Literature Review

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

Abstract

The world has been changed by the way we communicate across a geographic area. This project describes 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 looks at the dynamics of how to manage voice and video communications between SIP agents across different countries. It also looks at how best to implement traffic shaping for networks with limited international bandwidth. This project aims to be a blueprint for connecting African universities together via VoIP and video on a data network.

Introduction

Communication is one of the most vital tools that human beings use to survive and progress from day-to-day. With this in mind the telecommunications industry has been evolving since the 1800s when Alexander Graham Bell invented and patented the concept of the telephone [6]. In his day he envisioned telephony to be the dominant communications network over the telegraph network. In our time VoIP is fast becoming the dominant communications network over the traditional *Public Switched Telephone Network* (PSTN) [6]. VoIP may be considered to be the next generation to PSTN and by this association may be part of *Next Generation Networks* (NGNs). NGNs are described as key architectural evolutions in *Information Technology* (IT) and telecommunication (mobile, PSTN) [4]. NGNs have developed at a remarkable rate in the last few years. This has been due to the greater use of digital technologies related to the Internet. This literature review tries to show the in-roads made by VoIP into the traditional PSTN and subsequently answer why VoIP is part of the NGN. What is PSTN and why is there are need for change?

Public Switched Telephone Network (PSTN)

PSTN is a traditional network that has stood the test of time. The changes that have taken place over the years have been in the area of switching technologies and the media used to transport voice over the network. The telephone service provided by the PSTN is called *Plain Old Telephone Service* (POTS). POTS use circuit-switched connections. A circuit-switched connection is one that creates a dedicated line/link between two communicating devices. Through gradual development and refinement PSTNs have come to achieve great quality and reliability of service that we take for granted. The level of reliability that exists is famously known as the '*five nines*' because PSTNs are guaranteed to be 99.999% up and running [8].

What components are present to set up a modern PSTN to provide telephony services?

- Voice Encoding
- PSTN switches
- Private Branch Exchange (PBX)
- Signalling

- Legacy/Traditional Telephones

Voice Encoding

This is an essential component in the PSTN setup. When two end users wish to communicate they produce audio data. This data must be sent via some media from source to destination. When a user speaks the audio data is received by the telephone handset and sent to the entry point of the PSTN. At this point in modern PSTNs the audio data is converted into digital data and transmitted through the PSTN to the destination. At the destination's PSTN entry point the digital data is converted into audio data and output through the telephone handset. The process of coding and decoding the audio data is called voice encoding. There are several types of encoders available but the one used by PSTNs is G.711 known as the *pulse code modulation* (PCM). There are two types of PCMs, the G.711u and G.711a.

PSTN Switches

The switches are the ones that create the paths through which the voice data travels. There are two kinds of switches, a local switch and a Tandem switch. A local switch is one that connects the user to the PSTN and the Tandem switch is one that interconnects local switches. The link between a switch is a trunk. A trunk contains multiple voice channels.

Private Branch Exchange (PBX)

This device is usually found in corporate offices and/or small or home offices. The primary goal of this device is to act as a gateway between a private telephone network and the public telephone network. It also has added capabilities and features such an internal switching, conference calls, caller IDs and call waiting.

Signalling

Signalling is used when a voice call is set-up. Signalling helps create a dedicated path for a voice call. These signals inform network devices that a call needs to be set-up, torn-down, or that a device is faulty or unavailable. The signalling protocol that is widely used is *Signalling System 7* (SS7). The data transfer of SS7 does not take place in the same network path as the call. It is made up of *Signal Transfer Point* (STP) and *Session Control Point* (SCP). When a call is made SS7 will determine the best path to create a dedicated link for the communicating devices.

Legacy/Traditional Telephones

Telephones connect to the PSTN. There are two types of telephones analogue and digital. The older one is the analogue telephone which is found in homes. The digital telephone is more recent and usually found in offices where there is a PBX. Figure 1 shows a PSTN using SS7.

