Fall 2006

Evaluation of Voip Technologies As a Replacement for Traditional Pstn Based Pbx Systems

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A Project/Practicum Report submitted in partial fulfillment of the requirements for the degree of Master of Science in Computer Information Technology

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Date 08, 2006

An Abstract of a Project/Practicum Report Submitted to Regis University School for Professional Studies in Partial Fulfilment of the Requirements for the Degree of Master of Science in Computer Information Technology

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This project deals with a company in the SME sector with offices located in the midlands of Ireland. The company is well established in the field of Agri-Feed Manufacturing Process Control Systems or SCADA systems, and has been established for over 20 years. The communications requirements of the company have changed over these 20 plus years to a mix of various technologies from PSTN lines to Broadband ADSL. The present telephone system has been in use since 1991 and has several questions marks over it in terms of usage costs, usage reporting, support and maintenance and features available. This project is an evaluation of the possible benefits offered by the use of VOIP technologies and Asterisk Open Source PBX as a possible replacement for the existing telephone system in place. It attempts to look at the potential benefits costs, and risks associated with using such a system. A small pilot system is implemented and some key users test this and feedback on its usability is recorded. The current communications infrastructure is analysed in an effort to highlight systems where cost savings or benefits can be made by switching to these other technologies and a report was presented to the management in order to give the required information to make the best possible informed decision about the way forward for the company.

Acknowledgments

Eternal thanks to my wife Marcia, without your support this would not have been possible, also to my family and friends and all my work colleagues for their assistance and encouragement during my studies. Also to Des Chambers who provided some very insightful assistance.

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Chapter 1 Introduction / Executive Summary

The Company to whom this project refers to will remain un-named and be referred to as XYSystems wherever necessary.

Problem Statement and existing situation

XYSystems develops and supports SCADA systems for the Agri-Feed industry with customers mainly in Ireland and the UK. Its office is located in the Midlands region of Ireland. The company prides itself on its high levels of service and support it offers its customers. These support contracts include 24/7 year round support. To achieve these high levels of support the employees depend heavily on the communications infrastructure in place. Several key components make up this communications infrastructure but they all rely on the underlying voice and data communication lines in and out of the company. At the heart of the voice communications in and out is a 15-year-old Private Branch Exchange (PBX). This has been in place since 1991 and has several limitations that are to do with the changes in technologies in the last 15 years. The original manufacturer/supplier has been taken over and this new company can only provide best efforts support due to the age of the system. Local IT staff have limited knowledge and details about this system and can only perform routine tasks such as additions, moves or time changes. There are no reports or records available from the system at present and as such management cannot analyse usage information. Some features that are standard on a modern telephone system are absent such as Voicemail, Least Cost Routing, External Call Forwarding and Automated Attendant. The system as it stood functioned very well but it presented an unknown business risk going forward.

On the data communications side the company has had Broadband installed from August 2003. This was also used for supporting customers remotely where possible. There are also several Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN) dial-up modems and routers, which are used for remote support to customers with older systems or systems not yet connected through the internet using Virtual Private Networks(VPN's). Also there are some employees working from home and they use broadband to access the companies network. These employees tend also to use PSTN lines to communicate directly with the office. It was obvious to the company that there could be in-efficiencies in their current structure and possibly there were areas where they could save money and or provide better service and support.

Project Goals

This project's overall aim is to evaluate the telecom systems in place and to evaluate if there were any cost savings to be made for the company by moving some of its voice traffic from the Plain Old Telephone Network (POTS) across to utilise Voice over Internet Protocol (VoIP) technologies. A second goal of the project was to evaluate a VoIP capable software PBX which could be used to replace the current telephone system and as part of this goal it was hoped to set-up a small pilot system using Asterisk PBX to help evaluate this. A replacement system would offer more features. It would be better supported and could possibly offer some cost savings by using VoIP for some of the more expensive calls. Another goal of this project was to gain a better understanding of the communications infrastructure in place and to evaluate any possible risks to the company's business and look at areas where cost savings could be made.

Barriers and/or Issues

The existing telephone system works well and has a very high level of uptime, users are happy with the desk phones because they know how to use the main features of the system well. It provides good quality telephone conversations to and from the customer. Although the current PBX may have some inefficiencies, any system that hopes to replace it will have to be as least as reliable. The transition to a new system will have to be sensible in term of costs. Any savings made with call costs using VoIP will have to be weighed up against the costs in moving to VoIP. This is an important issue and costs will have a big role to play in the final decision. Call quality is important as some initial forays into VoIP have shown that at times there can be quality issues so this will need to be evaluated. Other issues include the inherent limitations of the existing telephone system. It may indeed be upgradeable in some areas but the chances are that it will not be part of an overall VoIP solution because of its age. The remote users will need special consideration because of their unique physical separation from the office.

Questions to be Answered

This project attempts to answer several questions. Firstly does VoIP offer any cost savings or benefits for the company? Also in terms of costs, where can any savings be made in relation to the overall communications infrastructure? Can a software PBX replace the existing hardware PBX? Also, are there any business risks associated with the current system?

Scope and Limits of this Project

Time is a big limiting factor as with all projects, this is being carried out at best effort alongside what can sometimes be a busy workload. Cost is another limiting factor and while the company may need to spend money to eventually save in the long term, the strategy for such a cost recovery will need to be clearly identified. Although part of the project involves the implementation of a small pilot project that will demonstrate the capabilities of Asterisk and VoIP, it is intended the pilot system will be used only to demonstrate the basic features of a more complete solution. In this testing environment the pilot system will use one analogue line and the companies ADSL broadband to connect to a VoIP service provider. In terms of hardware used, the testing was carried out with a small number of hardphones and softphones just to get a feel for the various different options available.

Summary

The current communications systems in XYSystems requires analysis to evaluate where, if any, cost savings can be made. It may be that utilising VoIP can make some cost savings, and this is to be investigated. The possibility of replacing the current system to allow savings, increase its features, or make it more maintainable is to be considered. In the project a pilot system will be setup to see first hand the features of a solution which offers to integrate VoIP, POTS & ISDN in a single solution.

The analysis and testing will serve to provide information to management as to the state of the communications infrastructure which in turn will allow for better management of this essential resource.

On a practical note this project will serve as a guide to others asking similar questions about their own systems and as such will be beneficial to an element of wider community in the small to medium sized business section.

Chapter 2 review of Literature / Research

Definition Of Terms

POTS	Plain old Telephone Service
PSTN	Public Switched Telephone Network
NAT	Network Address Translation
FXO	Foreign exchange Office
FXS	Foreign exchange Station
ISDN	Integrated Services Digital Network
BRI	Basic Rate Interface
PRI	Primary Rate Interface
DID	Direct Inward Dialling
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
SIP	Session Initiation Protocol
RTP	Real Time Protocol
ITU	International Telecommunications Union
PBX	Private Branch Exchange
ATA	Analogue Telephone Adaptor
IAX	Inter Asterisk Exchange Protocol
T1	Tier 1 Connection
MOS	Mean Option Score
QoS	Quality of Service

Overview of all literature and research sources on the project

The main focus of this chapter is to present to the reader the research and literature review that was carried out during this project. Literature on the relevant technologies was reviewed and there were several distinct areas researched here in an attempt to answer the questions posed in the project. The sources used were varied and included the Internet with search engines, user groups, wiki's and manufacture websites. Library sources were used through off-campus access and this provided access to previous academic work on the topics covered here. This included books, research/thesis papers. A wealth of information was found in online articles and online multimedia such as demonstration videos. Some of this came from enthusiasts in the area and also from end users.

In carrying out the existing literature review many sources were uncovered, some from manufacturers promoting a product, and some from end users detailing their experiences. This provided a balanced approach so as to prevent the sales pitch of the manufacturer from clouding the real details of a particular topic.

Research methods to be used in investigating the problem

Research for the project was carried out in several ways. Contact was made with several vendors and service providers in an effort to obtain relevant product information and pricing about costs for a replacement system and service charges associated. Several vendors arranged site visits to discuss face to face the requirements and to provide proposals for a replacement system. This was important as it provided the opportunity to interview a person face to face who has experience in implementing similar systems.

A survey of user requirements was also carried out to evaluate some of the features users would require of any new system. The management and staff were also queried in regard to their desires as to the system and the results of these surveys can be found in appendix 1.

The existing service providers were also contacted to provide reports and usage data. The relevant telecom bills were also analysed to provide indications of usage and existing costs.

Some research was carried out to determine the best approach to a pilot system and to understand how to setup a software PBX. This included fundamental issues like what Operating System would be used to run the PBX, to issues like what types of phones to test with. This again was done using Internet user groups and the main focus here was that any phone that would be tested had to have 90% of the features required by staff.

An analysis of the existing system was carried out to gather relevant information, which was necessary to evaluate its status in terms of upgradeability and support. Some of this research included searching for information using the Internet, posting to telecom user groups on the Internet calling spare parts vendors to assess availability of replacement parts. As was mentioned previously, the company that supplied the system originally had been taken over so the new company was contacted to research their support status and they sent out an engineer but he was more interested in replacing the existing system than supporting it as they could only offer best efforts.

Literature and research that is specific/relevant to the project

Although a lot of overlap occurs with the literature here, it can be broadly broken down in to these three areas.

- VOIP
- Asterisk
- Existing system

VOIP

Anyone who has been involved in the communications industry, either as an end user or supplier will be aware of the emergence of technologies from time to time which offer benefits and provide potential cost savings over the existing technology. An example of this is in the internet connectivity market, where in recent years the technology in Ireland has changed dramatically from a situation where you could only expect to have narrowband internet access in your home to a highly competitive marketplace for Broadband around the country. The market for Voice communications has experienced a similar shift and some technologies have become more mainstream. One such technology is Voice over Internet Protocol (VoIP), which is the routing of voice conversation over the Internet or through any IP-based network (Wikipedia 2006). VoIP has several advantages in terms of features over the traditional PSTN service. Using VOIP to route calls allows a person to receive calls as long as they have a connection to the Internet or the office network. It allows for the centralized management of call costs. It allows for more flexible integration with computer applications, such as click to call. To gain a better understanding of the background of VoIP and the specifics of what VoIP actually is used for, a good source of information is a book entitled "Voice over IP: Systems and Solutions" by R P Swale (Swale 2001). In chapter 1 the author points out why there is so much interest in VoIP because of the possibility of free telephone calls over the Internet and the potential effects for long-distance and international markets. According to Swale (Swale 2001) this hype has created a gold-rush effect that makes it difficult to separate fact from fiction and does not serve to clearly identify the areas where VoIP could be a sensible business venture. VoIP can be applied to many different applications and the most obvious is, of course making a long distance call over a broadband network to a long distance connection. It can be used to make calls to another VoIP user for free and it can be used to make a call to the PSTN network for substantially lower rates than making a normal PSTN to PSTN connection. Because of these cost savings there is huge growth in the popularity of applications such as Skype (SkypeLimited 2006), but there are other areas where VoIP is used. In a tutorial on VoIP and FoIP (Fax over IP) the ITU clearly identifies examples of applications which are made possible by using VoIP transmissions (Texas Instruments 2000). These include the branch office application where two or more branch offices are in separate locations but are interconnected by packet networks that are normally used for data transmission, but may be enhanced to also carry voice traffic. Jared Smith co-author of the book "Asterisk the Future of Telephony", can site an impressive example from his personal experience of an Asterisk VoIP deployment. "It currently handles approximately one million minutes of calls per month, serves several hundred employees, connects to 27 voice T1s, and saves the company around $20,000 (USD) per month on their telecom costs". A recent survey showed that although VoIP is an emerging technology and it's adoption is growing fast (Pereira 2006) with many business both large and small either evaluating or planning some VoIP technologies.

So it is obvious that VoIP is on the increase but what if any are the major issues or technical requirements for VoIP? Because of the nature of carrying voice-over-packet networks there are some inherent quality-of-service (QoS) issues. This is because IP packet networks do not provide for a guaranteed mechanism to ensure packets arrive in sequence (Wikipedia 2006). This can result in problems such as delay and packet loss which is identified by Hestnes et al (Hestnes 2003) as a characteristic of many networks and results in interference with real-time communications resulting in loss of quality and negative user perception. Also there is the fundamental issue of VoIP's bandwidth requirements. For a conversation between two endpoints to be a success, there needs to be sufficient bandwidth to allow for the transmission and receipt of voice packets. This may not always be available and in some situations dedicated networks are provided to ensure that the bandwidth is there. A case in point is the existing ADSL connection at XYSystems, which by design is intended to allow a large download bandwidth and a small upload capability to provide for the typical requirements of most Internet users. According to Brownworth (Brownworth 2006) a VoIP phone call requires bi-directional transfer with similar bandwidth requirements in both directions. Another issue identified (Melvin 2004) as causing delays and problems for VoIP are mouth-to-ear (M2E) delays experienced when using soft phones caused by different clock speeds on each sound card and a mismatch between the sound card driver design and VoIP application design. Understanding that these issues are present will help when it comes to designing any VoIP system that attempts to provide the end users with quality real-time communications. Some of the metrics used in the measure of the quality of a connection used for VoIP are

- Jitter – this is a measurement of the time variation between packets sent and packets arriving. In order for a VoIP call to work well this variation will need to be below a threshold that usually represents the amount of time it will take the receiving end to re-assemble the packets into a stream of audio that sounds like what was sent.

- Packet loss – simply put, some of the sent packets were dropped for some reason on the way to the sender and high rates will result in poor quality
- Packet Discards – here the packets are discarded because they arrive too late to be re-assembled into the audio stream.
- MOS or Mean Opinion Score – this is a measure of the perceived quality of the reconstructed audio (and/or video) after its transmission and compression. It is based on a scale of 1 to 5 where 1 is bad and 5 is excellent. It is based on recommendation P.800.1 from the International Telecommunications Union.

In this project the intention is not to get too involved with the more technical aspects of VoIP but to gain enough knowledge in order to evaluate it as a potential mechanism for voice communication. But it is important to be aware of the pitfalls. Some useful VoIP test programs are available on the Internet, which allow you to continually monitor your VoIP quality such as MyVoipSpeedServer from Visualware (2005). Initially some tests were carried out with this software to get a feel for the MOS score. Some testing was also carried out using Iperf to simulate the bandwidth requirements and packet sizes used in typical VoIP calls. According to Reynolds et al (2001) there is a technical challenge in delivering high quality speech while achieving high network efficiency . They discuss the aspects of a VoIP system design which have the greatest bearing on user perceived speech quality. Some of these include the type of codec or speech coding mechanism used. And they point out the highest quality codec used being G.711. It is also pointed out here that there are other codec's that employ compression that brings down the bit rate but also reduces the quality such as G.726 or GSM.

Asterisk

The second major area of literature review and research was into the Open Source PBX system Asterisk. A lot of good sources of information were available to help get a good understanding of what Asterisk is capable of. The wiki on voip-info.org was found to be a great starting point and lead to many other useful links on installing configuring and understanding asterisk. Asterisk is a software version of the hardware PBX and it runs mainly on the Open source Linux operating system. It was created by Mark Spencer and helped along the way by Jim Dixon and an enthusiastic open source community (Meggelen 2005). It was born out of a frustration at the high costs to purchase what seemed like a very basic telephone system for Mark Spencers Linux support business. His limited funds forced him down a road that was to eventually lead to the creation of the Asterisk project (Wen 2006), which is now at the heart of many sophisticated corporate phone systems (Charny 2005). It allows for the convergence of VoIP, PSTN ISDN. It's feature length is quiet extensive and I would argue that if it's not in this list its probably not a very common feature. Some of the main features include Voicemail, Conference Bridging, Call Queuing, and Call Detail Records (Digium 2006). Along with the features mentioned Asterisk also supports Computer-Telephony Integration allowing you to integrate applications with your dial-plan, it is scalable and supports many different codec's and Protocols. It can integrate with old analogue phones using analogue terminal adapters (ATA). So on the surface it seems to offer most of the requirements for a modern telephone system. According to one source (Meggelen 2005) asterisk is fuelling a revolution in the telecommunications sector, it is enabling the convergence of voice and data technologies. Another good source of information used in this project on asterisk was a book on building telephone systems with Asterisk (Gomillion 2005). This book is very good also as a starting point as it introduces some of the terminology to the novice user without going into too much detail but it does assume a working knowledge of Linux. It was because of its open source nature and its ability to replace the traditional telephone system and bring on board VoIP, that it was chosen for this project.

Linux was of course one of the other areas where literature review and research was carried out. Again the user forums and wiki's were very helpful for tips on how to install asterisk for the different distributions of Linux. One site that was very informative was a podcast site (Asteriskast 2006), which actually had some video demonstrations of setting up asterisk and these helped in getting the pilot system going.

When the pilot system was ready several tests were carried out to check if it was capable of performing some of the features requested and also feedback was solicited from the end users as to their experiences with it. Test's were also carried out to analyse the performance of the networks between the central and remote offices using an IP network performance testing tool called Iperf (NLANR 2006).

The existing telephone system and literature associated with it was also reviewed. Unfortunately there was very little documentation about this system available. Some information about the system was found from the Internet that gave a good overview about the system but didn't really get into detailed technical information. It was determined that there may be one solution to using the existing system with VoIP and this would need to be tested to see if it would work. The principle can be visualised in the following picture taken from one distributors (Kentel 2006) website.

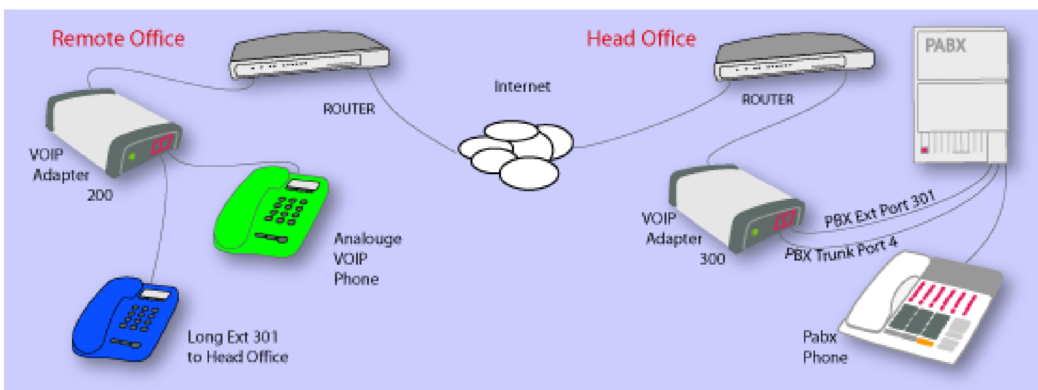

Figure 1 – A potential solution using VoIP with the existing system

This VoIP adapter could be used to plug in to the PBX and a similar unit sent to the remote offices. The same supplier was a stockist of refurbished parts for the existing PBX, so if necessary be they could be contacted to evaluate costs and availability.

