Re-establishing and improving the experimental VoIP link with the University of Namibia: A Case Study

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Abstract

The sharing of ideas and information comes standard in any area of research. The need for voice communication to do this is relatively high. African universities are no exception to this but the main drawback is that voice communication comes at a price. With the help of the Internet and open source products we may no longer see voice communication costs as an obstacle. This paper discusses what needs to be considered and known for the successful deployment of VoIP between different sites with differing network infrastructure and Internet connectivity. This project aims to be a blueprint for connecting African universities together via voice and video over IP on a data network.

Introduction

The use of legacy *Public Switched Telephone Networks* (PSTNs) as a means of communication is a worldwide phenomenon. For areas that may not have PSTN coverage mobile cellular networks seem to fill the gap. These two communication means may be adequate but are costly. The introduction of *Voice over Internet Protocol* (VoIP) in the mid-1990s added a new dimension to the communications market. The use of VoIP has grown from a small scale market penetration to becoming a major player in the communications market [8]. The reason for its popularity is based on its cost, value added services, and ease of deployment. When VoIP was first made available *quality of service* (QoS) was not a major deciding factor. As time progresses, users will soon demand *five-9s* (99.999%) reliability for their VoIP communication. In September 2004 Skype experienced a short downtime but there was not much of a loss in reputation as these were the early days of Skype.

In August 2007 Skype experienced a communication downtime of about two days and this lead to a loss in confidence among affected Skype users [1].
This paper looks at the deployment of *iLanga*, which is a computer based *Private Branch Exchange* (PBX) developed at Rhodes University [2]. This paper also discusses how best to deploy VoIP over different network infrastructures and maintain quality of service.

A test bed enables us to observe and analyze the behavior of applications in a lab environment that accurately emulate conditions on the current and/or planned production network [4]. Using the test bed we will assess whether there is a need for traffic shaping to sustain a reliable voice session over a congested network.

Background

VoIP and iLanga

VoIP essentially is the process of transmitting packaged voice samples over an IP network. VoIP commonly generates two types of traffic namely signalling and real time streaming. Signaling performs the task of establishing, managing and terminating VoIP sessions. Once a VoIP session is established a real time streaming protocol is used to transmit voice data. A common signaling protocol used in VoIP is *Session Initiation Protocol* (SIP) and a common real time streaming protocol used is *Real-time Transport Protocol* (RTP) [5]. The voice samples that are streamed are based on the voice encoding used. Voice encoding is the digitization of an analog audio stream. Codecs and transcoding are used to perform this task. A codec is used for audio data compression/decompression. Transcoding is used for direct digital-to-digital conversion, from one codec to another [6]. When choosing a codec a balance must be obtained between bandwidth utilization and quality of service.

iLanga is an open source computer-based PBX. iLanga is currently installed on the Rhodes University IP-network and has connections to the PSTN. iLanga has various components namely Asterisk, *SIP Express Router* (SER) and iLanga proxy [2]. Asterisk is an open source switching system that enables the communication between devices of similar or different telephony technologies. SER is an open source high-performance SIP proxy for user agents. The iLanga proxy provides the capability of multiple sessions to be made between the web front-end and *Asterisk Manager API*.

The Asterisk Manager API is responsible for interfacing between third party applications and PBX operations such as call recording/monitoring and originate calls [3]. Figure 1 illustrates the components of iLanga and how they relate to each other.