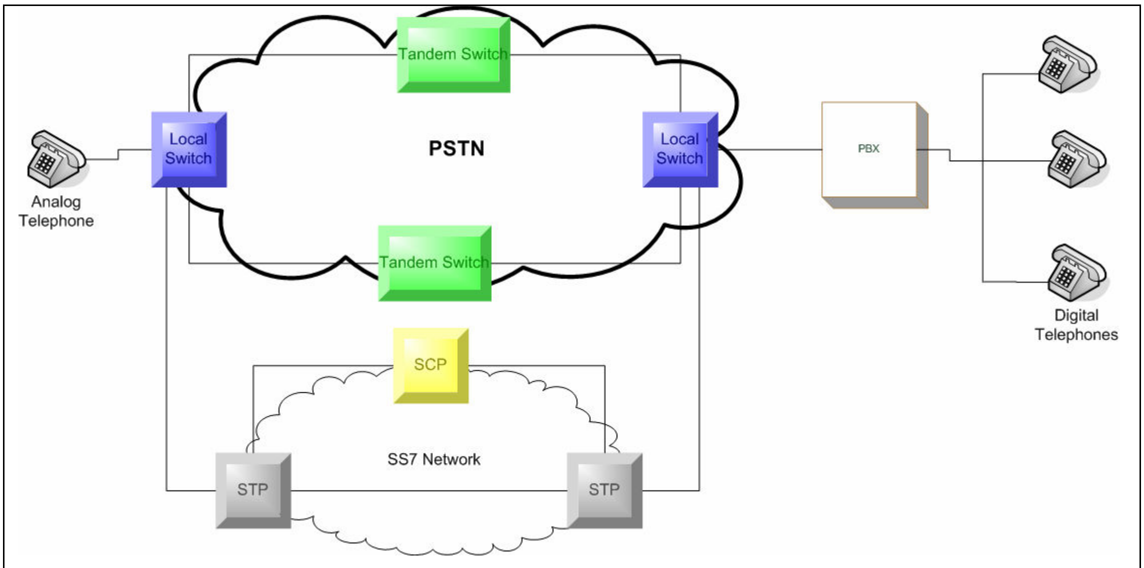

Figure 1: PSTN using SS7 – diagram adapted from: [8]

Figure 1 illustrates how the components of a PSTN are connected to provide a communication service. Taking these components into consideration VoIP should not only replicate but better the current structure of PSTNs so as to make it more scalable, attractive and efficient. Similar to what the telephone was to the telegram, VoIP is to the telephone. For VoIP to work there must be a seamless marriage between PSTNs and data networks (the Internet).

The Internet

The Internet is an international collection of interconnected computers across the world and spans over majority of industries and institutions (business, education, and government) [7]. The Internet has evolved from being an experimental way of transferring data between users in the 1970s to a commercial market place for virtually anything. The main protocol that drives the Internet and keeps data flowing is the *Internet Protocol* (IP). For devices to communicate across the Internet they need to follow a set of rules to ensure data transfer/exchange. ARPA conceptualised and funded the development of ARPANET which evolved into the Internet. ARPANET worked on packet-switching as a method of connecting communicating devices together [23]. ARPA initiated research on network standards that all communicating devices should adhere to and this resulted in the *Internet suite of protocols* now known as *Transmission Control Protocol/Internet Protocol* (TCP/IP). Small networks called *Local Area Networks* (LANs) are a set of devices that communicate over a small geographic area and are administered by a single network administrator. LANs also needed a set of guidelines on how to communicate and this resulted in the now popular Ethernet technology [7].

What components are needed to have an Internet connection?

- Internet Service Provider (ISP)
- Computer

Internet Service Provider
An ISP is a service provider who offers access to the Internet for a fee. There are two main links that a subscriber can establish an Internet link through an ISP (dial-up or dedicated line).

Computer
The computer in this case is a relative term used to represent a device an ISP subscriber uses to communicate with the ISP gateway or remote host. There are several methods through which a subscriber can establish an Internet session through the ISP (telephone line, wireless, or router). Figure 2 illustrates a basic user request process.