Knowns and unknowns about the project

There were several unknowns at the beginning of the project but the major one was the existing system. What was its status in terms of support going forward? Was it reasonable to assume that the company could expect that any problems encountered with it were easily fixable or was it a potential business risk that could cause havoc if some un-repairable fault occurred in it that could leave the company without lines and hence their customers without a proper support mechanism. Other unknowns existed about the system from a technical viewpoint. If it were to be replaced, how could one identify of which lines were data and which were voice? Could some lines be moved over and the rest left in place in a transition from old to new?

Also in terms of VoIP and ensuring quality, was the current connection good enough? If not what type of a connection was required for all or some of the current usage? Would any system based on PC be robust enough could it handle all the possible users without comprising on quality? Would there be a need to separate the VoIP infrastructure onto its own network because of security issues or quality issues? If a new system were implemented how much would it cost? How long would it take and what would be its standing in terms of support and maintenance. And of course would there be savings on the cost of communications and when would this happen immediately or after some period to payback costs?

What was known was, that if a new system were to be implemented it would have to encompass the features of the old system. It would also need to have management reporting abilities. Any equipment used would need to be very easy to use and training would need to be given to all staff.

The contribution this project will make to the field

The project will serve as an example of how a small business that relies heavily on telecom services can move forward with new technologies to provide better services and cost savings. It will be helpful to anyone in a similar situation who is considering his or her options but doesn't know where to start. It also may demonstrate to others how Asterisk can be used and give others the encouragement to try it for themselves.

It will provide for the company a better understanding of the current system and help management make decisions as to the best way forward. It should also help the IT staff to understand the system as some of the results of the systems analysis will detail the current system.

Chapter Three – Methodology

Formats for presenting results/deliverables

The results/deliverables from this project include a variety of items including bar charts to show usage figures, some diagrams depicting/modelling how the telecom system in the company is currently comprised and how it could evolve if it were to move to VoIP. There are also some cost figures relating to an alternative solution suggested by one supplier. These were presented to management in a separate report.

Life-cycle models to be followed

This project follows along roughly the systems development life cycle but as an overall project within a small company it has some aspects that are different. It starts out with the planning stage where the problems with the current system are identified and a decision to investigate solutions and alternatives is made. In this stage it was decided to plan the project as follows. To carry out an analysis of the current telecommunications infrastructure at the company and to look at the usage costs and the possibility of using VoIP for some of the voice calls as a possible cost saving mechanism. In this stage it was also realized that moving to a VoIP solution with the current system was probably not likely given the age of the system and it's unpredictable future, so the company would have to consider some other type of telephone system. It was at this stage that and idea for the for a practical project was required. This was chosen as it was a good candidate and some project advisors were contacted. Des Chambers from NUI Galway agreed to take on the roll of project advisor and made a suggestion that Asterisk could have some application here. Not having come across Asterisk before some preliminary research was carried out into Asterisk to evaluate its potential. After reading some of the articles from the user forums it was obvious that it would be a good candidate for testing VoIP technologies and evaluating a replacement PBX system. There was already a good background in Linux from working with it for several projects and some excitement about the idea of using an OpenSource product that would have all the capabilities suggested by Asterisk. So here in this stage it was decided that Asterisk would be used as a pilot system to aid in evaluating VoIP technologies to see if they offered any potential for XYSystems.

From the initial planning stage the project moved on to the analysis stage where the current system was studied and evaluated. The relevant phone bills were gathered detailing a six-month period up to Jan 2006 and these were scrutinized for information in relation to call costs and usages. An evaluation of the requirements of a replacement system was also carried out so as to understand user and management requirements of any possible replacement. The results of the user and management requirement analysis feed directly into the next stage which is the design of the pilot test system. An analysis of the alternatives available using proprietary systems was also carried out to evaluate its merits against an Asterisk system.

Then the required hardware was gathered together and this pilot system was then implemented and tested to evaluate its merits and to see how the test system worked in a real life situation. The last stage of this project was where this project differed slightly from the traditional SDLC model which would normally be involved in maintenance and support of a new system, but this stage was not required as the system was only a test system, but what was carried out here was an evaluation of the results of users experiences using various aspects of the pilot system. Its merits were also evaluated at this stage on a cost basis against a proprietary system and conclusions were made. Also carried out here was a presentation to management of the findings and conclusions made.

The Planning stage

This stage of the project began last year with the acknowledgment that the project to be undertaken was to review the current offerings in the VOIP marketplace followed by an analysis of our current systems in terms of cost and functionality. A meeting with my project supervisor helped clarify some of the objectives and at the time it was also decided that a small pilot system would be created to aid in this evaluation. As indicated in the literature review, sources relating to VoIP and Asterisk were researched in an effort to gain a better understanding so as to plan how to set-up a suitable test system. Initially it was decided to try and adhere to the following project plan, which gave some room for change if the project came up against some unforeseen obstacles.

Date	Task to be completed	Duration	Finish Date
01/09/05	Internet Research	5 months	01/02/06
20-Feb-06	Analysis of current system	1 week	27-Feb-2006
27-Feb-2006	Requirements gathering	2 days	01-Mar-2006
02-Mar-2006	System Design	2 days	03-Mar-2006
06-Mar-2006	Supplier Research	3 day	08-Mar-2006
20-Mar-2006	Installation of Linux	2 days	21-Mar-2006
24-Mar-2006	Install and config of asterisk	2 days	25-Mar-2006
10-Apr-2006	Hardware install	2 days	11- Apr -2006
17- Apr -2006	Telephone Configuration	2 days	18- Apr -2006
01-May-2006	Asterisk Configuration	10 days	9-May-2006
22-May-2006	Asterisk Testing	3 weeks	12-June-2006
26-Jun-2006	Analysis of findings	10 Days	7-Jul-2006
28-Jun-2006	Presentation to management	1 Day	28-June 2006

Table 1 - The project plan

Analysis Stage

Usage and rental analysis

The first major task was to gather information in relation to the existing telecom services both in terms of usage and equipment rental. Most of this information came from the telephone records from a six-month period up to and including Jan2006. The information gathered from the bills was very basic. It showed what services were being rented from Eircom and the costs associated. This is shown on the following table and refers to rental costs only, not usage. It shows the costs for each service and the amount of that service rented from Eircom.

Service	**Cost**	**Number of lines**	**Total Cost**
PSTN Line	19.98	15	299.70
ISDN Line	30.99	2	61.98
Broadband	169.00	1	169.00
		TOTAL	**508.04**

Table 2 – Existing Rental and Service Costs

Then the minutes used for each phone line were recorded against the relevant line and a bar chart, which depicted usage for each telephone line, was created to visually inspect the figures. This was used to highlight lines that had a large amount of calls so we could see if there was a possibility to put VoIP on these lines. These details on the usage were recorded into a spreadsheet so some calculations could be carried out against the figures. We ended up with average monthly figures for each line for the 6-month period.

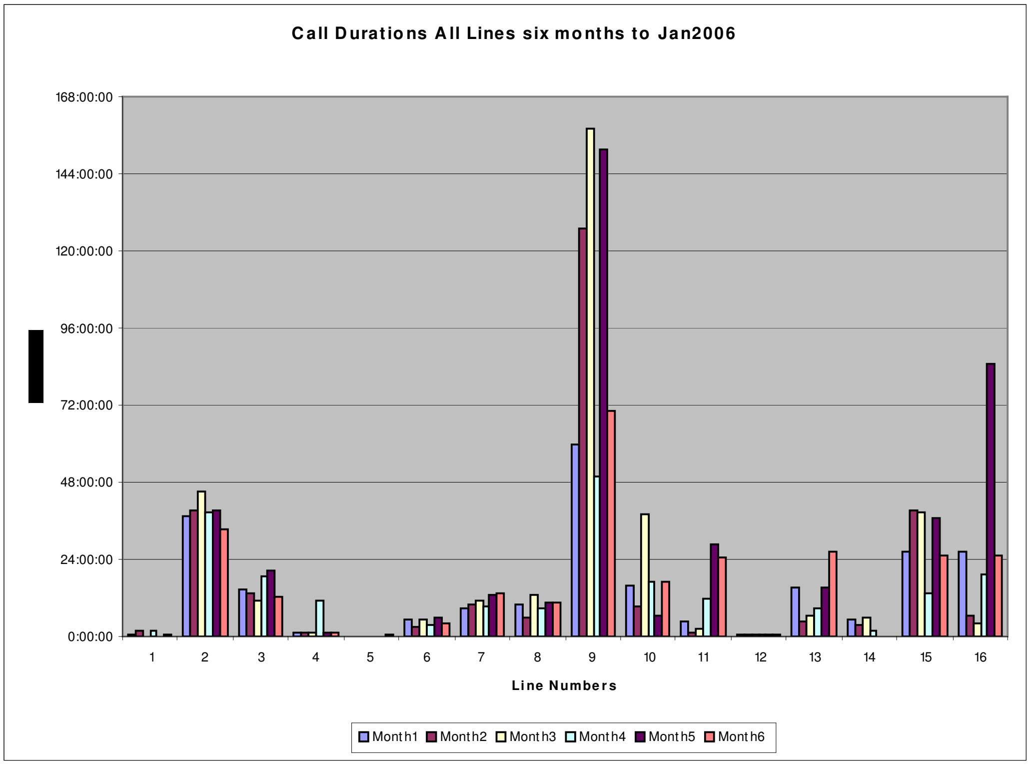

Figure 2 - Duration of calls for Six Months.

In the above chart Lines Numbers 1-8 are the Voice Lines and the rest are Data. From this it can be deduced that the data line usages are higher than the voice. It was expected that the data lines would have high costs because some of the calls were for long periods to the UK during peak hours. So on a six-month basis it was determined that the ratio between voice and data costs was 2:1. It is also evident that some data lines were extremely expensive on an ongoing basis and this would require further investigation as to the reasons for this but that is outside the scope of the current project as it is only concerned with voice costs. From the figures it can be seen that data calls represent twice the duration of voice calls as visualised in the pie chart in figure 2.

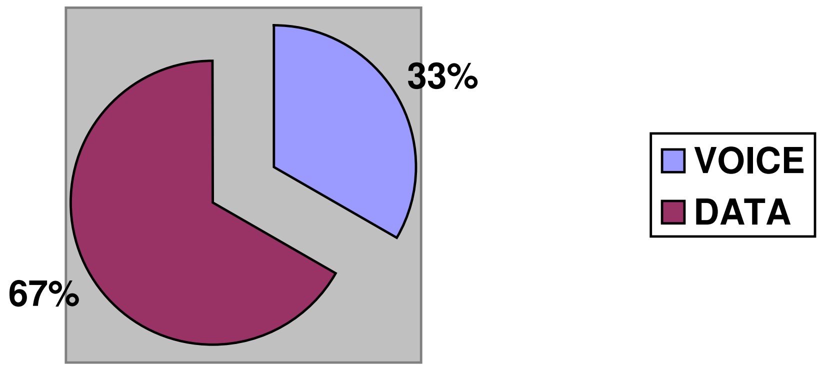

Figure 3 – Ratio of data to voice usage.

Of the voice calls there is a variety of call types, for example some were to mobiles and some were to international destinations. The majority of these phone calls were for local/national and or Britain. To evaluate any benefit in terms of call cost savings a comparison was carried out between the costs paid to Eircom for the voice calls and the costs that would have been paid to two VoIP service providers. Eircom rate structures are different from most VoIP Service providers, but by comparing the Eircom rates with costs for the same amount of minutes for carrying the calls with VoIP providers the costs should be lower if any savings are to be made. In this project two service providers have been evaluated that would be considered because of their support for both IAX and SIP protocols. The rates are similar but Blueface are a more established name with a larger customer base compared with IAX.ie whose website states that this company is just recently established since December 2005. Note the minimum call charges on Eircom's network are 5.4c/minute and with Blueface and IAX.ie it depends on the destination. IAX is lowest on local National and UK rates but more expensive when calling the GSM network. Here is a table showing the rates for July 2006.

	Eircom	**IAX.IE**	**Blueface**
Local	Day 4.07c Evening 1.04c	1.7c +min charge .85c	2c +min charge 1.9c
National	Day 6.8c Evening 4.1c W/end 1.04	1.7c +min charge .85c	2c +min charge 1.9c
Britain	Day 12.7c Evening 11.9c W/end 10.3c	1.78c +min charge .85c	2c +min charge 1.6c
Mobile Calls*	Vodaphone 19.1 O2 19.1 Meteor 22.67	22c +min charge 11c	2c +min charge.21c

Table 3 - Eircom Rates and Rates of 2 VoIP Service Providers

(*Vodaphone,O2 & Meteor are Mobile service providers in the Irish Market)

The costs for each of the VoIP providers were compared in a table with the Eircom cost for six months. It also must be pointed out that the Eircom rates shown are for individual calls and savings are applied as call spend increases and depending on the package agreed with Eircom.

	Call Type	Eircom Costs	Blueface Costs	IAX.ie	Best Cost
1	INTERNATIONAL MOBILES	€18.80	€10.96	€14.96	€10.96
2	NATIONAL CALL 0818	€6.00	€4.56	€7.92	€4.56
3	LOCALL 1890	€21.74	€25.90	€28.00	€21.74
4	LOCAL	€74.59	€33.42	€28.55	€28.55
5	LOCAL&NAT Min Talktime	469.55	€104.19	€75.28	€75.28
6	INTERNATIONAL	€119.30	€15.36	€82.42	€15.36
7	INLAND	€478.13	€202.84	€150.65	€150.65
8	FIXED2MOBILE Talktime	€481.41	€548.96	€961.84	€481.41
9	CROSS CHANNEL	€685.66	€81.20	€65.84	€65.84
10	CONDUIT 11850	€15.08	€13.02	€13.02	€2.06
11	CALLSAVE 1850	€5.33	€16.25	€10.32	€5.33
12	11811 Dire enq	€80.49	€97.71	€97.71	€80.49
13	087 MOBILE	€17.48	€13.72	€14.52	€13.72
14	086 MOBILE	€18.63	€15.51	€14.41	€14.41
15	085 MOBILE	€15.27	€2.44	€18.04	€2.44
		€2,507.48	€1,186.03	€1,583.49	€972.82

Table 4 – Cost comparison table for Eircom vs. Blueface and IAX.ie

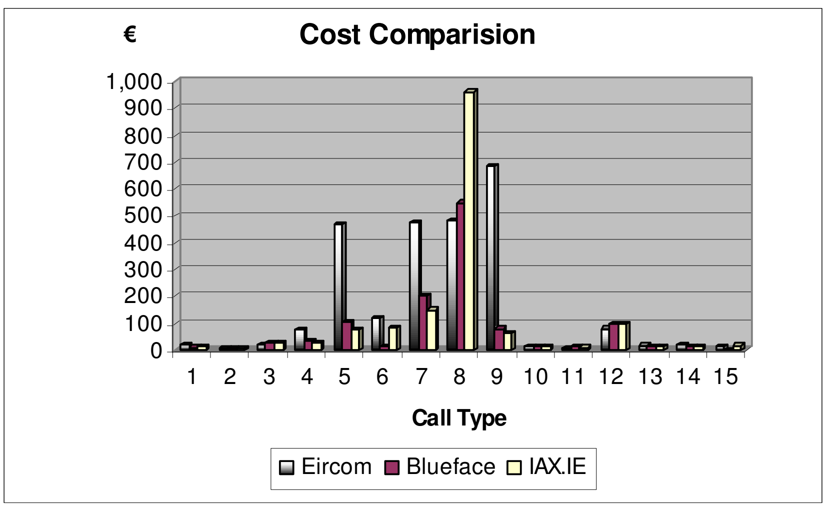

Figure1 – Comparison of using 2 VoIP providers against Eircom.

This shows that there are savings to be made on certain call types. For example call type 5 shows a very high cost for Eircom (€470) compared with IAX.ie at €75.28, but for call type 8 Eircom seems to be much better on the surface. Further exploration reveals that these rates are only available from Eircom when the bill is at certain levels so the more that is spent the higher the savings that can be applied to certain calls. So to get the best savings possible from all three of the providers, a mix of services would need to be used depending on the call type. Calls would still need to be carried with Eircom where their rates and costs were lower but on a 6 month period between two providers and this gives a saving of approx €1535.00 over six months. That would be great if it was the only cost but there is also possibly another cost. The costs would relate to the broadband connection that is being used now to carry these calls. To evaluate these costs it was necessary to carry out some further analysis on the existing telecom infrastructure to evaluate its VoIP capability.

The Existing Infrastructure

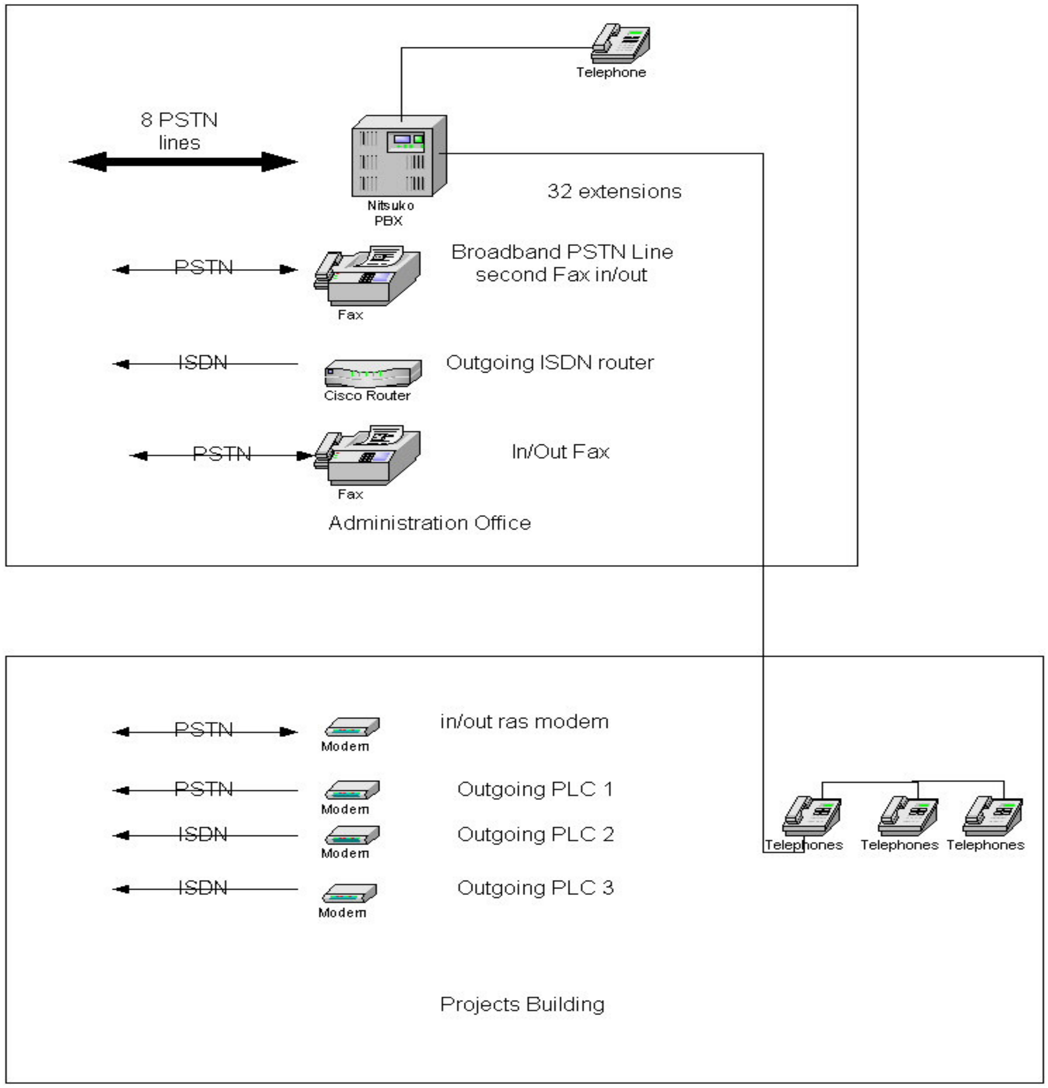

Figure 4 – Overview of the existing Telecomm layout

The system as it stands comprises of the following telecom equipment at the main office Site. A Nitsuko DXE PBX – this can have up to 40 extensions connected, but at present there are 32 stations with only 26 stations in use. The other equipment includes fax modems, data modems and routers. The layout is as shown in figure 4 above. It was discovered by earlier research into the system that there were no upgrade paths for this system to allow it to use VoIP technologies directly but there was a possibility of using an external adaptor to take advantage of VoIP. Spare parts could be sourced on the Internet or through some system refurbishers to provide a stock of parts for possible faults. It was also determined that the existing system is supported on a best efforts basis only and no service agreement was in place with any service provider therefore the support status was very hard to evaluate.