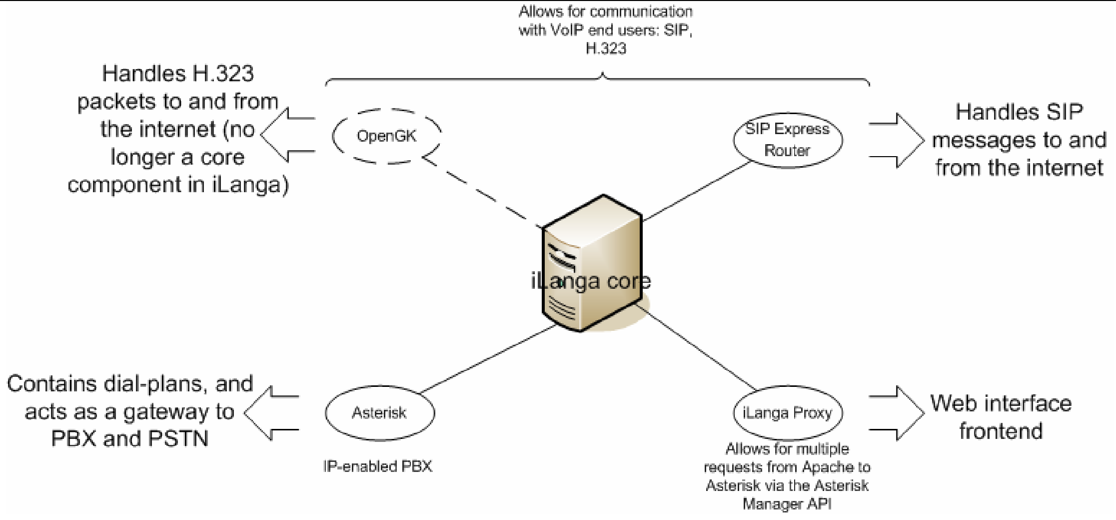

Figure 1: iLanga components

RTP

RTP sits at the application layer of the *Open Systems Interconnection* (OSI) model and encapsulates RTP packets in uses *User Datagram Protocol* (UDP) packets. The reason why UDP instead of TCP is used is because of its connectionless characteristic [6]. Within the VoIP framework if a voice packet is lost or dropped there is no need to request a re-send as all conversations occur at real-time. RTP is essential because it holds the payload (voice samples). A high rate of successful end-to-end RTP transmissions results in a high level call quality.

Codec and Transcoding

A codec is important in the VoIP set-up because it performs sampling of analog audio streams and determines the data rate [6]. Transcoding is also an important part of the VoIP set-up because not all devices use or support similar codecs. Transcoding overcomes this problem by performing conversions between different codecs. Although transcoding gives VoIP a level of interoperability it is costly in terms of delay and call quality. When selecting the appropriate codec in a VoIP set-up a balance must be obtained between maintaining intelligible communication and reducing bandwidth consumption.

Connecting Sites

The two sites that this project focuses on are *Rhodes University* (RU) and *University of Namibia* (UNAM). The two institutions have different internet bandwidth allocations and deploy a different LAN infrastructure. Table 1 is a comparative between RU and UNAM:

	Internet Link Speed	**LAN Speed**
RU	12 224 Kbps	Gigabit
UNAM	1 024 Kbps	100/1000 Mbps

Table 1: Comparison between RU and UNAM

From the table it is important to note that the number of users on the network is relatively similar.

Testing

VoIP Network Test bed

The main purpose of the test bed is to emulate the RU-UNAM link and to carry out qualitative tests. We may not achieve an exact replication of the existing link and bandwidth utilization between RU and UNAM but it does give a good base for conducting required tests. The results of these tests will influence the exact nature of VoIP deployment using iLanga. The tests will monitor and analyse how traffic flows between a congested network and a less congested network using different scenarios with varying demands on available bandwidth. Using this data we can then see how best to ensure that reliable and sustainable voice conversations can be made regardless of the level of congestion. In this case the critical packets to be monitored are the voice data packets (UDP carrying voice data). Figure 2 illustrates the test bed:

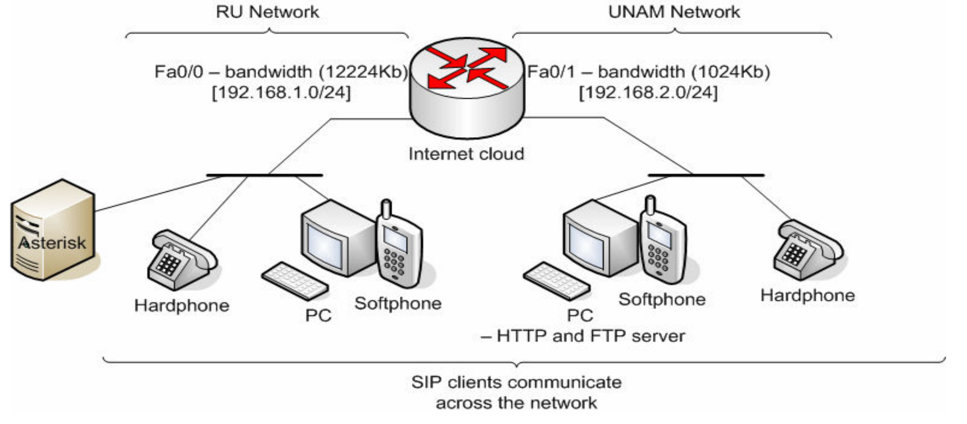

Figure 2: Network Test bed

The settings on the 2600 series Cisco router are listed in table 2:

Setting	FastEthernet 0/0	FastEthernet 0/1
Bandwidth (kbps)	12 224	1 024
Rate limit (bps)	12 224 000	1 024 000
Max burst (bytes)	1 528 000	128 000
Min bursts (bytes)	1 528 000	128 000

Table 2: Cisco router interface configurations

Analysis

The following steps were carried out to achieve a better insight into bandwidth usage on the test bed.