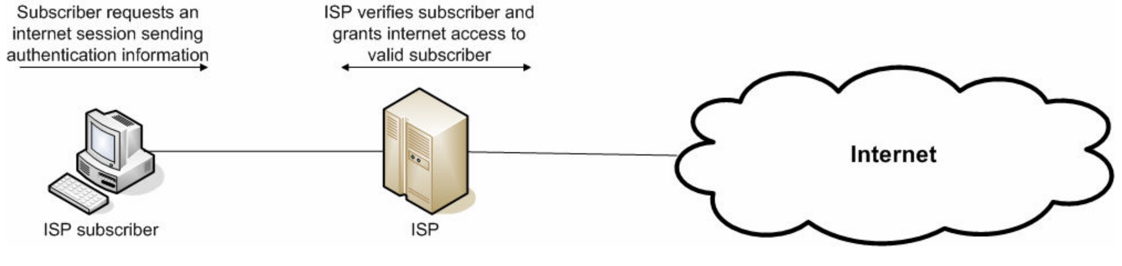

Figure 2: Internet subscriber request process

Although data networks were initially intended for file transfer/exchange it has numerous capabilities and this is what makes it the perfect candidate for the convergence point for all aspects of ICT. One such growth area that is converging towards data networks is real-time voice and video transmission.

Voice over Internet Protocol

VoIP has become an emerging competitor to the dominant traditional telephone network. VoIP essentially is the transmission of voice over a data network. Standards and protocols have been inherited from existing data transmission protocols and standards to ensure that voice is transmitted over a circuit-switched link from source to destination [9]. The question to be asked now is how does VoIP work and how much better is it than the traditional telephone network? Two end systems wish to communicate. The source speaks and their analogue data is retrieved and converted into digital data for transmission across the data network to the destination. At the destination the digital data is converted back into analogue data. This is a very simplistic overview of how the process actually works.

Similar to a traditional telephone network the following components are required for a VoIP telephony service [8].

- Voice Encoding
- TCP/IP and VoIP protocols
- IP telephony servers and PBXs
- VoIP gateways and routers
- IP Telephones

Voice Encoding

Just as in a traditional telephone network analogue data needs to be converted into digital data so that it can be transmitted from source to destination. As opposed to the traditional telephone network there are various voice encoders. The reason for this is that the data network on which voice is transmitted varies in bandwidth size. It is required to choose an optimum *codec* (codes and decodes audio data from audio to digital and digital to audio respectively) to ensure an optimum quality of service. A codec converts the analogue waveform to a digital form. Table 1 shows the different codecs available:

Codec	Bandwidth	Sample Period	Ethernet Bandwidth
G.711 (PCM)	64 Kbps	20 ms	95.2 Kbps
G.723.1A (ACELP)	5.3 Kbps	30 ms	26.1 Kbps
G.723.1A (MP-MLQ)	6.4 Kbps	30 ms	27.2 Kbps
G.726 (ADPCM)	32 Kbps	20 ms	63.2 Kbps
G.728 (LD-CELP)	16 Kbps	2.5 ms	78.4 Kbps

Table 1: Codecs - table from: [11]

TCP/IP and VoIP protocols

Data networks rely on the TCP/IP suite to ensure data is sent and received on the network. TCP ensures that data sent is received by the destination device. IP ensures that the path from source to destination is known. It is important to note here that due to the fact that VoIP transmissions occur in real-time voice packets that are lost are not recovered because this will lead to delays and thereby creating a poor quality of service [8]. There are several VoIP protocols used to set-up and tear down sessions but the one that is most prominent is *Session Initiation Protocol* (SIP) [16]. In a VoIP network SIP maybe used for creating and tearing down voice sessions and UDP is used to encapsulate the packets carrying voice data.

VoIP protocols are higher-layer protocols that work hand-in-hand with lower-layer protocols. SIP uses the open architecture of the Internet to set-up and terminate VoIP calls. SIP inherits the HTTP structure and this is what gives it an extra edge over other protocols. When a VoIP call is set-up, real-time voice streaming needs to take place. The VoIP protocol used is a higher-layer protocol called *Real-Time Transport Protocol* (RTP). RTP is encapsulated in UDP as it travels across a data network. Figure 3 illustrates the SIP trapezoidal model:

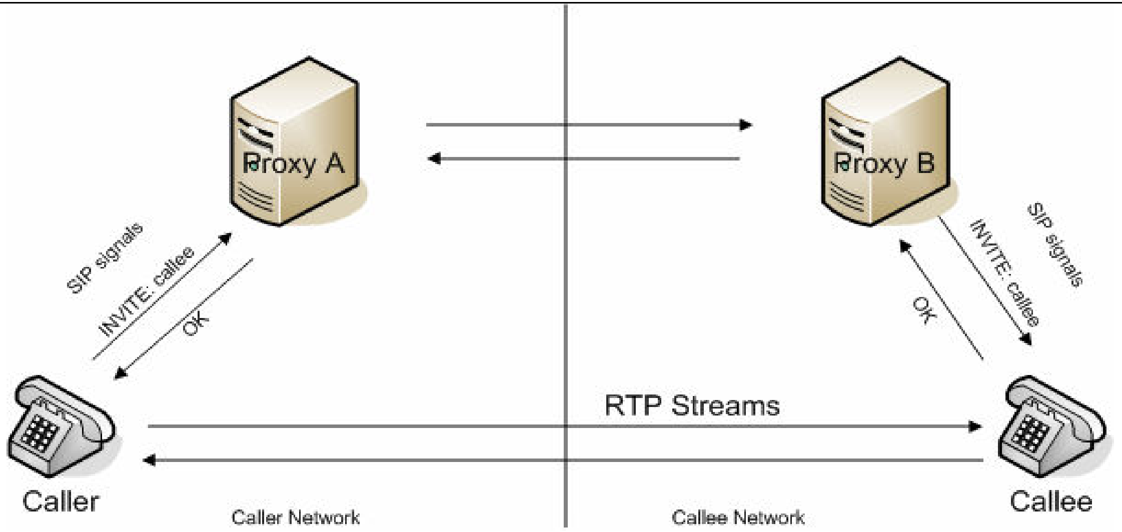

Figure 3: SIP trapezoidal model – diagram adapted from: [17]

IP telephony servers and PBXs

In a VoIP network all devices have client and server capabilities because they can initiate (UAS – User Agent Server) or accept calls (UAC – User Agent Client). There are three main servers to consider in a VoIP environment and they are proxy, registrar and redirect. The proxy server is used to route SIP requests to UAS and route SIP responses to UAC [22]. When a caller requests a call set-up with a callee the proxy server will act as an intermediate across the data network [12]. All calls will be created through the proxies which will have access to SIP URL to device mapping and will perform the search for the callee using the callee's SIP URL.

Registrar servers are used by the user agents to keep all devices aware that they are alive on the network. The SIP registrar server periodically receives updates from all devices on its network. This process also acts as a way of authentication that only authorised user agents operate on the network. SIP proxy servers make use of the SIP registrar server to lookup the location of user agents when calls are initiated [12].

The SIP redirect server is used to return error messages when destinations are unknown. The redirect server does not forward SIP INVITE requests instead it returns the destination IP address of the requested device to the UAS in a 3XX class response message [12].

PBXs are used in the same way as they are used in a traditional digital telephone network. They act as a gateway/ telephony router for a set of local set of IP phones.

The PBX provides voice services for all clients on a network such as conference calls, call waiting, call forwarding and many other customised services.

VoIP gateways and routers

The VoIP gateway provides a connection between the VoIP network and the PSTN. Since PSTN networks still exist these gateways are still needed to maximize coverage of VoIP networks. Some VoIP gateways use SS7 signalling to make sure that voice data travels across the two networks. Since PSTN only uses G.711 codecs all VoIP data must be converted to G.711 when they differ. In some VoIP deployments routers are used to determine the paths that VoIP traffic should travel. This means they should be able to disassemble VoIP packets retrieve necessary information to determine the next hop and reassemble them and send them off. Most networks use Ethernet technologies which makes it easier for path determination.

IP Telephones

These are essential components in the VoIP network in that they are the end systems. The analogue data is input into the IP telephones which in turn convert the audio data into digital data. If the devices are old then the PBX will perform the conversion. One thing to consider is that the UAS and UAC must use compatible conversions, if they don't this will give the PBX some overhead in ensuring that the digital data is compatible [8]. Figure 4 illustrates a connection of PSTN and VoIP network via a VoIP gateway.