The existing data network in the company's office and the connections to the remote offices were examined. The network infrastructure in place consists of Cat5 cabling in the office buildings with 4 Core Fibre joining the buildings. Each existing phone point was adjacent to at least 1 network point. The remote users connected to the Office using ADSL through specially established IPsec VPN tunnels. This was important as it meant SIP phones could be used without having to worry about crossing the firewall and introducing problems with Network Address Translation. The connection to the Internet was identified as ADSL with 5M download and 512k upload on a 24:1 contention ratio. In the remote offices the connections were 3M download and 384kbs up with a 24:1 contention ratio. The costs associated with telephone calls from the remote offices were not fully analysed, instead an estimate was given by the remote users, as these bills were not paid by the company but paid by the remote users. From the remote users opinion the usage costs were approximately Euro 40.00 per month.

Gathering Requirements

From user surveys it was determined that the system had certain features that were considered necessary in any replacement. The most requested feature was speed dials, and most people wanted the following features, call forwarding, call transfer, caller id, speaker phone, call pickup, redial, paging and call parking. On the management side several things were required of any replacement system but at the top of the list was the ability to get call analysis reporting. Other requirements related to the support status of any replacement system and it was felt by the management that a much better support status for the phone system was required.

Other VOIP systems.

Several service providers were also contacted in relation to the services they offered and a quotation for a proprietary IP PBX system was requested to evaluate the costs associated with not going down the route of using Asterisk as the replacement. It was discovered in this research analysis that the current telecom structure would have to change and would have to move away from PSTN to an ISDN based system to support the systems offered by the two suppliers contacted. But this also offered advantages in terms of lowering monthly rental costs. This was because the cost of a Fractional Rate ISDN service from Eircom was 158.72 ex Vat and this could be used to replace the rental on 15 pstn lines and 2 isdn lines resulting in a saving of Euro 202.96 per month or Euro 2435 per year (ex VAT). A fractional rate line from Eircom is expensive to install with the current cost approx Euro 3,300.00 ex VAT. This change on its own would require the existing system to be replaced, as it is not ready for ISDN. The quotation received from one supplier was for a Siemens HiPath 3550 VoIP Telephone System. It is included in Appendix 2. The cost to purchase outright was Euro 10,950.00 ex VAT with an annual maintenance of Euro 985.00 ex VAT. So with these figures it would cost Euro 14,250 ex VAT for a new system, and there would be annual maintenance of 985.00 ex VAT after the first year. On rental savings alone it would take between 6-7 years to break even on the cost of the new system.

Design of VoIP Pilot System

There was enough information to start making decisions about the pilot system. From the users requirements it was decided what type of telephones to use. Lots of users requested the paging feature so a phone with this ability was chosen as most of the phones reviewed supported all the other features through Asterisk. So it was decided to test with two models the Aastra 480i and the Aastra 9133i (Voip-info.org 2006).

From the initial findings it was decided to use a fairly basic PC. It would not need huge resources, as it was only an evaluation rather than a production system. Several sources suggested that the minimum specifications required for any such system were (Sourceforge 2005)(VoIP-info.org 2005) at least a Pentium 133Mhz. Fortunately the company had some spares which were a little better and the spec that was chosen was a Pentium III 400Mhz with 384 MB Ram as this according to the VoIP-info source would be capable of handling several concurrent calls. The operating system chosen was Suse Linux 9.1, mainly because it had been used on other Linux projects and there was an established familiarity with this distribution of Linux. A review of the voip-info wiki on asterisk OS platforms stated that asterisk was known to work on the Linux 2.6 Kernel with Suse 9.1 (Digium 2006). It was also decided to purchase a low cost PSTN FXO card for integrating with the PSTN network. The card chosen was the XP100p. Integration with the PSTN network would be needed for some incoming calls and for costs savings as shown for some outgoing calls. As the system would also be used for remote users it was decided at this stage to set up one phone in the main office and one at a remote office for testing call quality and system features.

Some research was carried out into the type of protocol to use for VoIP. Asterisk itself supports a large range of protocols including

- IAX™ (Inter-Asterisk Exchange)
- H.323
- SIP (Session Initiation Protocol)
- MGCP (Media Gateway Control Protocol)
- SCCP (Cisco® Skinny

For the pilot system the test phones were going to be using SIP as there was no issue with firewalls because the remote users were on a secure IPsec tunnel and the local users were on the same LAN. For testing the server from two providers it was decided to use the IAX protocol. The simple reason here was we would not have to worry about any problems that typically would be encountered when using SIP across a Firewall that employs NAT. In this pilot system it was not possible to provide the VoIP server with a public IP address as suggested to get around this problem (VoIP-info.org 2006).

Because some of the literature reviewed suggested that quality-of-service (QoS) may be an issue the possibility of using QoS on the firewall (Watchguard Firebox III) was investigated. It was found that it doesn't currently support it (Watchguard 2006). Also several bandwidth tests were carried out to measure what is know as the Mean Opinion Score (MOS) score for the current connections in the remote office and the main office. The results from TestyourVoIP.com were as follows (results are in Appendix 3)

Remote Office MOS   4.4

Main Office MOS   4.3

Other objective testing was carried out using a network performance analysis tool called Iperf. This was setup to simulate various call scenarios that could be possible. These results are also in the appendix and show that several VoIP calls are possible but when more than two are active the quality degrades.

Another test was carried out where a VoIP test Server was set-up on a PC in a remote Office and the Main office then ran a test to check its VoIP Quality. This software was available on a trial basis but could be used to implement continuous monitoring of quality levels if necessary.

The test results available in the Appendix 3 serve to show there is ample bandwidth for a Voip call to or from the remote office or to a customer.

The test system was to be connected to the Office LAN. This would allow the current cabling in the office to be evaluated as if a system were to be deployed the existing cabling would probably be used for the phone system and new CAT6 cabling would be run for the PC network. It also meant that the pilot system could be easily tested without creating a separate network and having to re-configure the VPN endpoints of the remote office to allow testing. Note this would have to be considered in an overall deployment, as there would be costs associated with new cabling and router/firewall configuration changes for this separate phone network.

Pilot system Evaluation

After the successful implementation of the pilot Asterisk system an evaluation of its potential was carried out which will provide management and technical staff with information in relation to the requirements for a full deployment of such a system. A full presentation of the results were given to management. This included a demonstration of the test system and several of the deliverables from the project.

Resource Requirements

The project required the following resources to be acquired

- 2 Sip Hardphones
- 1 PC Pentium III 400Mhz 384Ram
- 1 XP100p – Card for Connection to PSTN

- SIP Service provider credit – some funds to make calls using a service provider

Review of Deliverables

The Deliverables from this project are listed below

- Analysis of current system
- Cost benefit analysis of a VoIP system
- Pilot system
- Quality of service tests results
- End User requirements survey
- Management requirements
- Pilot system evaluation report
- Supplier VoIP system quotation
- Presentation of findings to management

Outcomes

Current System Analysis

Of the deliverables some were achieved without looking at VoIP or any new technology such as the analysis of the current system. This yielded some interesting facts about the present Voice and Data needs

- The current PBX is unsupported
- It cannot natively use VoIP technologies but some vendors claim to have a unit capable of connecting it to VoIP services.
- A new system would require changing the hardware and possibly the phone lines

- 33% of calls are voice and 67% Data.

Pilot System

The design of the pilot system was decided upon after a review of the relevant literature as discussed in chapter two. Suse Linux was installed on a Compaq Deskpro Ex Pentium III 400Mhz with 384 Ram and 20GB Hard Disk. The default installation choices were taken. After the Linux installation certain Asterisk pre-requisites were satisfied. Then the Asterisk packages were installed. These packages were libpri, zaptel and asterisk. The procedure followed for this is available from Asteriskguru.com (Digium 2006). The install went okay and that evening several calls were made to the demonstration server at Digium in the US.

The next stage was to integrate with some hardware. The SIP phones and the XP100p card arrived and so it was decided to go about installing the XP100p card first. This proved to be a very difficult task. In the end the problem was down to the card sharing interrupts with other devices in the PC. And it was not possible to get it to work on its own, several other PC's were tested and eventually it was suspected that it may be something wrong with the card as it had been bought off EBay. After over two weeks of trial and effort with several PC's and different Linux distributions it was decided not to waste any further time and a proper Digium card a TDM400p was ordered. This had 1 FXO analogue model to connect with a PSTN Line. By this time it was also decided to move on to a Linux distribution called Slackware. This version was recommended by several training videos that were downloaded from the asterikast podcast site (Asterikast 2006). This web site provides a good introduction to getting started with asterisk and goes through the installation on Slackware Linux. After the new card was received some attempts were made to install it in the original PC but there were also problems installing with this. Eventually after trying several different PC's including a Dell Optiplex GX1 which also had the resource sharing issue. The card was got working on a less well known PC with a gigabyte Motherboard (GA-8SIMLH). This was important as this card would be used for testing the PSTN interface. Then the hardphones and some softphones were configured. The hardphones chosen were connected through the network and used DHCP to acquire their initial IP-Address. Once the phones were on the network they could be configured using a web page. The web page allowed configuration to point the phones at the Asterisk server. Once the phones were configured the asterisk server's configuration files were changed to setup the new phones in the system. The system's configuration files are included in the Appendix 4. Voicemail was also set-up for 4 users. A deskphone was delivered to one of the remote offices and the end users were given some brief instructions as to what testing was to be carried out. This involved several calls between the main office and the remote office. In the dialplan that was created the system was set-up so that the following occurred when a certain number was dialled first.

Number Dialled	**Action Performed**
2XXXXXXX	Dialled XXXXXX through IAX.ie
7XXXXXXX	Dialled XXXXXX through Blueface
9XXXXXXX	Dialled XXXXXX through a ZAP/PSTN channel
Not 2XX,9XX or 7XX but other combination XXX	XXX extension Dialled

Table 5 – Dial plan basics showing some of the possible dial sequences

When all this was up and running several simultaneous calls were made on the system to try and test it. Some bandwidth measurements were also recorded during these calls and these can be seen in appendix 5. The testing showed that it was possible to make a certain amount of calls but that as the number of calls increased above two the call quality started to degrade. This testing was done using two codec's ulaw and gsm. The ulaw codec produced better quality. But because of the lower bandwidth requirements for gsm it was tested to see what it was like for quality. The results are shown in Appendix 6. The quality of the call was definitely lower with the gsm codec, there was a definite increase in background noise or a background hiss on the call so the larger codec ulaw had better quality as expected. Calls were also carried out using the Digium TDM400p FXO card or Zap channel as its known in asterisk. This meant that asterisk could be used as a replacement if the company wanted to keep some of the existing analogue lines. Some experimentation was carried out on incoming IAX calls and customisation of the dial plan to ring various extensions in sequence and eventually drop to voicemail. The Voicemail system was tested also. When a user got a voice mail the system emailed the user as was configured in the voicemail.conf and included the message as a wav file. Some speed dials were set-up in the extensions.conf to and these were tested ok. Two softphones were evaluated and these were XLITE (Counterpath 2006) and Firefly (FreshTel 2006). Firefly was unusual in that it supported the IAX protocol but both were connected via SIP to the Asterisk Pilot system. Of the two tested XLITE was found to be the best mainly because of its features. The quality with the softphone was quite good and had the added advantage that they could easily be deployed without the same high costs of desktop phones.

Chapter 4 Project History

Here a brief overview of the project is presented. It should give the reader a good understanding of the overall project and help them understand what VoIP can offer XYSystems or any small company in Ireland.

How the project began

This project came about initially as part of a review of current systems in XYSystems. It was raised as an issue as to what would happen if there was to a failure on some part of the telephone system. The question was asked, what would be the support status? At the same time some of the telephone costs were raising concerns in the accounts department as to the high costs for the telecommunications needs. As a cost saving mechanism for the remote users some people had started to use Skype to call the remote staff. So the additional question was raised, what, if any potential did VoIP technologies offer for the company. Because of the lack of familiarity with VoIP there was a little reluctance at first to take on the project as there was already a busy workload. It was then decided that it would be a good project for the masters' thesis professional project. From there it was developed slightly to include a pilot implementation of the Asterisk PBX which offers VoIP and PSTN, ISDN capabilities.

How the project was managed

The project was managed mainly within the company. Some assistance was obtained from the project advisor to review the findings and initially help with some advice on the test system. When the project advisor agreed to take on the role an initial meeting was carried out and a demonstration and test on an existing asterisk system was carried out on the university campus in Galway. A project plan was drawn up after this meeting and this included dates for the deliverables in the project. The planned dates were as shown in Table 1 in Chapter 3. The plan was decided on and started out okay with the first few deliverables occurring on schedule. Unfortunately it was a bit ambitious and it had to be revised because of time constraints. The revised project plan is shown below and meant that the project was not entirely completed at the time of writing this report. It was further complicated by difficulties with the hardware chosen initially not working as desired.

Date	**Task to be completed**	**Duration**	**Finish Date**
01/09/05	Internet Research	5 months	01/02/06
20-Feb-06	Analysis of current system	1 week	27-Feb-2006
27-Feb-2006	Requirements gathering	2 days	01-Mar-2006
02-Mar-2006	System Design	2 days	03-Mar-2006
06-Mar-2006	Supplier Research	3 day	08-Mar-2006
20-Mar-2006	Installation of Linux	2 days	21-Mar-2006
24-Mar-2006	Install and config of asterisk	2 days	25-Mar-2006

10-July-2006	Hardware install	2 days	12- Jul -2006
12- Jul -2006	Telephone Configuration	2 days	14- Apr -2006
17-Jul-2006	Asterisk Configuration	10 days	28-Jul-2006
2-Aug-2006	Asterisk Testing	2 weeks	12-Aug-2006
14-Aug-2006	Analysis of findings	5 Days	18-Aug-2006
23-Aug-2006	Presentation to management	1 Day	23-Aug-2006

Table 7 – The revised project plan.

As can be seen from the revised project plan some of the new dates were cut short because of time constraints. Originally it was hoped to test the full set of features described as must haves by the employees but only a smaller subset of these features were tested. It was agreed that the remaining features could be tested a later date as part of the overall company project on the telecom system but they would be excluded from the results of this project. These features untested were

- Paging
- Management Reporting Features

Significant events/milestones in the project

As the project was carried out some of the deliverables were met and this created milestones in the project, some of the significant milestones included the following

1. Analysis of current System - Completed 27-Feb-2006
2. Requirements gathering - Completed 1-Mar-2006
3. Pilot System – Completed 28-Jul-2006

4. Cost Benefit analysis for VoIP system – Completed 18-Aug-2006
5. Supplier Voip Quotation –Completed 28-Mar-2006
6. VoIP Quality test reports – Completed 12-Aug-2006
7. Codec bandwidth test results – Completed 12-Aug-2006
8. Presentation to management – Completed 23-Aug-2006

Changes to the project plan

As was pointed out some changes had to be made to the project plan. The main reasons for the changes related to not having enough time with pressures from home, work and study forcing the project back by two months. There were also some issues in relation to the hardware originally planned for the pilot system. The XP100p card that was ordered from EBay never worked and meant an extra week and a half delay to an already delayed project. Because of these changes as was mentioned earlier some of the features of the pilot system were not fully evaluated including paging, management reporting.

Chapter 5 Project Results

Analysis of Results

The testing has shown that it is possible to use Asterisk for VoIP but bandwidth is an issue as this Internet connection is also heavily used for other purposes from time to time. From the research and tests carried out with the pilot system it was discovered that there could be seen that the existing broadband connection could support around 2 simultaneous calls using VoIP. But with an office staff of possibly 18 on a full day that would not be enough to handle the voice traffic. The savings in call costs made by moving some calls to a VoIP system would be around 1535.00 over a six-month period. But to achieve these savings some other costs needed to be factored in.

The cost of a better Internet connection.

At present there is a MAN or Metropolitan Area Network installed close to the companies premises. A connection to this network may provide a guaranteed bandwidth that would ensure a connection of higher quality and a guarenteed bandwidth for VoIP calls. One of the problems with ADSL is that it is shared among users. The companies bandwidth is 5M download and 512Kbps up, but there is a contention ratio to apply to this as well. ADSL services are 'contended' or shared among subscribers at a ratio given by your service provider, in this case it is a 24:1 ratio. The actual contention ratio being experienced depends on the number of active users on the service, and the bandwidth available to the network connection servicing them. So if all 24 users that share the bandwidth were on at the same time then the max each would receive could be 512/24 = 21kbs which is very low for a business VoIP connection. A provisional quotation for connecting to this MAN was received from Smart Telecom. The cost for connecting it would include a fractional Rate ISDN connection. So this would result in a line rental saving of 202.96 Euro per month. But the cost of the connection itself is very expensive at 8000.00 Euro. It would mean the company could get rid of the existing broadband connection as well bringing the monthly savings up to 371.96 Euro. When the savings on call rates are added into this the monthly savings become 570 Euro. But unfortunately the monthly rental on such a connection is 500.00 so there is no benefit in going down this road at the moment. An alternative supplier was also contacted that offered a better broadband package than Eircom. This was Irish Broadband – They are offering a 4m up 4m down package for 250 Euro per month. The problem is this is not available just yet but it may be worth looking into. They can guarantee the contention ratio of 8:1 giving a minimum bandwidth at all times of 512kb compared to Eircom's 21kb.