- Using *ManageEngine VQManager* which is a VoIP network traffic monitoring software tool, *windump* which is a Windows based packet sniffer and *iperf* which is an end-to-end bandwidth measuring tool. The network traffic was analysed to see which packets were bandwidth intensive under a default *fair queue* (FQ) traffic shaping method on a 2600 series *Cisco* router [7]
- Under a default FQ on the router and streaming video from the UNAM PC to the RU PC the network was then analysed and voice calls initiated between the SIP clients
- Under a *Class Based Weighted Fair Queue* (CBWFQ) [7] on the router and streaming video from the UNAM PC to the RU PC the network was then analysed and voice calls initiated between the SIP clients

The tests conducted were a basic measure to see how best to maintain an intelligible voice conversation without increasing available bandwidth.

Results

The results obtained from the analysis method are in table 3. The *Mean Option Score scale* (MOS) is used to grade the quality of a voice conversation [6]. MOS is an accepted yet basic method of measuring voice quality and is described in ITU P.800. The MOS values range from 1 to 5 where 1 is bad (communications breakdown) and 5 is excellent (perfect auditory reception).

Test	Result
FQ	Voice Quality Rating: 4.4 Available end- to-end Bandwidth: 1.04 Mbits/sec
FQ with video streaming	Voice Quality Rating: 3.5 Available end- to-end Bandwidth: 242 Kbits/sec
CBWFQ	Voice Quality Rating: 4.4 Available end- to-end Bandwidth: 1.05 Mbits/sec
CBWFQ with video streaming	Voice Quality Rating: 4.3 Available end- to-end Bandwidth: 263 Kbits/sec

Table 3: Summary of Analysis Results

Future Work

The next phase of the project is to finalise VoIP deployment. Using the results of the tests and knowledge gained through the literature review this process should be manageable. The process involves:

- Setting up and tweaking iLanga at UNAM
- Perform tests on the live RU-UNAM VoIP link
- Recommend border router configurations at UNAM and UNAM's ISP
- Verify a fully sustainable VoIP link

After a successful deployment it will be interesting to look into video implementation so as to create a total package.

Conclusion

It is important to know the capacity and capability of a network on which VoIP is to be deployed. The main reason is to achieve five-9s product reliability. If periodic network congestion exists the best method of avoiding poor quality voice conversations is to implement QoS. This paper gives a solid answer to the problem of deploying VoIP over low bandwidth networks with high rates of network congestion. In relation to the RU-UNAM link we are now better equipped to solve bandwidth related problems in VoIP deployment. The tests on the network test bed provided more insight into where the problem areas lie and how best to solve them. The results of this paper will help achieve our overall goal of deploying a sustainable and reliable VoIP service between different sites with differing network infrastructure and Internet connectivity.

Reference

[1] Skype. *August 2007 – Skype. Take A Deep Breath*. [on-line]. Available: http://heartbeat.skype.com/2007/08/ . Last Accessed: September 2007

[2] Penton, Jason. *iLanga: A Next Generation VoIP-based, TDM-enabled PBX*. South African Telecommunications Networks and Applications Conference. September 2004

[3] Digium. *[FAQ] Integration - AGI/FastAGI, Manager API & AJAM*. [on-line]. Available: http://forums.digium.com/viewtopic.php?p=10164&. December 2006 Accessed: September 2007

[4] Telecom Reseller. *Using a Virtual Network to Ensure VoIP Success*. [on-line]. Available: http://www.telecomreseller.com/migration/voip_success_1.html. Last Accessed: September 2007

[5] Minoli, Daniel. Minoli, Emma. *Delivering Voice over IP Networks.* Wiley, Canada 1998

[6] Walker, Q, John. Hicks, T, Jeffrey. *Taking Charge of Your VoIP Project*. Cisco Press, USA 2004

[7] Riley, Charles et al. *Best Damn Cisco Internetworking Book Period*. Syngress, USA 2003

[8] IT Facts. *VoIP*. [on-line]. Available: http://www.itfacts.biz/index.php?id=C0_28_1. Last Accessed: September 2007