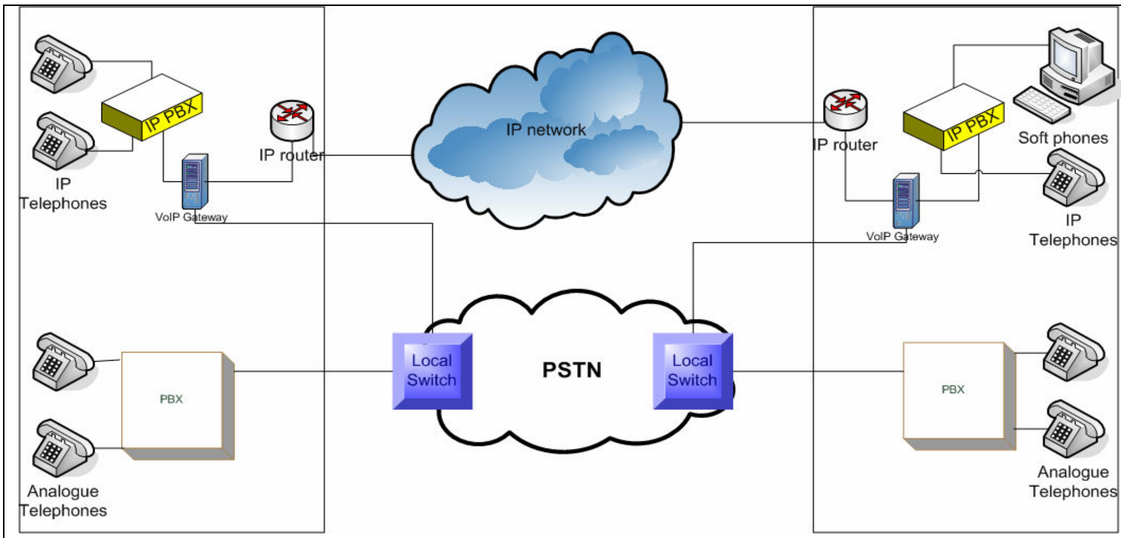

Figure 4: VoIP network connected to PSTN - diagram adapted from: [8]

Asterisk

Asterisk is an open source IP-enabled PBX. It was developed by Mark Spencer of Digium Inc in 1999 and was released under the GNU General Public License (GPL) [13]. Asterisk is software running a PBX system and offers existing traditional PBX services and tailor-made services for its users [6]. The beauty about Asterisk is that it can talk to any communications device from the 1960s to the latest digital wireless VoIP telephone because it supports many codecs, protocols and traditional telephony standards. The word asterisk was chosen because it is a key on the telephone keypad and is also used to mean anything is a search string [10].

iLanga

iLanga is a computer based PBX which was integrated in the Computer Science Department at Rhodes University [2]. iLanga has three core open source components [1]:

- Asterisk
- SIP Express Router (SER) – handles SIP traffic
- Gate Keeper (OpenGK) – handles H.323 traffic

Asterisk on its own cannot cater for large scale VoIP networks and that is why the iLanga core has a SER and OpenGK as core components. Figure 5 illustrates how the components combine to create a more effective VoIP box:

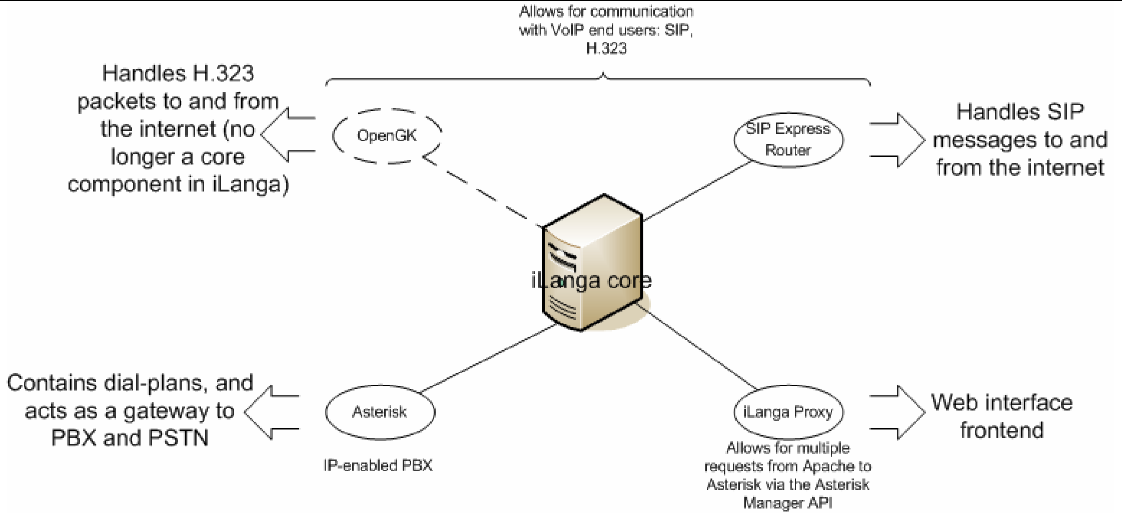

Figure 5: Components of iLanga core

In Figure 5 it should be noted that OpenGK is no longer in use as iLanga prefers SIP agents over H.323 agents. The iLanga core is a suitable fully functional PBX to implement for VoIP deployment and that is why we will be using it for this project.