The costs of hardware and software for an Asterisk system.

This depends a lot on the level of integration with any existing telephone services but if the company were to use ISDN lines instead of PSTN they could save money on rental because a Basic Rate ISDN service can be used for two outgoing/incoming calls and the cost of this is 30.99

compared with 39.96 for two analogue lines. So the company could have 3 isdn BRI channels from Eircom connected to an Asterisk PBX. The hardware costs for such a system would be as follows

PC with asterisk installed		€1000.00
Quad ISDN Card		€550.00
22 X Aastra 9133i	€120.00 X 22	€2640.00
4 X Aastra 480i	€160.00 X 4	€640.00
26 X POE adaptors	€28.00 X 26	€768.00
2 X 16 Port switches	€145.00 X 2	€290.00
2 boxes of Cat6 cabling	€70 X 2	€140.00
2 X24 Port Cat6 patch	€38.00 X 2	€76.00
32 network points	€6.60 X 32	€212.00
32 1/2meter flyleads	€4.00 X 32	€128.00
QoS Router/Firewall	€500.00	€500.00
Total		€6944.00

There may also be some benefit in fitting a GSM card to this to carry some of the mobile calls as for six months the duration of these calls was approx 100 Hours. Considering the calls to mobiles cost was 530.00 over the six-month period this may be worth investigating further if a new system were to be implemented.

The costs in terms of staff time in setting up the system and user training.

To evaluate these costs a work schedule was drawn up and it consisted of the following tasks

- New cabling infrastructure – 1 man 5 days. This consists of pulling in the cat6 cabling and terminating it in patch panels and at wall boxes at each workstation.
- Asterisk install and config – 1 man 10 days. This consists of installing the operating system configuring Asterisk with the hardware. It involves setting up the dialplan and configuring the system and the phones.
- User training and documentation - 3 days. This would involve group-training sessions at the main office in small groups to demonstrate the features and 1 on 1 training where necessary.

The cost of a better Internet connection capable of sustaining more voice calls. – Possibly €250.00 per month

The cost of hardware and software required to run any new system Euro €6944.00

The costs in terms of staff time in setting up the system and user training. One off staff time cost of approx 18 man-days.

But by moving to this system there would be a monthly rental saving of €64.00 and a monthly call savings of around €250.00.

On a pure cost basis this would not seem like a good decision to invest in a new system that would cost nearly €7000.00 to save just over €64.00 a month. But the cost would have to be weighed up against the benefits received.

- ❖ The main benefits would be to the staff with increased features at their disposal. These new features such as voicemail, call forwarding would make the employees more productive.
- ❖ Costs of calls to and from remote staff and to and from customers could be lowered substantially.
- ❖ The system would then be under the support of local IT staff which may reduce costs in terms of long-term support.
- ❖ Although not tested it is assumed that management would be able to get usage reporting information.
- ❖ The removal of a significant business risk would also be a big reason for installing a new system.

Evaluation of whether or not the project met project goals

The first goal of the project was to evaluate the existing systems to see if there were any cost savings to be made for the company by moving some of its voice traffic from the POTS across to utilise VoIP technologies. The analysis of the phone records for the six months period show that there are cost savings to be made. What it doesn't show is whether the existing Internet connection is capable of carrying the voice traffic at acceptable levels of quality. From testing the system it has been shown that as the number of calls increases so too does the bandwidth usage and the quality suffers- Packets are lost or discarded. From some of the testing that was carried out 2 simultaneous calls were fine but any more created problems.This was evident from listening to the call quality and from the testing with iperf . So if the bandwidth were to potentially increase then there would be additional capacity for carrying more VoIP calls without quality issues. But increasing the bandwidth comes at a cost and this cost outweighs any cost savings benefit arrived at from moving the traffic to VoIP.

The second goal was to evaluate Asterisk as a possible replacement for the current PBX. This was achieved and it was agreed that Asterisk is more than up to the task. It has many features that were not tested but from what was tested it performed as it was claimed from research about its capabilities. It also offered the company the ability to take more control of its own telecom system in order to customize it.

The other goal of the project was an evaluation of the current telecom infrastructure. This was achieved and now the company has a better understanding of what they have in terms of hardware, its support and maintenance status, and its capabilities in terms of VoIP. Management now also have some more information about the call costs and based on the findings several areas that will need further investigation including the Data line usage. In relation to the costs associated to the data traffic a carrier has been found who will carry some of these calls at lower rates than Eircom, which should offer some savings even if we don't change to VoIP.

Discussion of what went right and what went wrong in the project

As with all projects some things will go according to plan but others will not. In this project 2 major things went wrong. Firstly the project did not follow the original schedule. The combined demands of home, work and study caused unforeseen pressures, which lead to major delays in getting certain tasks started. It was unfortunate because when things did get underway several parts of the project had to be downsized.

The other major thing that went wrong was a bad choice of hardware for the test system. The evaluation that lead to deciding on the hardware to use was making the assumption that cheaper hardware would save money on the project. But eventually it caused a loss of time. It may be no harm as in the long run as it meant several different Linux distributions had been tried and valuable experiences were gathered along the way.

Finding Analysis/results

This project has found that it is possible to use VoIP to save money on calls over Eircom rates. It also found that as the number of calls increases the quality degrades. It found that the Eircom Enhanced package can be used for two simultaneous calls without major problems, but after two calls the quality get worse and worse. This degradation because of bandwidth issues and lack of QoS features on the

Firewall/router meant the company would not be able to use the existing Broadband connection.

It was also found that the current system is only supported on a best efforts basis. This means if it goes down the company doesn't have any guarantees of when it will be repaired. It was also identified that 30% of the usage is voice and the remainder data.

The pilot system showed some of the potential of the open source software PBX Asterisk. It identified a possible replacement with this solution and the costs involved. The findings also show that a Siemens HiPath 3550 VoIP capable system would cost nearly 3 times as much and would require moving to Fractional Rate ISDN.

Conclusions

What was learned from the project

The project has taught me several things. The first is that things do not always go as planned. The initial project plan had to be changed but even so the project did provide some answers to the questions asked. It also gave a very good insight into the world of VoIP and the uses of Asterisk. It is felt that even though the current infrastructure at XYSystems is not ready for VoIP it will be very soon and the company have had a very good lesson in the pitfalls and merits ahead of the rush. Some unknowns have been removed from the picture and they now know that their current system is really not well supported and poses a significant business risk. It has been learned that the different bandwidths are used by different codec's. It was also noticed that the different codec's used had a difference in quality. Going forward if the company were to implement a VoIP system, they will have learned that it is important to find a router/firewall that had support for the QoS. From looking at softphones it was learned they could be used for quick deployment without too much cost, but users preferred to have a hard phone on their desk.

What I would have done differently if I had to do the project again

The pilot system would have been started earlier. Also the life-cycle model might be changed to one that allowed several stages of testing. This would allow for the set-up of the system to carry out some tests, make some changes and then test again. The other things that would be changed are to do with the specifics of the research and the scope of the project. The scope would be reduced to exclude looking at other systems at all and concentrate on Asterisk and some of its more advanced features such as the Flash Operator panel & database integration. Some further testing with ISDN would also be carried out.

Summary of the project

The project was carried out to evaluate any possible benefits or cost savings for the company from VoIP technologies and to carry out an evaluation of the current systems. The costs for the current system were analysed and the costs for a similar amount of calls over a VoIP system were calculated. It was found that call savings could be made by using VoIP technologies, but that in order to be able to get these savings the bandwidth needed to increase. It was also discovered that in order to ensure bandwidth was assigned the company would need a router/firewall capable of supporting QoS. With Asterisk some hardphones and softphones were tested both locally and at a remote office and the quality was found quite good for 1- 2 calls. Some calls were made and received using a PSTN card installed in the Asterisk Server. Some of the features of asterisk were evaluated including call forwarding, voicemail, call parking, automated attendant, speed dials.

Did the project meet expectations?

The project expectations were not entirely met. Some of the features of the pilot system were not evaluated and this would require further work to complete this. For the work that was carried out the system performed very well and even better than expected. As for some of the other expectations in relation to VoIP the suspicion was there that it could be used to save money but the bottom line is the bandwidth is not there to support the necessary voice traffic, so you could say that this expectation was not met. The expectation that the project would be finished earlier was not lived up to and indeed toward the end it was a hard struggle to get completed. But the expectation that answers could be provided for the questions asked in this project was completed. The potentials offered to the company from VoIP were evaluated. The current system was evaluated and the company now have details from this. There was a good evaluation of Asterisk as a replacement PBX.

What would be the next stage if the project continued

Of course the two major items that were missed in the testing of the pilot system to be evaluated were – Management reporting and Paging. Also if there were no time constraints some more analysis would be carried out into the ISDN and GSM integration with asterisk and as mentioned earlier. The possibility of using asterisk for faxing would also be investigated.

Conclusions/Recommendations

After looking at Asterisk and seeing some of its potential it should be considered as a very serious contender for replacing the existing system. The company should start by using some softphones in the central office and one or two of the hardphones in key locations. For example one of the remote workers is a software developer and he spends a lot of time talking with the software development manager. A deskphone would be a great benefit to both of them and would allow some cost savings. Other staff members could be issued with a softphone and could be trained on using it for some of the calls to the UK or long duration national calls with the advice that if the quality is bad to switch to using the old phone and let the IT department know so they can track occurrences of poor quality.

When packages such as the one promised from Irish broadband materialise it would be worth testing it to check if it's okay for VoIP. They claim it is possible to run VoIP over their service but if its wireless there may be latency issues. The company should be constantly monitoring the available packages to see if there are any new deals that offer lower contention ratios and higher bandwidths.

In terms of the Data line usage the company should consider having these calls carried by a lower cost carrier to make some savings here.

Having reviewed all the findings and literature from the project I have come to the conclusion that the world of telephony is changing. No longer are we at the mercy of our Telecom supplier who traditionally could charge large figures for small changes and upgrades, VoIP will be something that will become more and more used in the future and eventually when more people are connected to broadband the PSTN line usage in some countries will decline. Using SIP has its limitations because of the problems posed by NAT. I think this may be a stumbling block that will cause problems in most Asterisk implementations but is surmountable by using IAX.

Works Cited

1. Wikipedia(2006) Voice over IP. Retrieved Jan, 2006, from http://en.wikipedia.org/wiki/VoIP
2. Swale, R. (2001). *Voice over IP : systems and solutions*. London: Institution of Electrical Engineers.
3. SkypeLimited. (2006). Skype - The whole world can talk for free. Retrieved Jan 06, from http://www.skype.com/
4. Instruments, T. (2000). Voice and Fax over Internet Protocol (V/FoIP). Retrieved Jan 2006, from http://www.iec.org/online/tutorials/vfoip/index.html
5. Bjørn Hestnes, Peter Brooks, Svein Heiestad, Trond Ulseth, Carl Aaby (2003). *Quality of Experience in real-time person to person communication* Paper presented at the International Symposium on Human Factors in Telecommunication Berlin.
6. Brownworth, A. (2006). Bi-directional Transfer Rate. Retrieved Jan, 2006, from http://www.bandwidth.com/media/voiptest_bidirectional.mp3
7. Melvin, B. F. a. H. (2004). An Evaluation and Simulation of VoIP Tested Conditions Resulting in Additional Mouth-to-Ear Delays: Dept. of Information Technology National University of Ireland.
8. Meggelen, J. V., Madsen, L., Smith, J., & Safari Tech Books Online (Online service). (2005). Asterisk the future of telephony.

9. Wen, H. (2006). From Analog to VoIP: Asterisk Brings Telephony Together Under One Open-Source Platform. Retrieved Jan, 2006, from http://osdir.com/Article8156.phtml
10. Charny, B. (2005). Is the telephone industry ready for open source? Retrieved Jan, 2006, from http://news.com.com/Is+the+telephone+industry+ready+for+open+source/2008-1082_3-5737703.html?part=rss&tag=5737703&subj=news
11. Digium. (2006). Asterisk® Features. Retrieved Jan, 2006, from http://www.asterisk.org/features
12. D. Gomillion, B. D. (2005). Building Telephony Systems With Asterisk. Retrieved Jan, 2006, from http://www.whichvoip.com/voip/articles/voip_asterisk.htm
13. Asteriskast. (2006). Asterikast #1 Telephony HOWTO Retrieved Jan, 2006, from http://www.asterikast.com/
14. Kentel Ltd. (2006). CONVERT YOUR PABX TO VOIP Retrieved Jan, 2006, from http://www.usedsystems.co.uk/voip/index.htm
15. Voip-info.org. (2006). Asterisk Paging and Intercom Retrieved Jan, 2006, from http://www.voip-info.org/wiki/view/Asterisk+Paging+and+Intercom
16. VoIP-info.org (2005) Asterisk dimensioning. Retrieved 01/12/05, from http://www.voip-info.org/wiki/view/Asterisk+dimensioning

17. Sorceforge.net (2005) Asterisk@Home Handbook - Hardware requirements. Retrieved 01/12/05, from http://asteriskathome.sourceforge.net/handbook/#Section_2.1

18. Digium. (2006). Installation on SuSE - NEW Retrieved Jan, 2006, from http://www.asteriskguru.com/tutorials/asterisk_installation_compilation_suse.html

19. Voip-info.org. (2006). NAT and VoIP. Retrieved Jan, 2006, from http://www.voip-info.org/wiki/view/NAT+and+VOIP

20. WatchguardForum. (2005). Firebox III 500 & QoS? . Retrieved Jan 2006, from https://www.watchguard.com/forum/default.asp?action=9&boardid=2&read=5007&fid=42

21. Counterpath. (2006). XLite Softphone. Retrieved July, 2006, from http://www.counterpath.com/

22. FreshTel (2006). Firefly Softphone. Retrieved July, 2006, from http://www.freshtel.net/

23. Pereira (2006). It's Time to Take VOIP Seriously Retrieved August 2006 from http://www.thechannelinsider.com/article/Its+Time+to+Take+VOIP+Seriously/177938_1.aspx

24. Richard J. B. Reynolds and Antony W. Rix. BT Technology Journal, 19 (2), 23-32, April 2001. Psytechnics reference number 2001-P01.

25. Ajay Tirumala , F. Q., Jon Dugan ,Jim Ferguson,Kevin Gibbs. (2003).

IPERF: NLANR.

Appendix 1 – Results of Users Survey

	Must Have	Nice to have	Not needed
Voicemail	37.50%	62.50%	0.00%
Voicemail Forwarding internal	12.50%	62.50%	25.00%
Voicemail Forwarding external	0.00%	75.00%	25.00%
DND Standard	37.50%	50.00%	12.50%
DND Custom	12.50%	50.00%	37.50%
Call Forwarding inter	75.00%	25.00%	0.00%
Call Forwarding external	50.00%	50.00%	0.00%
Call transfer inter	87.50%	12.50%	0.00%
Call transfer external	75.00%	25.00%	0.00%
Caller id internal	87.50%	12.50%	0.00%
Caller id external	75.00%	25.00%	0.00%
Speed Dials	100.00%	0.00%	0.00%
Call Parking	87.50%	12.50%	0.00%
External Voicemail Access	12.50%	62.50%	25.00%
Speaker Phone	87.50%	12.50%	0.00%
Message Waiting	62.50%	25.00%	12.50%
Headset Ability	87.50%	12.50%	0.00%
Paging	87.50%	12.50%	0.00%
Phone Directory	25.00%	75.00%	0.00%
On Hold Music	25.00%	62.50%	12.50%
Desktop Faxing	37.50%	50.00%	12.50%
Conference call	12.50%	62.50%	25.00%
Cordless	37.50%	37.50%	25.00%
Call Intrusion	25.00%	75.00%	0.00%
Call Pickup	25.00%	75.00%	0.00%
Redial	25.00%	75.00%	0.00%
Call register	25.00%	75.00%	0.00%
Phone Display	25.00%	75.00%	0.00%

Appendix 2 – Quotation from Supplier regarding a replacement system.

RE: Proposed New Telephone System

Dear Sir,

Further to our recent discussions regarding the supply of a new Telephone System, I now have pleasure in submitting my proposal with equipment and cost details for your consideration:

Conversation Piece are recommending the:

(-) *Siemens HiPath 3550 Digital VoiP Telephone System*

Therefore, the details are as follows:

Over/…

PROPOSAL

The company to supply and install:

1 Siemens HiPath 3550 Digital VoiP Telephone System

Initially equipped

To cater for:

1 x HiPath 3550 Digital VoiP Telephone System
6 x Basic Rate ISDN (12 Channels)
(Includes Lines,Prolinks,Beelines)
32 x Digital Extension positions
4 x Analogue Extension Positions
1 x 2 Port Voicemail (24 Mailboxes)
22 x OptiPoint 500 Standard 12 Key LCD H/Free Telephone
2 x OptiSet Memory 32 Key LCD H/Free Telephone
1 x Hi Path AM Light Call Accounting Package
2 x Gigaset Cordless Telephones
1 x Lightning Protection

VoiP Connection:

3 x Optipoint 410 Ecco Plus IP Telephones
1 x HG 1500 2 Channel VoIP Gateway Card
3 x IP Licence Works Points

All Connections need to be set up with Voice Tunnel and static IP Addresses

COST DETAILS

Option – Sale:
The company to supply, install and leave in good working order, all the above equipment at an outright sale figure of **€10,949.00.**

All our equipment is supplied with a full twelve months parts and labour warranty, subject to fair wear and tear, during normal working hours, thereafter an **Annual Maintenance Agreement** maybe entered into at a figure of **€985.00.**

Option – Rental:

The company to supply install and maintain under guarantee, all the above equipment, at an inclusive quarterly rental of **€998.00.** For a period of 5 years.

Our rental includes all service/maintenance and replacement of spare parts, due to normal fair wear and tear, during normal working hours, for the period of the agreement

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PLEASE NOTE:

Terms and Conditions

- *The above prices do not include Value Added Tax, which will, of course, be chargeable to you at the time of invoicing at the rate then ruling.*
- *Our quotation is valid for thirty days. No other conditions will apply, unless agreed in writing, by* ***Conversation Piece Ltd.***

I trust this is the information you require, however, should you require any further details regarding my proposal, please do not hesitate to contact me.

Assuring you of our best attention at all times.

Yours sincerely,
Conversation Piece Ltd.

