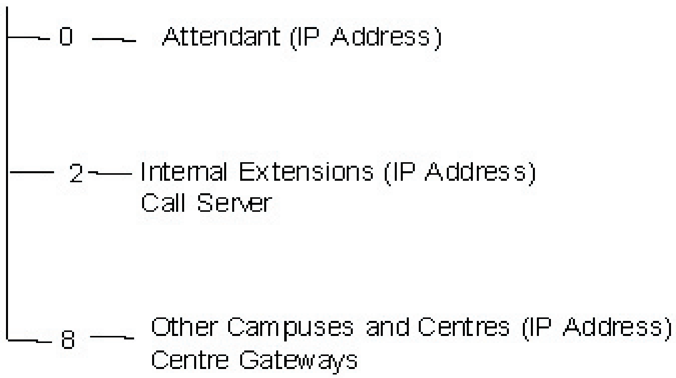



**Figure 7:** Dial Map Operation

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.

## 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.

## 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.

## Conclusions and Recommendations

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.

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.

## 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

## Reliability

The current 88% data network is 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.

## 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 4 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 4: 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***

## Plan for deployment

Implementation can be done in two stages as follows.
<u>Stage 1</u>

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

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

Bandwidth requirements for stage 1 can be calculated based on number of erlangs and codec required as indicated in Table 4.

<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.

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 section 7 of this paper.

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 Real Time Multimedia Communication over packet networks 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.

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 the implementation section of this paper. This could possibly be done as research projects for final year students.

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.

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