University of Namibia (UNAM) and Rhodes University (RU) will both have these boxes to on their respective LAN. Figure 6 illustrates the VoIP network:

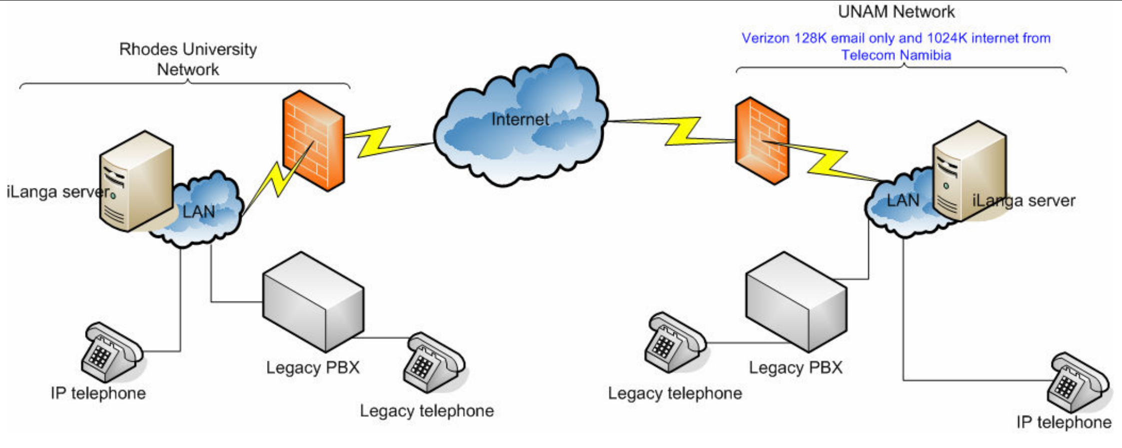

Figure 6: Project deployment of VoIP network

Quality of Service

What does this mean for a VoIP network? The purpose of Quality of Service (QoS) is to deliver data from source to destination in the bounds of a set time. VoIP is a real-time interaction which relies heavily on voice data arriving on time at a desired destination. If voice is delivered with a maximum delay of 300ms a decent voice conversation can occur but if there more than 500ms of delay then the conversation becomes overlapped and awkward [10]. The main drawback that most VoIP implementations have over large networks is the quality of voice. Skype, the first VoIP client based on peer-to-peer technology is one of the most popular VoIP tools available today. In the early release it did not always adequately deliver voice across data networks as the traditional PSTN do [15]. Many Skype users bear with this poor service because of the cheap costs that VoIP provides as compared to PSTN calls. The minimum uplink and downlink bandwidth required by Skype to deliver reasonable voice output is 2 kilobytes pre second anything below this renders a call unintelligible [15]. QoS is a key factor to consider when deploying a VoIP network especially across a large data network and that means that traffic shaping can be in implemented as a way of meeting the needs of QoS.

SIP vs IAX

Quality of service has a major role to play in the livelihood and ascendance of VoIP. If one is to enjoy VoIP the time it takes to establish a call must be short, the rate of voice packet drops must be negligible, and the amount of bandwidth used must be low. SIP is a published and well documented *Internet Engineering Task Force* (IETF) standard and that gives it credibility. *Inter-Asterisk Exchange* (IAX) is a protocol developed by the creator of Asterisk Mark Spencer to solve SIP drawbacks. IAX is a protocol built with VoIP in mind [18]. Table 2 compares the two standards.