Appendix 3 – VoIP quality testing results

Remote office1 - with 1 84 k stream of 160 byte packets

```
iperf -s -p 5002 -u -l160 -i6
------------------------------------------------------------
Server listening on UDP port 5002
Receiving 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1932] local 192.168.0.144 port 5002 connected with 192.168.0.139 port 2941
------------------------------------------------------------
Client connecting to 192.168.0.139, UDP port 5002
Sending 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1864] local 192.168.0.144 port 1676 connected with 192.168.0.139 port 5002
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932]  0.0- 6.0 sec  63.8 KBytes  87.0 Kbits/sec  5.471 ms 1380275029/  408 (3.4e+008%)
[1864]  0.0- 6.0 sec  61.4 KBytes  83.8 Kbits/sec
[1932]  6.0-12.0 sec  61.7 KBytes  84.3 Kbits/sec  0.606 ms    0/  395 (0%)
[1864]  6.0-12.0 sec  60.8 KBytes  83.0 Kbits/sec
[1932] 12.0-18.0 sec  61.6 KBytes  84.1 Kbits/sec  0.408 ms    0/  394 (0%)
[1864] 12.0-18.0 sec  62.0 KBytes  84.7 Kbits/sec
[1932] 18.0-24.0 sec  61.6 KBytes  84.1 Kbits/sec  0.822 ms    0/  394 (0%)
[1864] 18.0-24.0 sec  61.6 KBytes  84.1 Kbits/sec
[1932] 24.0-30.0 sec  61.6 KBytes  84.1 Kbits/sec  3.433 ms    0/  394 (0%)
[1864] 24.0-30.0 sec  61.9 KBytes  84.5 Kbits/sec
[1932] 30.0-36.0 sec  61.4 KBytes  83.8 Kbits/sec  1.953 ms    0/  393 (0%)
[1864] 30.0-36.0 sec  61.6 KBytes  84.1 Kbits/sec
[1932] 36.0-42.0 sec  61.4 KBytes  83.8 Kbits/sec  0.435 ms    0/  393 (0%)
[1864] 36.0-42.0 sec  61.6 KBytes  84.1 Kbits/sec
[1932] 42.0-48.0 sec  61.6 KBytes  84.1 Kbits/sec  5.567 ms    0/  394 (0%)
[1864] 42.0-48.0 sec  61.6 KBytes  84.1 Kbits/sec
[1932] 48.0-54.0 sec  61.7 KBytes  84.3 Kbits/sec  4.494 ms    0/  395 (0%)
[1864] 48.0-54.0 sec  61.4 KBytes  83.8 Kbits/sec
[1932]  0.0-59.8 sec   615 KBytes  84.3 Kbits/sec  1.406 ms    0/ 3939 (0%)
[1864] 54.0-60.0 sec  61.6 KBytes  84.1 Kbits/sec
[ ID] Interval       Transfer     Bandwidth
[1864]  0.0-60.1 sec   615 KBytes  83.9 Kbits/sec
[1864] Server Report:
[1864]  0.0-60.4 sec   471 KBytes  64.0 Kbits/sec  8.565 ms  922/ 3939 (23%)
[1864] Sent 3939 datagrams

iperf -s -u -l 160 -i1 -o c:\call.txt
------------------------------------------------------------
Server listening on UDP port 5001
Receiving 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1932] local 192.168.0.144 port 5001 connected with 192.168.85.2 port 1193
------------------------------------------------------------
Client connecting to 192.168.85.2, UDP port 5001
Sending 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1864] local 192.168.0.144 port 1625 connected with 192.168.85.2 port 5001
[ ID] Interval       Transfer     Bandwidth
[1864]  0.0- 1.0 sec  20.6 KBytes   169 Kbits/sec
[1932]  0.0- 1.0 sec  15.6 KBytes   128 Kbits/sec  8.899 ms 1179603536/  100 (1.2e+009%)
[1864]  1.0- 2.0 sec  20.5 KBytes   168 Kbits/sec
[1932]  1.0- 2.0 sec  16.7 KBytes   137 Kbits/sec  7.824 ms    0/  107 (0%)
[1864]  2.0- 3.0 sec  20.5 KBytes   168 Kbits/sec
[1932]  2.0- 3.0 sec  18.1 KBytes   148 Kbits/sec  6.672 ms    3/  119 (2.5%)
[1864]  3.0- 4.0 sec  20.6 KBytes   169 Kbits/sec
[1932]  3.0- 4.0 sec  23.1 KBytes   189 Kbits/sec  9.280 ms   11/  159 (6.9%)
[1864]  4.0- 5.0 sec  20.5 KBytes   168 Kbits/sec
[1932]  4.0- 5.0 sec  22.2 KBytes   182 Kbits/sec  9.337 ms    0/  142 (0%)
[1864]  5.0- 6.0 sec  20.5 KBytes   168 Kbits/sec
```

```
[1932]  5.0- 6.0 sec  21.6 KBytes   177 Kbits/sec  6.477 ms    0/  138 (0%)
[1932]  6.0- 7.0 sec  20.3 KBytes   166 Kbits/sec  13.764 ms    0/  130 (0%)
[1864]  6.0- 7.0 sec  20.5 KBytes   168 Kbits/sec
[1864]  7.0- 8.0 sec  20.6 KBytes   169 Kbits/sec
[1932]  7.0- 8.0 sec  23.1 KBytes   189 Kbits/sec  7.071 ms    0/  148 (0%)
[1864]  8.0- 9.0 sec  20.5 KBytes   168 Kbits/sec
[1932]  8.0- 9.0 sec  21.7 KBytes   178 Kbits/sec  7.710 ms    0/  139 (0%)
[1864]  9.0-10.0 sec  20.5 KBytes   168 Kbits/sec
[1932]  9.0-10.0 sec  19.1 KBytes   156 Kbits/sec  11.475 ms    0/  122 (0%)
[ ID] Interval       Transfer     Bandwidth
[1864] 10.0-11.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 10.0-11.0 sec  20.2 KBytes   165 Kbits/sec  7.820 ms    0/  129 (0%)
[1864] 11.0-12.0 sec  20.0 KBytes   164 Kbits/sec
[1932] 11.0-12.0 sec  17.8 KBytes   146 Kbits/sec  7.077 ms    0/  114 (0%)
[1864] 12.0-13.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 12.0-13.0 sec  22.3 KBytes   183 Kbits/sec  9.373 ms    0/  143 (0%)
[1932] 13.0-14.0 sec  16.9 KBytes   138 Kbits/sec  12.499 ms    0/  108 (0%)
[1864] 13.0-14.0 sec  19.8 KBytes   163 Kbits/sec
[1932] 14.0-15.0 sec  20.9 KBytes   172 Kbits/sec  6.911 ms    0/  134 (0%)
[1864] 14.0-15.0 sec  20.8 KBytes   170 Kbits/sec
[1932] 15.0-16.0 sec  19.4 KBytes   159 Kbits/sec  5.854 ms    0/  124 (0%)
[1864] 15.0-16.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 16.0-17.0 sec  19.4 KBytes   159 Kbits/sec  9.779 ms    0/  124 (0%)
[1864] 16.0-17.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 17.0-18.0 sec  20.9 KBytes   172 Kbits/sec  11.062 ms    0/  134 (0%)
[1864] 17.0-18.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 18.0-19.0 sec  22.8 KBytes   187 Kbits/sec  9.925 ms    0/  146 (0%)
[1864] 18.0-19.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 19.0-20.0 sec  19.5 KBytes   160 Kbits/sec  11.562 ms    0/  125 (0%)
[1864] 19.0-20.0 sec  20.5 KBytes   168 Kbits/sec
[ ID] Interval       Transfer     Bandwidth
[1864] 20.0-21.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 20.0-21.0 sec  22.8 KBytes   187 Kbits/sec  9.193 ms    0/  146 (0%)
[1864] 21.0-22.0 sec  20.8 KBytes   170 Kbits/sec
[1932] 21.0-22.0 sec  16.6 KBytes   136 Kbits/sec  9.490 ms    0/  106 (0%)
[1864] 22.0-23.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 22.0-23.0 sec  20.8 KBytes   170 Kbits/sec  8.888 ms    0/  133 (0%)
[1932] 23.0-24.0 sec  19.7 KBytes   161 Kbits/sec  11.883 ms    0/  126 (0%)
[1864] 23.0-24.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 24.0-25.0 sec  20.9 KBytes   172 Kbits/sec  6.639 ms    0/  134 (0%)
[1864] 24.0-25.0 sec  20.0 KBytes   164 Kbits/sec
[1932] 25.0-26.0 sec  22.2 KBytes   182 Kbits/sec  8.591 ms    0/  142 (0%)
[1864] 25.0-26.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 26.0-27.0 sec  20.9 KBytes   172 Kbits/sec  9.018 ms    0/  134 (0%)
[1864] 26.0-27.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 27.0-28.0 sec  19.2 KBytes   157 Kbits/sec  8.517 ms    0/  123 (0%)
[1864] 27.0-28.0 sec  20.5 KBytes   168 Kbits/sec
[1864] 28.0-29.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 28.0-29.0 sec  21.1 KBytes   173 Kbits/sec  7.850 ms    0/  135 (0%)
[1864] 29.0-30.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 29.0-30.0 sec  19.7 KBytes   161 Kbits/sec  7.856 ms    0/  126 (0%)
[ ID] Interval       Transfer     Bandwidth
[1864] 30.0-31.0 sec  19.8 KBytes   163 Kbits/sec
[1932] 30.0-31.0 sec  22.0 KBytes   180 Kbits/sec  5.233 ms    0/  141 (0%)
[1864] 31.0-32.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 31.0-32.0 sec  18.6 KBytes   152 Kbits/sec  12.541 ms    0/  119 (0%)
[1864] 32.0-33.0 sec  20.0 KBytes   164 Kbits/sec
[1932] 32.0-33.0 sec  21.9 KBytes   179 Kbits/sec  9.525 ms    0/  140 (0%)
[1864] 33.0-34.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 33.0-34.0 sec  19.7 KBytes   161 Kbits/sec  10.290 ms    0/  126 (0%)
[1864] 34.0-35.0 sec  19.8 KBytes   163 Kbits/sec
[1932] 34.0-35.0 sec  21.6 KBytes   177 Kbits/sec  9.081 ms    0/  138 (0%)
[1932] 35.0-36.0 sec  19.7 KBytes   161 Kbits/sec  9.649 ms    0/  126 (0%)
[1864] 35.0-36.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 36.0-37.0 sec  20.9 KBytes   172 Kbits/sec  7.878 ms    0/  134 (0%)
[1864] 36.0-37.0 sec  20.3 KBytes   166 Kbits/sec
[1864] 37.0-38.0 sec  20.2 KBytes   165 Kbits/sec
[1932] 37.0-38.0 sec  19.4 KBytes   159 Kbits/sec  7.652 ms    1/  125 (0.8%)
[1864] 38.0-39.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 38.0-39.0 sec  20.6 KBytes   169 Kbits/sec  12.791 ms    0/  132 (0%)
[1864] 39.0-40.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 39.0-40.0 sec  21.1 KBytes   173 Kbits/sec  7.380 ms    0/  135 (0%)
```

```
[ ID] Interval       Transfer     Bandwidth
[1864] 40.0-41.0 sec  20.0 KBytes   164 Kbits/sec
[1932] 40.0-41.0 sec  20.5 KBytes   168 Kbits/sec  6.773 ms    0/  131 (0%)
[1864] 41.0-42.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 41.0-42.0 sec  19.4 KBytes   159 Kbits/sec  7.884 ms    0/  124 (0%)
[1864] 42.0-43.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 42.0-43.0 sec  20.5 KBytes   168 Kbits/sec  5.073 ms    0/  131 (0%)
[1864] 43.0-44.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 43.0-44.0 sec  20.0 KBytes   164 Kbits/sec  9.036 ms    0/  128 (0%)
[1864] 44.0-45.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 44.0-45.0 sec  20.2 KBytes   165 Kbits/sec  9.243 ms    0/  129 (0%)
[1864] 45.0-46.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 45.0-46.0 sec  20.2 KBytes   165 Kbits/sec  7.809 ms    9/  138 (6.5%)
[1864] 46.0-47.0 sec  19.8 KBytes   163 Kbits/sec
[1932] 46.0-47.0 sec  18.6 KBytes   152 Kbits/sec  10.614 ms    2/  121 (1.7%)
[1864] 47.0-48.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 47.0-48.0 sec  20.6 KBytes   169 Kbits/sec  9.215 ms    0/  132 (0%)
[1932] 48.0-49.0 sec  20.2 KBytes   165 Kbits/sec  8.551 ms    5/  134 (3.7%)
[1864] 48.0-49.0 sec  20.6 KBytes   169 Kbits/sec
[1864] 49.0-50.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 49.0-50.0 sec  20.0 KBytes   164 Kbits/sec  10.234 ms    9/  137 (6.6%)
[ ID] Interval       Transfer     Bandwidth
[1864] 50.0-51.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 50.0-51.0 sec  19.2 KBytes   157 Kbits/sec  10.503 ms    2/  125 (1.6%)
[1864] 51.0-52.0 sec  21.1 KBytes   173 Kbits/sec
[1932] 51.0-52.0 sec  19.8 KBytes   163 Kbits/sec  8.199 ms    0/  127 (0%)
[1864] 52.0-53.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 52.0-53.0 sec  20.6 KBytes   169 Kbits/sec  8.428 ms    8/  140 (5.7%)
[1864] 53.0-54.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 53.0-54.0 sec  20.3 KBytes   166 Kbits/sec  8.527 ms    5/  135 (3.7%)
[1864] 54.0-55.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 54.0-55.0 sec  19.1 KBytes   156 Kbits/sec  10.118 ms    1/  123 (0.81%)
[1864] 55.0-56.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 55.0-56.0 sec  21.1 KBytes   173 Kbits/sec  7.933 ms    0/  135 (0%)
[1864] 56.0-57.0 sec  20.6 KBytes   169 Kbits/sec
[1932] 56.0-57.0 sec  22.2 KBytes   182 Kbits/sec  9.434 ms    0/  142 (0%)
[1864] 57.0-58.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 57.0-58.0 sec  18.8 KBytes   154 Kbits/sec  8.321 ms    0/  120 (0%)
[1864] 58.0-59.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 58.0-59.0 sec  21.4 KBytes   175 Kbits/sec  7.669 ms    0/  137 (0%)
[1864] 59.0-60.0 sec  20.5 KBytes   168 Kbits/sec
[1932] 59.0-60.0 sec  20.0 KBytes   164 Kbits/sec  9.019 ms    0/  128 (0%)
[ ID] Interval       Transfer     Bandwidth
[1864]  0.0-60.0 sec  1.20 MBytes   168 Kbits/sec
[1932]  0.0-60.4 sec  1.19 MBytes   166 Kbits/sec  5.194 ms   56/ 7877 (0.71%)
[1864] Server Report:
[1864]  0.0-60.2 sec   930 KBytes   127 Kbits/sec  8.158 ms 1920/ 7873 (24%)
[1864] Sent 7873 datagrams
```

Report carried out from remote office to main office simulating 2 calls. Notice how packet loss has gotten worse with the second call.

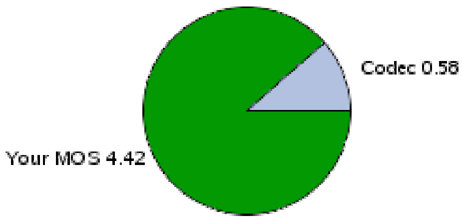
Here we have a simulation of 3 calls and we can clearly see an increase to lost datagrams.

```
iperf -s -u -l 160 -i1 -o
------------------------------------------------------------
Server listening on UDP port 5001
Receiving 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1932] local 192.168.0.144 port 5001 connected with 192.168.85.2 port 1190
------------------------------------------------------------
Client connecting to 192.168.85.2, UDP port 5001
Sending 160 byte datagrams
UDP buffer size: 8.00 KByte (default)
------------------------------------------------------------
[1864] local 192.168.0.144 port 1508 connected with 192.168.85.2 port 5001
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932]  0.0- 1.0 sec  16.4 KBytes   134 Kbits/sec  12.355 ms 1179603536/  105 (1.1e+009%)
[1864]  0.0- 1.0 sec  29.5 KBytes   242 Kbits/sec
[1932]  1.0- 2.0 sec  23.6 KBytes   193 Kbits/sec  10.864 ms   42/  193 (22%)
[1864]  1.0- 2.0 sec  29.4 KBytes   241 Kbits/sec
[1932]  2.0- 3.0 sec  22.7 KBytes   186 Kbits/sec  4.087 ms   40/  185 (22%)
[1864]  2.0- 3.0 sec  29.2 KBytes   239 Kbits/sec
[1932]  3.0- 4.0 sec  23.4 KBytes   192 Kbits/sec  7.818 ms   40/  190 (21%)
[1864]  3.0- 4.0 sec  29.4 KBytes   241 Kbits/sec
[1932]  4.0- 5.0 sec  23.0 KBytes   188 Kbits/sec  5.336 ms   38/  185 (21%)
[1864]  4.0- 5.0 sec  29.2 KBytes   239 Kbits/sec
[1932]  5.0- 6.0 sec  23.6 KBytes   193 Kbits/sec  12.701 ms   39/  190 (21%)
[1864]  5.0- 6.0 sec  29.4 KBytes   241 Kbits/sec
[1932]  6.0- 7.0 sec  22.3 KBytes   183 Kbits/sec  2.439 ms   42/  185 (23%)
[1864]  6.0- 7.0 sec  29.2 KBytes   239 Kbits/sec
[1932]  7.0- 8.0 sec  23.3 KBytes   191 Kbits/sec  6.170 ms   39/  188 (21%)
[1864]  7.0- 8.0 sec  29.4 KBytes   241 Kbits/sec
[1932]  8.0- 9.0 sec  23.0 KBytes   188 Kbits/sec  12.550 ms   42/  189 (22%)
[1864]  8.0- 9.0 sec  29.2 KBytes   239 Kbits/sec
[1932]  9.0-10.0 sec  23.0 KBytes   188 Kbits/sec  1.936 ms   40/  187 (21%)
[1864]  9.0-10.0 sec  29.4 KBytes   241 Kbits/sec
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932] 10.0-11.0 sec  23.3 KBytes   191 Kbits/sec  8.649 ms   40/  189 (21%)
[1864] 10.0-11.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 11.0-12.0 sec  22.5 KBytes   184 Kbits/sec  3.229 ms   41/  185 (22%)
[1864] 11.0-12.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 12.0-13.0 sec  23.0 KBytes   188 Kbits/sec  5.358 ms   41/  188 (22%)
[1864] 12.0-13.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 13.0-14.0 sec  22.8 KBytes   187 Kbits/sec  8.212 ms   44/  190 (23%)
[1864] 13.0-14.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 14.0-15.0 sec  22.8 KBytes   187 Kbits/sec  9.155 ms   40/  186 (22%)
[1864] 14.0-15.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 15.0-16.0 sec  23.6 KBytes   193 Kbits/sec  10.028 ms   38/  189 (20%)
[1864] 15.0-16.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 16.0-17.0 sec  22.8 KBytes   187 Kbits/sec  10.695 ms   40/  186 (22%)
[1864] 16.0-17.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 17.0-18.0 sec  23.0 KBytes   188 Kbits/sec  5.535 ms   39/  186 (21%)
[1864] 17.0-18.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 18.0-19.0 sec  23.0 KBytes   188 Kbits/sec  12.418 ms   42/  189 (22%)
[1864] 18.0-19.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 19.0-20.0 sec  22.7 KBytes   186 Kbits/sec  1.448 ms   41/  186 (22%)
[1864] 19.0-20.0 sec  29.4 KBytes   241 Kbits/sec
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932] 20.0-21.0 sec  23.6 KBytes   193 Kbits/sec  12.143 ms   38/  189 (20%)
[1864] 20.0-21.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 21.0-22.0 sec  23.0 KBytes   188 Kbits/sec  7.130 ms   42/  189 (22%)
[1864] 21.0-22.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 22.0-23.0 sec  23.0 KBytes   188 Kbits/sec  3.728 ms   37/  184 (20%)
[1864] 22.0-23.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 23.0-24.0 sec  22.8 KBytes   187 Kbits/sec  8.319 ms   44/  190 (23%)
[1864] 23.0-24.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 24.0-25.0 sec  23.0 KBytes   188 Kbits/sec  8.562 ms   40/  187 (21%)
[1864] 24.0-25.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 25.0-26.0 sec  23.4 KBytes   192 Kbits/sec  10.820 ms   38/  188 (20%)
[1864] 25.0-26.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 26.0-27.0 sec  23.0 KBytes   188 Kbits/sec  0.295 ms   39/  186 (21%)
[1864] 26.0-27.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 27.0-28.0 sec  23.3 KBytes   191 Kbits/sec  8.201 ms   41/  190 (22%)
```

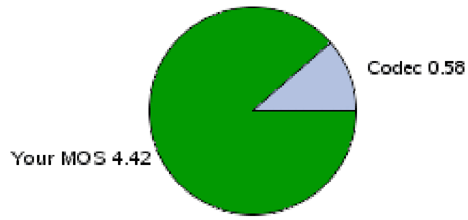
```
[1864] 27.0-28.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 28.0-29.0 sec  23.0 KBytes   188 Kbits/sec  4.279 ms   37/  184 (20%)
[1864] 28.0-29.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 29.0-30.0 sec  23.3 KBytes   191 Kbits/sec  9.740 ms   41/  190 (22%)
[1864] 29.0-30.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 30.0-31.0 sec  23.0 KBytes   188 Kbits/sec  7.419 ms   37/  184 (20%)
[1864] 30.0-31.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 31.0-32.0 sec  23.4 KBytes   192 Kbits/sec  9.975 ms   41/  191 (21%)
[1864] 31.0-32.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 32.0-33.0 sec  23.4 KBytes   192 Kbits/sec  7.489 ms   38/  188 (20%)
[1864] 32.0-33.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 33.0-34.0 sec  22.8 KBytes   187 Kbits/sec  2.786 ms   39/  185 (21%)
[1864] 33.0-34.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 34.0-35.0 sec  23.4 KBytes   192 Kbits/sec  9.385 ms   40/  190 (21%)
[1864] 34.0-35.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 35.0-36.0 sec  23.0 KBytes   188 Kbits/sec  1.848 ms   39/  186 (21%)
[1864] 35.0-36.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 36.0-37.0 sec  23.4 KBytes   192 Kbits/sec  13.322 ms   38/  188 (20%)
[1864] 36.0-37.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 37.0-38.0 sec  23.1 KBytes   189 Kbits/sec  5.539 ms   38/  186 (20%)
[1864] 37.0-38.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 38.0-39.0 sec  23.6 KBytes   193 Kbits/sec  11.213 ms   39/  190 (21%)
[1864] 38.0-39.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 39.0-40.0 sec  22.5 KBytes   184 Kbits/sec  3.900 ms   41/  185 (22%)
[1864] 39.0-40.0 sec  29.4 KBytes   241 Kbits/sec
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932] 40.0-41.0 sec  23.0 KBytes   188 Kbits/sec  1.519 ms   41/  188 (22%)
[1864] 40.0-41.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 41.0-42.0 sec  22.8 KBytes   187 Kbits/sec  10.147 ms   43/  189 (23%)
[1864] 41.0-42.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 42.0-43.0 sec  23.1 KBytes   189 Kbits/sec  5.134 ms   37/  185 (20%)
[1864] 42.0-43.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 43.0-44.0 sec  23.3 KBytes   191 Kbits/sec  5.129 ms   42/  191 (22%)
[1864] 43.0-44.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 44.0-45.0 sec  22.5 KBytes   184 Kbits/sec  6.246 ms   41/  185 (22%)
[1864] 44.0-45.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 45.0-46.0 sec  23.0 KBytes   188 Kbits/sec  5.522 ms   40/  187 (21%)
[1864] 45.0-46.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 46.0-47.0 sec  23.0 KBytes   188 Kbits/sec  6.568 ms   43/  190 (23%)
[1864] 46.0-47.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 47.0-48.0 sec  23.0 KBytes   188 Kbits/sec  5.995 ms   41/  188 (22%)
[1864] 47.0-48.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 48.0-49.0 sec  23.3 KBytes   191 Kbits/sec  8.683 ms   38/  187 (20%)
[1864] 48.0-49.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 49.0-50.0 sec  23.0 KBytes   188 Kbits/sec  1.471 ms   38/  185 (21%)
[1864] 49.0-50.0 sec  29.2 KBytes   239 Kbits/sec
[ ID] Interval       Transfer     Bandwidth       Jitter   Lost/Total Datagrams
[1932] 50.0-51.0 sec  23.6 KBytes   193 Kbits/sec  11.899 ms   39/  190 (21%)
[1864] 50.0-51.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 51.0-52.0 sec  22.5 KBytes   184 Kbits/sec  8.151 ms   41/  185 (22%)
[1864] 51.0-52.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 52.0-53.0 sec  22.8 KBytes   187 Kbits/sec  1.041 ms   42/  188 (22%)
[1864] 52.0-53.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 53.0-54.0 sec  23.4 KBytes   192 Kbits/sec  9.678 ms   40/  190 (21%)
[1864] 53.0-54.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 54.0-55.0 sec  22.8 KBytes   187 Kbits/sec  4.680 ms   39/  185 (21%)
[1864] 54.0-55.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 55.0-56.0 sec  23.4 KBytes   192 Kbits/sec  7.621 ms   40/  190 (21%)
[1864] 55.0-56.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 56.0-57.0 sec  23.0 KBytes   188 Kbits/sec  7.016 ms   38/  185 (21%)
[1864] 56.0-57.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 57.0-58.0 sec  23.4 KBytes   192 Kbits/sec  13.329 ms   39/  189 (21%)
[1864] 57.0-58.0 sec  29.2 KBytes   239 Kbits/sec
[1932] 58.0-59.0 sec  23.0 KBytes   188 Kbits/sec  2.660 ms   39/  186 (21%)
[1864] 58.0-59.0 sec  29.4 KBytes   241 Kbits/sec
[1932] 59.0-60.0 sec  23.3 KBytes   191 Kbits/sec  9.563 ms   40/  189 (21%)
[1864] 59.0-60.0 sec  29.2 KBytes   239 Kbits/sec
 [1864]  0.0-60.0 sec  1.72 MBytes   240 Kbits/sec
[1932]  0.0-60.4 sec  1.36 MBytes   188 Kbits/sec  11.163 ms 2372/11253 (21%)
[1864] Server Report:
[1864]  0.0-60.0 sec  1.72 MBytes   240 Kbits/sec  7.015 ms    0/11253 (0%)
[1864] Sent 11253 datagrams
```

VoIP Quality Test Results – Remote Office

MOS Analysis From You TO London



MOS Analysis FROM London To You



Media Quality

MOS	**4.4** / 5.0 (Best with G.711 is 4.4) **Degradation Sources** Codec 0.58 100.0% Latency 0.00 0.0% Packet Discards 0.00 0.0% Packet Loss 0.00 0.0%
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)
Round-Trip Latency	129 ms
Packet Discards	0.0%
Packet Loss	0.0%
Loss Periods	Min: 0 ms Avg: 0 ms Max: 0 ms *No Loss*
Jitter	Min: 0 ms Avg: 5 ms Max: 13 ms

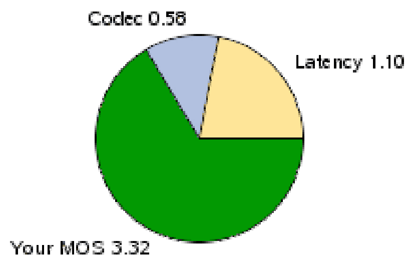
Media Quality

MOS	**4.4** / 5.0 (Best with G.711 is 4.4) **Degradation Sources** Codec 0.58 100.0% Latency 0.00 0.0% Packet Discards 0.00 0.0% Packet Loss 0.00 0.0%
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)
Round-Trip Latency	129 ms
Packet Discards	0.0%
Packet Loss	0.0%
Loss Periods	Min: 0 ms Avg: 0 ms Max: 0 ms *No Loss*
Jitter	Min: 4 ms Avg: 6 ms Max: 20 ms

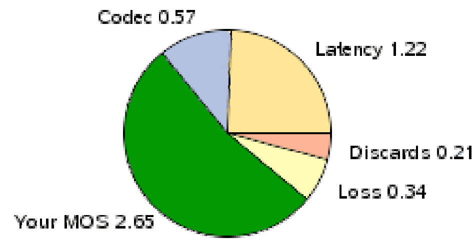
The test results above were generated from an online testing (testyourvoip .com) service which is useful to get a feel for the potential call quality. Several points can be derieved from the results . This test indicated that the remote office could sustain a single VoIP to London call with no problems.

VoIP Quality Test Results – Remote Office

MOS Analysis From You TO Sydney



MOS Analysis FROM Sydney To You



Media Quality

MOS	**3.3** / 5.0 (Best with G.711 is 4.4) **Degradation Sources** Codec 0.58 34.3% Latency 1.10 65.7% Packet Discards 0.00 0.0% Packet Loss 0.00 0.0%
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)
Round-Trip Latency	807 ms
Packet Discards	0.0%
Packet Loss	0.0%
Loss Periods	Min: 0 ms Avg: 0 ms Max: 0 ms *No Loss*
Jitter	Min: 0 ms Avg: 5 ms Max: 16 ms

Media Quality

MOS	**2.6** / 5.0 (Best with G.711 is 4.4) **Degradation Sources** Codec 0.57 24.4% Latency 1.22 52.1% Packet Discards 0.21 9.0% Packet Loss 0.34 14.4%
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)
Round-Trip Latency	807 ms
Packet Discards	0.7%
Packet Loss	1.1%
Loss Periods	Min: 20 ms Avg: 40 ms Max: 80 ms *Burst Loss*
Jitter	Min: 2 ms Avg: 10 ms Max: 27 ms

The above test indicates a problem with calling to Sydney. Some of the metrics like latecy are to be expected because of the distances involved. We can also see here that packets are lost and discarded. The MOS score is affected by latency most in this case but it is also influenced by packet loss and discards.

Appendix 4 – Asterisk Configuration files

4.1 – voicemail.conf

```
; Asterisk Test System XYSystems ltd
; Voicemail Configuration
[general]
format=wav49|gsm|wav
serveremail=asterisk
attach=yes
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3
emaildateformat=%A, %B %d, %Y at %r
sendvoicemail=yes       ; Context to Send voicemail from [option 5 from the advanced menu]
[zonemessages]
eastern=America/New_York|'vm-received' Q 'digits/at' IMp
central=America/Chicago|'vm-received' Q 'digits/at' IMp
central24=America/Chicago|'vm-received' q 'digits/at' H N 'hours'
military=Zulu|'vm-received' q 'digits/at' H N 'hours' 'phonetic/z_p'
[default]
171 =>1212,astra9133,reception@dg.net
151 =>1151,user1,user1@ XYSystems.ie
129 =>1129,user2,user2@ XYSystems.ie
147 =>1147,user3,user3@ XYSystems.ie
116 =>1116,user4,user4@ XYSystems.ie
```

4.1 – zapata.conf

```
; Asterisk Test System XYSystems
; Zapata.conf
[channels]
signalling=fxs_ks
loadzone=uk
defaultzone=uk
channel => 4
```

4.3 – extensinons.conf

```
; Asterisk Test System XYSystems
; Dialplan configuration - extensions.conf
[general]
autofallthrough=yes
[globals]
dialoutpstn=Zap/4
RECEPTION=astra480L1
ALL=astra480L1/astra9133
POTSPhone=iaxy1
speeddial40=***********      <=Numbers are removed here
speeddial41=***********      <=Numbers are removed here
[default]
include => sip-icmlocal
include => sip-icmremote
include => sip
include => inbound-pstn
include => outbound-pstn
include => outbound-iadotie

exten =>12541759,1,answer
exten =>12541759,2,Dial(SIP/${RECEPTION},20)
exten =>12541759,3,Playback(/toddsounds/high)
exten =>12541759,4,Playback(/toddsounds/this-call-may-be)
exten =>12541759,5,Playback(/toddsounds/recorded)
exten =>12541759,6,WaitMusiconhold(2000)
exten =>12541759,6,Playback(/toddsounds/busy-pls-hold)
exten =>12541759,7,Dial(SIP/user2,15)
exten =>12541759,8,Voicemail(129@default)
exten =>12541759,9,hangup
```

```
[macro-fastbusy]
exten => s,1,Answer
exten => s,2,Wait 1
exten => s,3,Playback(/toddsounds/thnk-u-for-patience)
exten => s,4,Wait(30)
exten => s,5,Hangup

[sip]
;exten =>2000,1,Dial(SIP/astra480,20)
;exten =>2000,2,Hangup

exten =>101,1,Dial(SIP/astra480L1),20
exten =>101,2,Hangup

exten =>102,1,Dial(SIP/astra480L2),20
exten =>102,2,Hangup

;exten =>2003,1,Dial(SIP/astra9133,30)
;exten =>2003,2,PLayback(tt-allbusy)
;exten =>2003,3,Congestion

exten =>171,1,Voicemailmain

[sip-icmlocal]

exten =>516,1,Dial(Sip/user4,20)
exten =>516,2,Playback(tt-albusy)
exten =>516,3,WaitMusiconhold(20)
exten =>516,4,Dial(Sip/user4,20)
exten =>516,5,Voicemail(116@default)
exten =>516,6,Playback(vm-goodbye)
exten =>516,7,hangup

exten =>116,1,Answer
exten =>116,2,Voicemailmain

exten =>529,1,Dial(Sip/user2,20)
exten =>529,2,Playback(tt-albusy)
exten =>529,3,WaitMusiconhold(20)
exten =>529,4,Dial(Sip/user2,20)
exten =>529,5,Voicemail(129@default)
exten =>529,6,Playback(vm-goodbye)
exten =>529,7,hangup

exten =>129,1,Answer
exten =>129,2,Voicemailmain


exten =>547,1,Dial(Sip/user3,20)
exten =>547,2,Playback(tt-allbusy)
exten =>547,3,Voicemail(147@default)
exten =>547,4,Playback(vm-goodbye)
exten =>547,5,hangup

exten =>147,1,Answer
exten =>147,2,Voicemailmain

exten =>300,1,answer
exten =>300,2,Playback(tt-allbusy)
exten =>300,3,Dial(IAX2/iaxy1,20)
exten =>300,4,Playback(vm-goodbye)
exten =>300,5,hangup



include => sip
include => outbound-iaxdotie
include => outbound-pstn
include => outbound-blueface
include => sip-icmremote
include => speeddials
```

```
[sip-icmremote]

exten =>929,1,Dial(Sip/user2_home,20)
exten =>929,2,Playback(/toddsounds/call-fwd-no-ans)
exten =>929,3,Dial(${dialoutpstn}/*********)      <=Numbers are removed here
exten =>929,4,hangup

exten =>85,1,Dial(Sip/user1home,20)
exten =>85,2,Playback(tt-allbusy)
exten =>85,3,Voicemail(151@default)
exten =>85,4,Playback(vm-goodbye)
exten =>85,5,hangup

exten =>86,1,Dial(Sip/user1homesoft,20)
exten =>86,2,Playback(tt-allbusy)
exten =>86,3,Voicemail(151@default)
exten =>86,4,Playback(vm-goodbye)
exten =>86,5,hangup
exten =>185,1,Answer
exten =>185,2,Voicemailmain

include => outbound-iaxdotie
include => outbound-pstn
include => sip-icmremote
include  => sip-icmlocal
include => sip


[inbound-pstn]
exten =>s,1,ANSWER()
exten =>s,2,Dial(SIP/${RECEPTION},5)
exten =>s,3,WaitMusiconhold(2000)
exten =>s,4,Dial(SIP/astra480L2,15)
exten =>s,5,Voicemail(171@default)
exten =>s,6,Playback(vm-goodbye)
exten =>s,7,Hangup

[outbound-pstn]

exten =>_9.,1,Dial(${dialoutpstn}/${EXTEN:1})
exten =>_9.,2,macro(fastbusy)
exten =>_9.,3,Hangup


[outbound-iaxdotie]
exten => _2.,1,Dial,IAX2/iaxdotie/${EXTEN:1}
exten => _2.,2,Hangup
exten => _2.,102,Hangup

[iaxdotie-incoming]
exten => s,1,Answer
exten => s,2,Dial(SIP/${RECCEPTION},20)
exten =>s,3,Playback(/toddsounds/busy-pls-hold)
exten =>s,4,Dial(SIP/astra480L2,15)
exten =>s,5,Voicemail(171@default)
exten =>s,6,Playback(vm-goodbye)
exten =>s,7,Hangup

exten =>12541759,1,answer
exten =>12541759,2,Dial(SIP/${RECEPTION},20)
exten =>12541759,3,Playback(/toddsounds/high)
exten =>12541759,4,hangup

[outbound-blueface]
exten => _7.,1,Dial,IAX2/blueface/${EXTEN:1}
exten => _7.,2,Hangup
exten => _7.,102,Hangup

exten => 99,1,Dial(IAX2/blueface/303)
exten => 99,2,Hangup

[bluefacein]
```

```
exten => s,1,Answer
exten => s,2,Dial(SIP/${RECCEPTION},20)
exten =>s,3,Playback(/toddsounds/busy-pls-hold)
exten =>s,4,Dial(SIP/user2,15)
exten =>s,5,Voicemail(171@default)
exten =>s,6,Playback(vm-goodbye)
exten =>s,7,Hangup


[speeddials]
exten => 640,1,Dial(IAX2/blueface/${SpeedDial40})
exten => 640,2,Hangup
exten => 640,102,Hangup

exten => 641,1,Dial(IAX2/blueface/${SpeedDial41})
exten => 641,2,Hangup
exten => 641,102,Hangup
```

4.4 – sip.conf

```
; Asterisk Test System XYSystems
; sip.conf
[general]
port = 5060
bindaddr = 0.0.0.0
context = default


[astra480]
type=friend
secret=password
disallow=all
allow=speex
host=dynamic
context=default

[astra480L1]
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=default

[astra480L2]
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=default

[astra9133]
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=default

[user2_home]
callerid="929 User2 Home"
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=sip-icmremote

[user2]
callerid="529 User2 Office"
```

```
type=friend
secret=password
disallow=all
allow=gsm
host=dynamic
context=sip-icmlocal

[User3]
callerid="547 – User3"
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=sip-icmlocal

[user4]
callerid="516 – User4"
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic

[user1home]
callerid="85 – User1"
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=sip-icmremote

[user1homesoft]
callerid="86 – User1"
type=friend
secret=password
disallow=all
allow=ulaw
host=dynamic
context=sip-icmremote


[test]
callerid="test"
type=friend
secret=password
allow=all
host=dynamic
context=sip-icmlocal
```