	SIP	**IAX**
Standard	IETF standard	Not a published standard (yet?)
Ports	Requires 3 ports to achieve voice communication. 1 port for call setup and 2 ports for voice transmission (RTP)	Requires 1 port to achieve voice communication
LANs	Not costly (data flow) on LANs because of less hardware requirements. RTP does not flow through servers on a LAN	Costly (data flow) on LAN because of hardware requirements. IAX must flow through servers at all times
WANs	Costly on WANs (security) because RTP requires a range of ports to be available for VoIP sessions	Not costly on WANs (security) because less ports are open for VoIP sessions
NAT Handling	NAT traversal is a problem because signalling and voice data travel on different ports. During voice data transmissions the mapping of private to public IP addresses may mismatch	NAT traversal is not a problem because signalling and voice data travel together on the same port and therefore IP address mappings remain the same throughout a session
Complexity	A more intensive implementation process required because it is a general purpose protocol	Less intense implementation because it is an Asterisk based protocol

Table 2: Comparison between SIP and IAX – comparisons from: [18], [19]

For this project SIP will be used. Since there is competition among packets to reach their desired destinations VoIP traffic will have to be prioritised to ensure a sustained end-to-end voice connection. Traffic shaping is the answer to prioritisation.

Traffic shaping

Traffic shaping is a form of controlling how packets should flow through a network. The essence of traffic shaping is to guarantee performance [20]. The reason why traffic shaping is important is to ensure that voice is not delayed by more than 500ms and few voice packets are dropped with in a session. For this project traffic shaping must take place at the ISP. Figure 7 illustrates where traffic shaping should be implemented on the router interface that connects the ISP to UNAM.

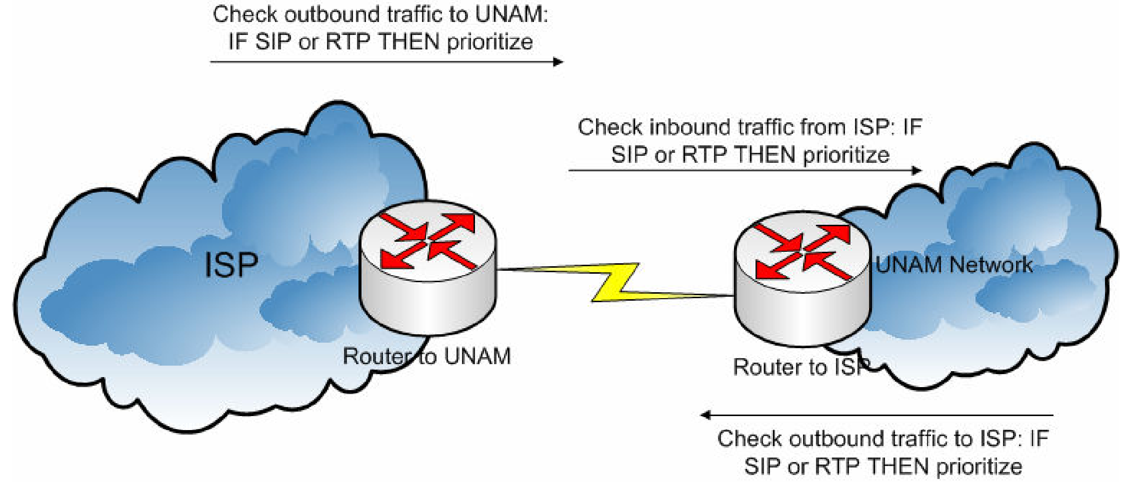

Figure 7: Traffic shaping between UNAM and ISP

In Figure 7, when UNAM bound traffic arrives at the ISP each packet is put into a queue. Each packet is stripped and analysed to determine the class/type of the packet. If the packet is SIP or RTP then it is put into a priority level queue [21]. This means that all traffic in the priority queue will leave the ISP for UNAM first. The same process of class/type determination occurs on the UNAM router for inbound and outbound traffic. The main points to consider are whether the end routers can perform traffic shaping and determine VoIP protocols.

Conclusion

In this literature review telephony aspects were described so that a firm understanding is gained on the PSTN and VoIP networks and how they work. This leads to a better understanding on how to best tackle the deployment of VoIP at UNAM and to set-up a VoIP link between UNAM and RU. To obtain an optimum link between the two institutions an appropriate bandwidth calculation is required so as to choose the best codec to use between the two links. Due to the fact that UNAM has a 1MB international bandwidth connection shared campus wide, a QoS policy needs to be put in place so that a sustainable link is established for any voice session. No matter how bad the internet link is, a call should be established between the two sites.

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