4.5 – zaptel.conf

```
#  Asterisk Test System ICMUNICOMP
# Zaptel Configuration File
#
fxsks=4
loadzone=uk
defaultzone=us
```

4.6 – iax.conf

```
[general]

register => 888:paswrd@212.147.134.3
register => myname:mypass@iax.blueface.ie
; added for www.iax.ie config 020806
bandwidth=low
disallow=lpc10                  ; Icky sound quality...  Mr. Roboto.

jitterbuffer=no
forcejitterbuffer=no
tos=lowdelay
```

```
autokill=yes
[guest]
type=user
context=default
callerid="Guest IAX User"

;
; Trust Caller*ID Coming from iaxtel.com
;
[iaxtel]
type=user
context=default
auth=rsa
inkeys=iaxtel

;
; Trust Caller*ID Coming from iax.fwdnet.net
;
[iaxfwd]
type=user
context=default
auth=rsa
inkeys=freeworlddialup

[demo]
type=peer
username=asterisk
secret=supersecret
host=216.207.245.47

[iaxdotie]
type=friend
host=212.147.134.3
username=510
secret=xxxxxxxx   <= removed password
auth=md5
context=iaxdotie-incoming
disallow=all
allow=gsm
allow=ulaw
jitterbuffer=yes
dropcount=1
tos=0x18

[iaxy1]
type=friend
accountcode=iaxy1
host=dynamic
secret=password
countext=sip-icmlocal
disallow=all
allow=ulaw
callerid="My IAXY"
trunk=no


[blueface]
type=friend
host=iax.blueface.ie
username=my name
secret=xxxxx    <= removed password
context=bluefacein
```

Appendix 5 – Bandwidth measurements for several asterisk calls

No outgoing channels – Figure 1

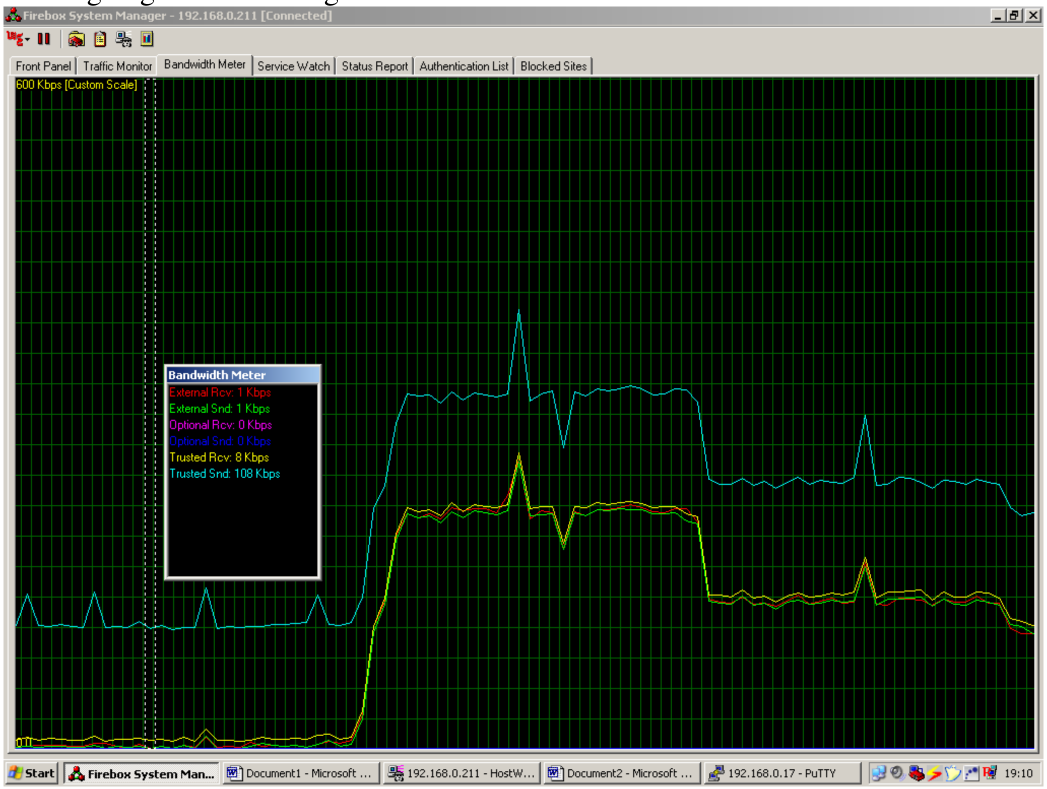


Figure A5.1 External Send = 1Kbs External Receive = 1Kbps

1 called was placed from a softphone and the result was as follows

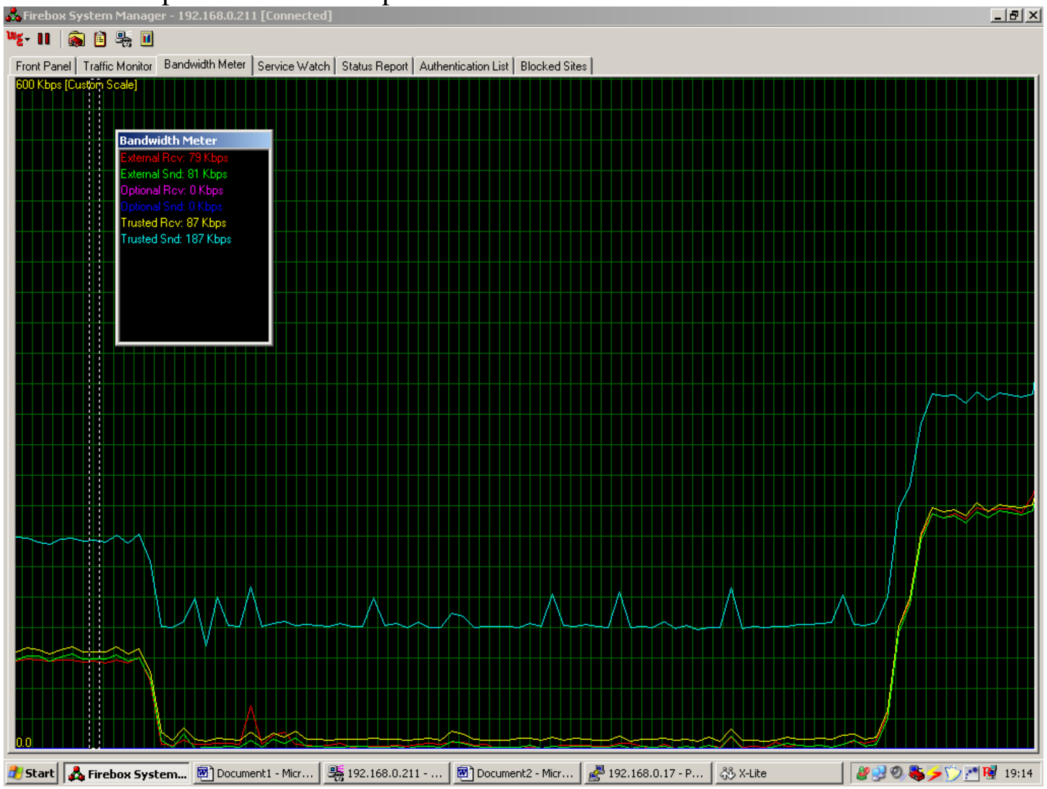


Figure A5.2 External Send = 81Kbs External Receive = 79Kbps

Channel Peer Username ID (Lo/Rem) Seq (Tx/Rx) Lag Jitter JitBuf Format
IAX2/iaxdotie-6 212.147.134.3 510 00006/00014 00012/00016 00000ms 0000ms 0040ms ulaw
1 active IAX channel
slacklinux*CLI> sip show channels
Peer User/ANR Call ID Seq (Tx/Rx) Form Hold Last Message
192.168.0.97 test 1a41462f467 00101/00003 ulaw Yes Rx: ACK
1 active SIP channel

We place a second call using the Softphone – and bandwidth increases as per Fig 2

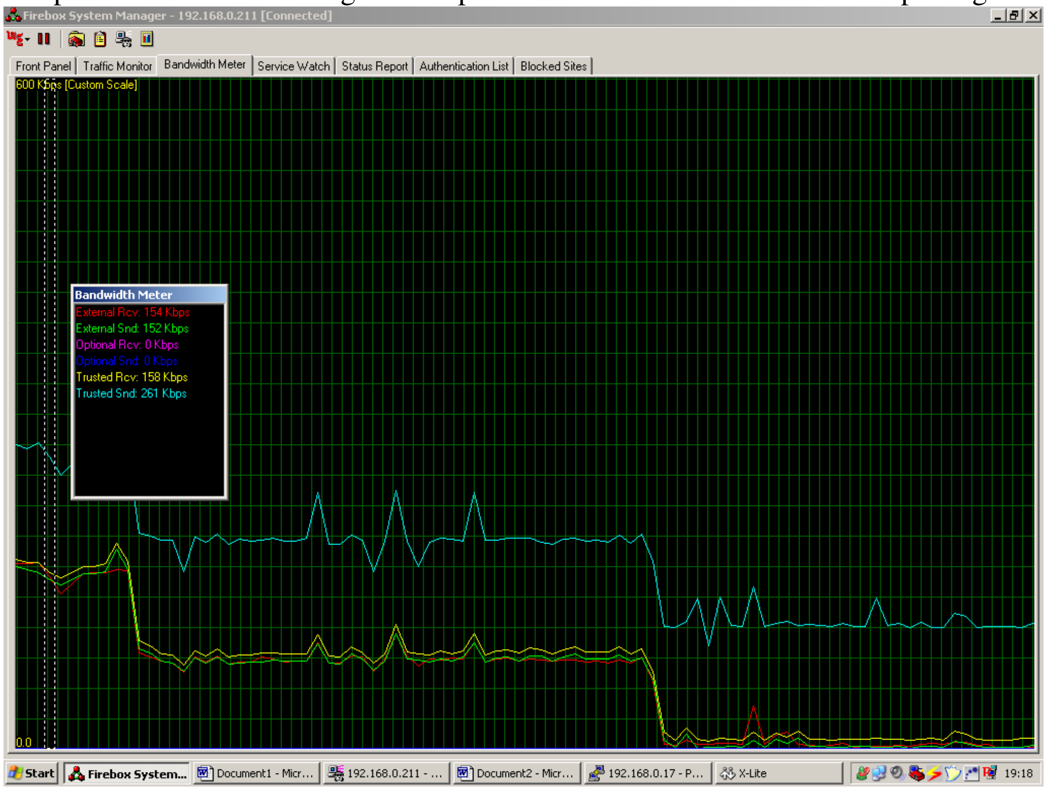


Figure A5.3 External Send = 152Kbs External Receive = 154Kbps

```
slacklinux*CLI> sip show channels
Peer           User/ANR   Call ID      Seq (Tx/Rx)  Form  Hold    Last Message
192.168.0.97   test       3114ac4ad76  00101/00003  ulaw  Yes     Rx: ACK
192.168.0.97   test       1a41462f467  00101/00005  ulaw  Yes     Rx: ACK
2 active SIP channels
slacklinux*CLI> iax2 show channels
Channel           Peer            Username   ID (Lo/Rem)  Seq (Tx/Rx)  Lag      Jitter  JitBuf  Format
IAX2/iaxdotie-3   212.147.134.3   510        00003/00008  00031/00033  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-6   212.147.134.3   510        00006/00014  00105/00104  00000ms  0000ms  0040ms  ulaw
2 active IAX channels
```

Another call was made with hardphone – to a pstn destination through VoIP using iax

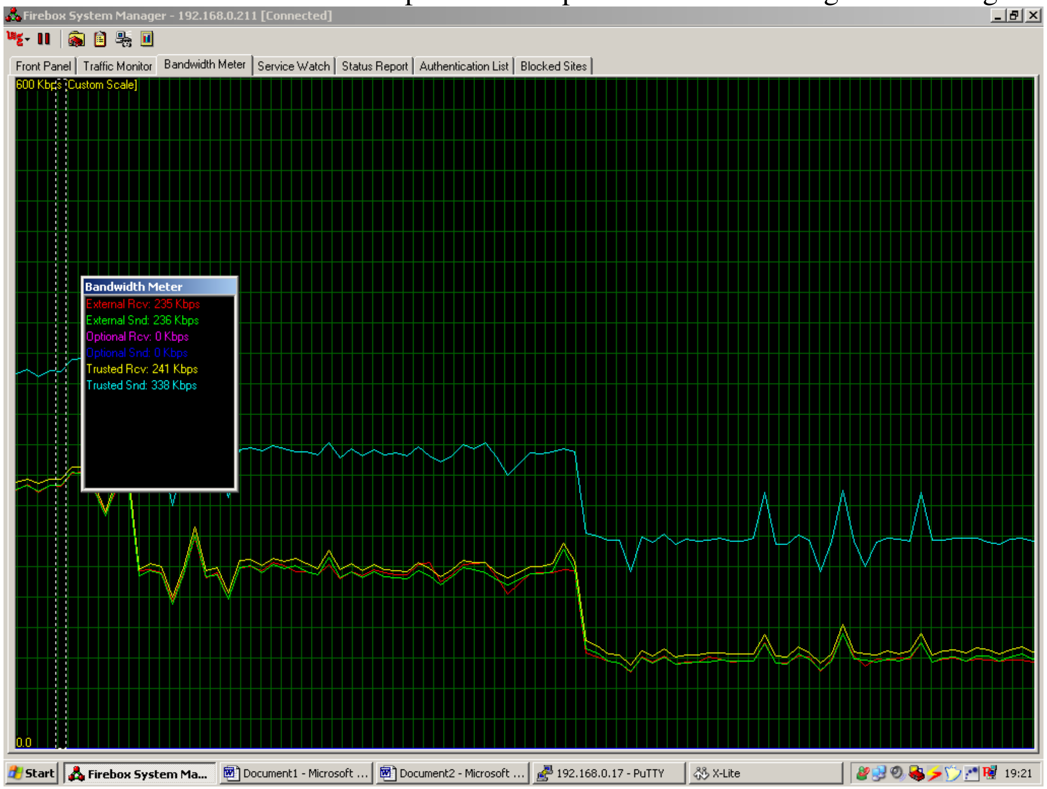


Figure A5.4 External Send = 236Kbs External Receive = 235Kbps

```
slacklinux*CLI> sip show channels
Peer            User/ANR    Call ID      Seq (Tx/Rx)  Form  Hold     Last Message
192.168.0.149   astra480L1  6aac9bfc2ed  00101/454202673  ulaw  Yes     Rx: ACK
192.168.0.97    test        3114ac4ad76  00101/00003  ulaw  Yes     Rx: ACK
192.168.0.97    test        1a41462f467  00101/00005  ulaw  Yes     Rx: ACK
3 active SIP channels
slacklinux*CLI> iax2 show channels
Channel           Peer            Username  ID (Lo/Rem)  Seq (Tx/Rx)  Lag      Jitter  JitBuf  Format
IAX2/iaxdotie-3   212.147.134.3   510       00003/00008  00092/00091  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-5   212.147.134.3   510       00005/00012  00029/00031  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-6   212.147.134.3   510       00006/00014  00167/00164  00000ms  0000ms  0040ms  ulaw
3 active IAX channels
slacklinux*CLI>
```

Now we add another call from the hard phone – to pstn and bandwidth increase as before

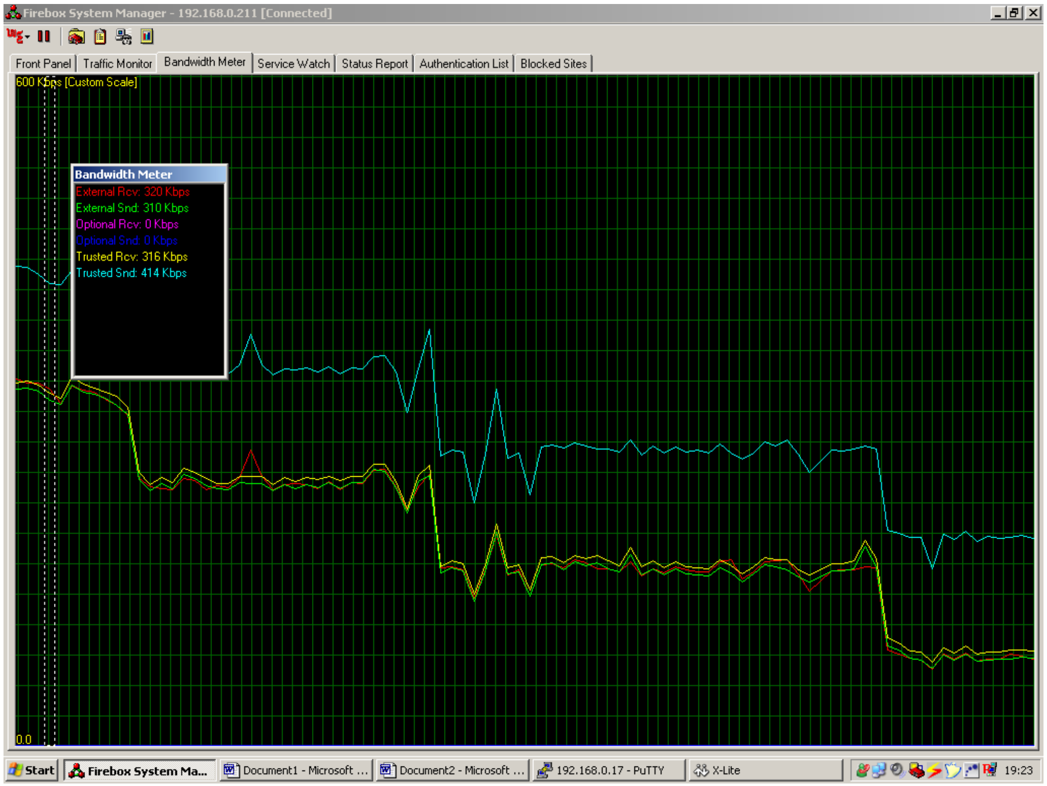


Figure A5.5 External Send = 310Kbs External Receive = 320Kbps

```
slacklinux*CLI> sip show channels
Peer             User/ANR    Call ID      Seq (Tx/Rx)  Form  Hold     Last Message
192.168.0.149    astra480L2  9a434a1583d  00101/657390368  ulaw  Yes     Rx: ACK
192.168.0.149    astra480L1  6aac9bfc2ed  00101/454202673  ulaw  Yes     Rx: ACK
192.168.0.97     test        3114ac4ad76  00101/00003  ulaw  Yes     Rx: ACK
192.168.0.97     test        1a41462f467  00101/00005  ulaw  Yes     Rx: ACK
4 active SIP channels
slacklinux*CLI> iax2 show channels
Channel              Peer            Username   ID (Lo/Rem)  Seq (Tx/Rx)  Lag      Jitter  JitBuf  Format
IAX2/iaxdotie-3      212.147.134.3   510        00003/00008  00132/00129  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-5      212.147.134.3   510        00005/00012  00069/00069  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-6      212.147.134.3   510        00006/00014  00205/00200  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-9      212.147.134.3   510        00009/00010  00025/00027  00000ms  0000ms  0040ms  ulaw
4 active IAX channels
```

Now we dial in through iax and then we have 5 calls

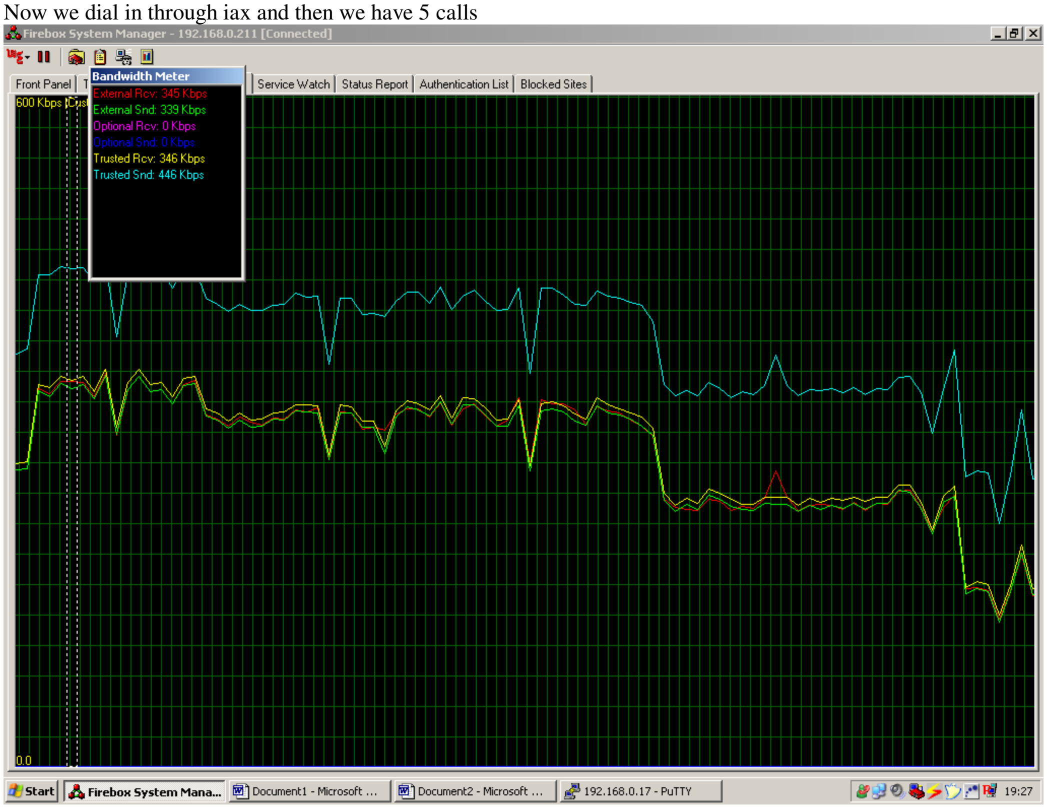


Figure A5.6 External Send = 339Kbs External Receive = 345Kbps

Channel	Peer	Username	ID (Lo/Rem)	Seq (Tx/Rx)	Lag	Jitter	JitBuf	Format
IAX2/iaxdotie-3	212.147.134.3	510	00003/00008	00213/00207	00000ms	0000ms	0040ms	ulaw
IAX2/iaxdotie-4	212.147.134.3	510	00004/00016	00006/00009	00000ms	0000ms	0040ms	ulaw
IAX2/iaxdotie-5	212.147.134.3	510	00005/00012	00151/00147	00000ms	0000ms	0040ms	ulaw
IAX2/iaxdotie-9	212.147.134.3	510	00009/00010	00111/00109	00000ms	0000ms	0040ms	ulaw
IAX2/iaxdotie-11	212.147.134.3	guest	00011/00023	00043/00042	00000ms	-0001ms	0000ms	gsm

5 active IAX channels
slacklinux*CLI>

With seven channels this it the picture

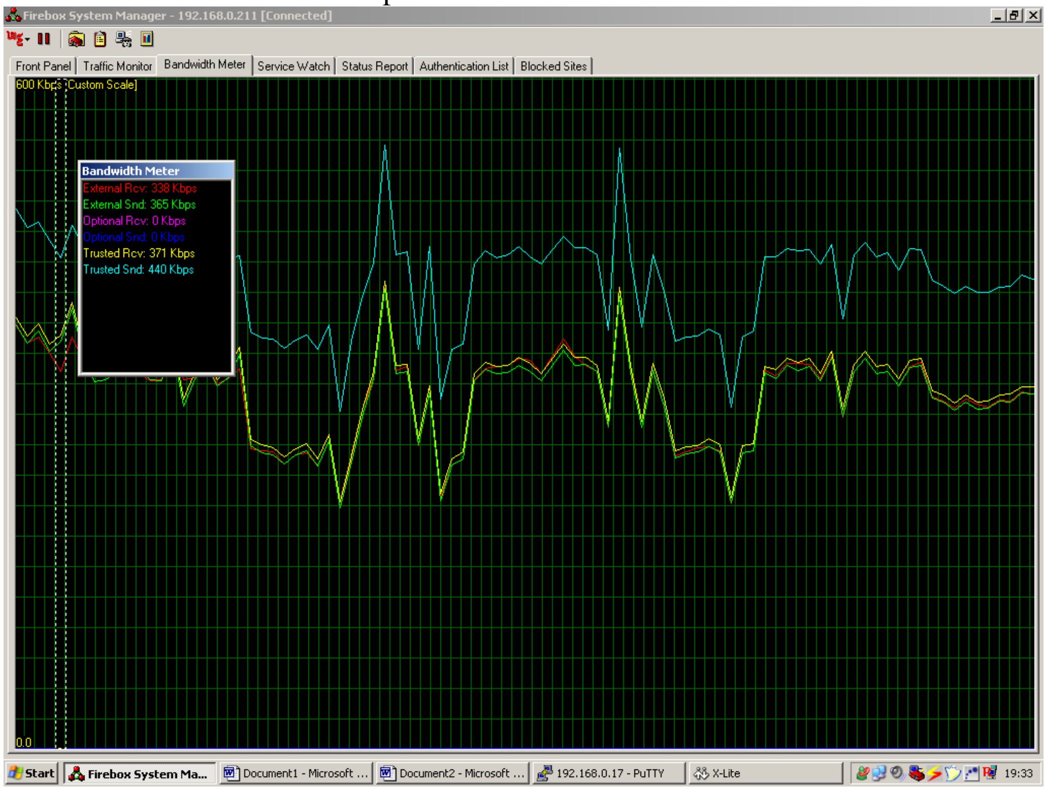


Figure A5.7 External Send = 365Kbs External Receive = 338Kbps

```
Channel            Peer           Username    ID (Lo/Rem)  Seq (Tx/Rx)  Lag      Jitter  JitBuf  Format
IAX2/iaxdotie-1      212.147.134.3    510        00001/00018  00056/00058  00000ms  0000ms  0040ms  ulaw
(None)             213.202.151.119  (None)      00003/07259  00001/00001  00000ms  -0001ms  0000ms  unknow
IAX2/iaxdotie-4      212.147.134.3    510        00004/00016  00092/00091  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-6      212.147.134.3    510        00006/00001  00031/00034  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-9      212.147.134.3    510        00009/00010  00195/00189  00000ms  0000ms  0040ms  ulaw
IAX2/iaxdotie-10     212.147.134.3    510        00010/00009  00009/00011  00000ms  0000ms  0040ms  gsm
IAX2/iaxdotie-11     212.147.134.3    guest      00011/00023  00129/00128  00000ms  -0001ms  0000ms  gsm
7 active IAX channels
```

```
slacklinux*CLI> show channels
Channel              Location             State   Application(Data)
Zap/4-1              s@default:3          Up      WaitMusicOnHold(2000)
IAX2/iaxdotie-10     s@iaxdotie-incoming: Up      Bridged Call(SIP/Albert-081a92
SIP/Albert-081a9288  20578663557@sip-icml Up      Dial(IAX2/iaxdotie/0578663557)
IAX2/iaxdotie-6      s@iaxdotie-incoming: Up      Bridged Call(SIP/test-081d9880
SIP/test-081d9880    20578622651@sip-icml Up      Dial(IAX2/iaxdotie/0578622651)
IAX2/iaxdotie-1      s@iaxdotie-incoming: Up      Bridged Call(SIP/astra480L1-08
SIP/astra480L1-081d2 20578622651@default: Up      Dial(IAX2/iaxdotie/0578622651)
IAX2/iaxdotie-4      s@iaxdotie-incoming: Up      Bridged Call(SIP/test-081b4818
SIP/test-081b4818    20578622651@sip-icml Up      Dial(IAX2/iaxdotie/0578622651)
IAX2/iaxdotie-11     12541759@default:6   Up      WaitMusicOnHold(2000)
IAX2/iaxdotie-9      s@iaxdotie-incoming: Up      Bridged Call(SIP/astra480L2-08
SIP/astra480L2-081e9 20578622651@default: Up      Dial(IAX2/iaxdotie/0578622651)
12 active channels
7 active calls
```

After all the call were dropped the bandwidth used drops

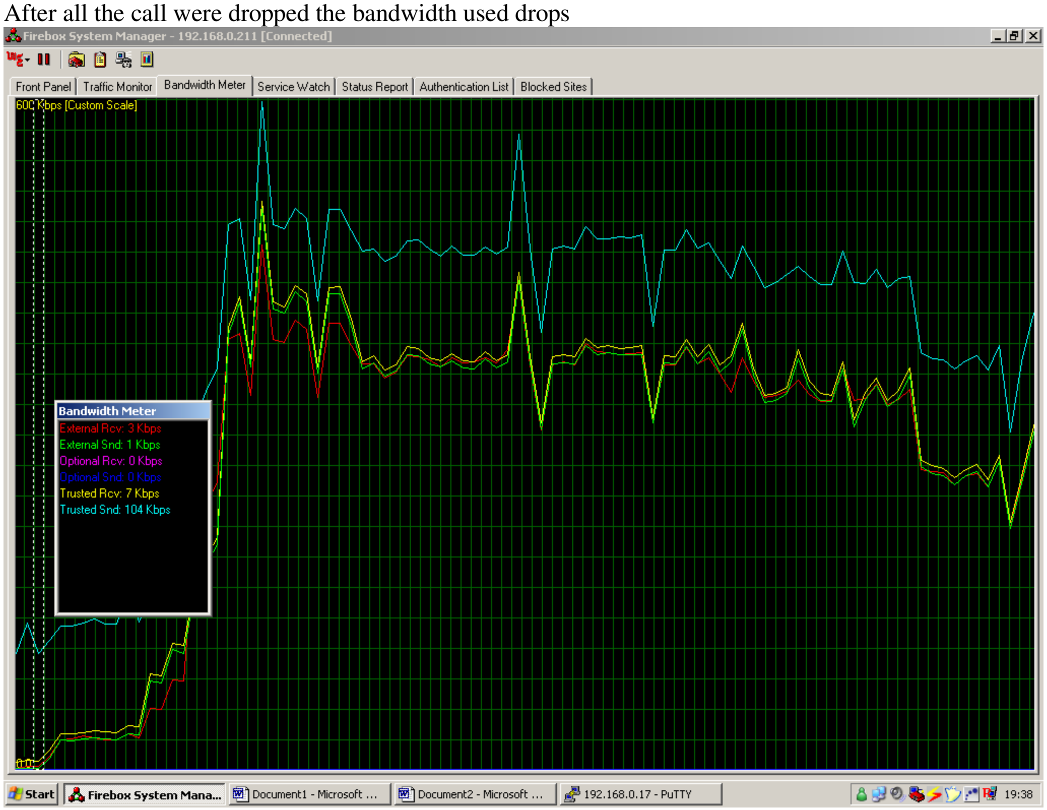


Figure A5.8 External Send = 1Kbs External Receive = 3Kbps

Appendix 6 – Bandwidth measurements for several asterisk calls

1 Call using GSM Bandwidth at 27-36KB

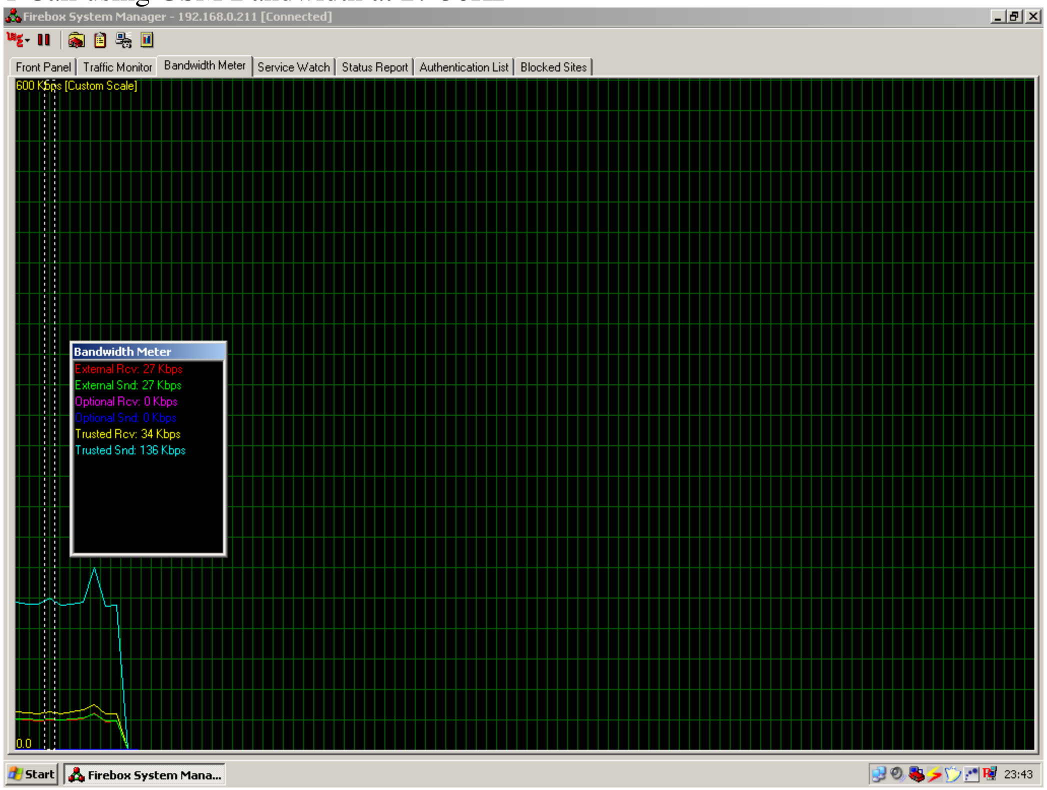


Figure A6.1 External Send = 27Kbs External Receive = 27Kbps

iax2 show channels

Channel	Peer	Username	ID (Lo/Rem)	Seq (Tx/Rx)	Lag	Jitter	JitBuf	Format
IAX2/iaxdotie-1	212.147.134.3	guest	00001/00012	00013/00010	00000ms	-0001ms	0000ms	gsm

1 active IAX channel

Second call was using ulaw codec bandwidth now up to 106Kbps

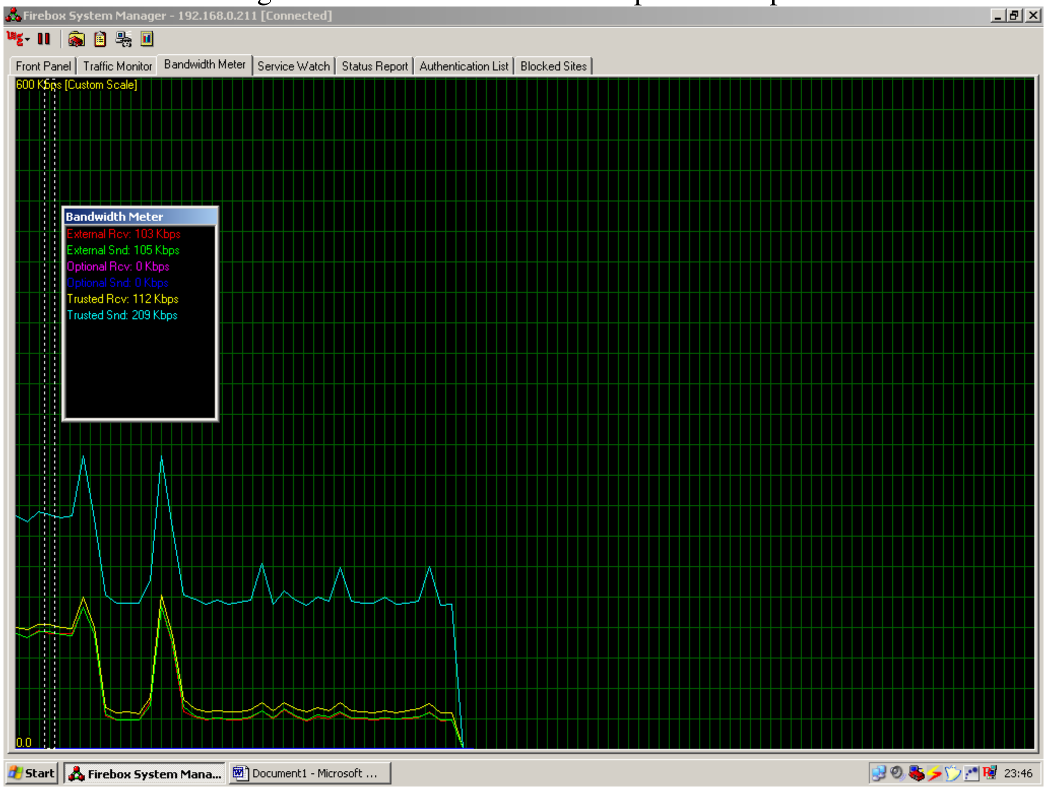


Figure A6.2 External Send = 105Kbs External Receive = 103Kbps

```
*CLI> iax2 show channels
Channel            Peer            Username   ID (Lo/Rem)  Seq (Tx/Rx)  Lag      Jitter  JitBuf  Format
IAX2/iaxdotie-1    212.147.134.3   guest      00001/00012  00118/00115  00000ms  -0001ms  0000ms  gsm
IAX2/iaxdotie-6    212.147.134.3   510        00006/00005  00050/00051  00000ms  0000ms  0040ms  ulaw
2 active IAX channels
```

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