Voice Over IP

BTech Project

Semester 2 Report

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Abstract

July 4th 2005, AUT's VOIP system went alive. The university's 1,700+ staff started to use a proprietary digital phone system with an in-house PABX. This made AUT become the first New Zealand university to move to a VOIP system. [1] March 2005, Tauranga Hospital set VOP phones for their 800 staff…… VOIP has hit New Zealand, more and more people and organizations start to use VOIP. Companies such as Telecom, TelstraClear, Woosh, IHug and SipServe have launched their VOIP plans. VOIP has brought us new concept of making voice communication. So what is VOIP?

Voice-over-Internet Protocol, Voice-over-IP, or more simply VOIP is a term used in telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). It is a technique that allows voice traffic to be transported across an IP-based data network or the Internet. The voice signal is sampled, compressed and encapsulated into data packets to allow it to be switched, routed and bridged along with all other data packets across the Local and Wide area network. [2]

After one year study and research on VOIP and relevant technology, our project is finally coming to an end. During this year, we planned, organized, researched, compared and finally concluded. This report will help us to gain a good review of the work done during the whole year, make a summary and conclude the project. The first part of this report brings us a brief summary of the work done and result created in the first half of this project. Study and implementation taken in the second half will be stated in detail in the second part of the report. Lastly, a working summary and assessment will be given to conclude the report.

1. Project Review

This section takes a back look at the definition and scope of the project and work completed during the first half. This section helps us understand the project and gives us a big picture of the network we have developed.

1.1 Project description

VOIP project is a continuation of the work done in the last year. The goal of this project is to replace the original analogue PBX system in the Compucon House with an IP PABX system to integrate voice with the existing data network to provide same or better service to the Compucon House. The main purpose of implementing the VOIP system is to reduce the cost of using and maintaining the original telephone system in the next five years. Further research is required to be done based on the previous work in the first part. In the second part, we are required to design a network that supports the VOIP technology, followed by a detailed installation and test plan.

1.2 Project scope

Our main task in this project is to have a relevant and detailed research on the VOIP topic. Following this, we categorize the project scope into vertical scope and horizontal scope.

Vertical Scope:

- Objective and definition of technology
- Functionality, scalability, availability, security, ease of use
- Information flow block diagram of related technologies (the big picture)
- Standards, approving organization, chronologic information of endorsement, Maturity of standard and technology.

Horizontal Scope:

- Relevance to BTECH syllabus
- Product Comparison such as with popular brands
- Deployment information worldwide and in New Zealand.
- Approaches used by System Integrators especially in New Zealand.
- Examples of system integrators are EDS, IBM, HP, Datacom, Telecom Advanced Solutions, Geni and Axon.

1.3 VOIP technology

VOIP converts the voice signal from your telephone into a digital signal that travels over the Internet. There are several ways that VOIP is used. The most common three ways are ATA, IP phones and computer-to-computer.

- *ATA* (Analog Telephone Adapter) is way that enables you to connect your normal analogue telephone through an adapter to your computer or other network devices.
- *IP Phones* is another way to use VOIP. IP phone acts like a network device which has its own IP address and is connected using the Ethernet connector.
- *Computer-to-computer call* is a software solution that is a simple and cheap way to make VOIP calls.

So why should we use VOIP? We listed 3 main advantages in the mid semester report. They are:

- *Cost*: VOIP technology utilizes the network by integrating data and voice service which saves the money originally spent on the analogue PABX.

- *Functionality:* Being integrated with the network services, VOIP facilitates some features that are difficult to achieve for the original telephone network.
- *Mobility:* VOIP allows users to travel anywhere in the world and still make and receive phone calls. Video conversation, message data or file exchange are in parallel with audio conferencing and information passing.

A set of concerns including quality of service, compatibility and availability had also been studied by the end of the last semester. We finally concluded that the future of VOIP is very solid and will be a definite solution after the analog telephony.

1.4 Possible solutions

After the study of VOIP, we started to think about the implementation in Compucon House. There were three possible solutions available for Compucon House.

- *Current PSTN:* Having considered and compared the international call rate provided by Telecom, we concluded that this option would not reach our goal of saving the cost from telephone calls.
- *VOIP service provider:* We introduced two examples of VOIP service provider. This one seemed to be a good solution, however, extra parties are needed to be involved and extra fees are needed to be paid.
- *Build own VOIP network:* Building own VOIP network requires the installation VOIP routers that supports the technology. We compared the Cisco VOIP router solution and the IPX-2000 solutions. It seemed both are very good and suitable for the company.

Having considered the cost and functionality issues, we finally decided to build the company's own VOIP system using IPX-2000 router. IPX-2000 is used to build an "all-in-one" system that supports digital and IP-based communication, PSTN access, voice conference and centralized call control. This feature-rich PBX system allows us to have seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints. It allows telephone call to go through both PSTN and Internet through single device. This highly integrated system will benefit the company from the reduced cost for maintaining the data and voice networks.

Then we came up with a big picture (Figure 1) of the network which briefly shows the network arrangement of the VOIP system we were going to build.

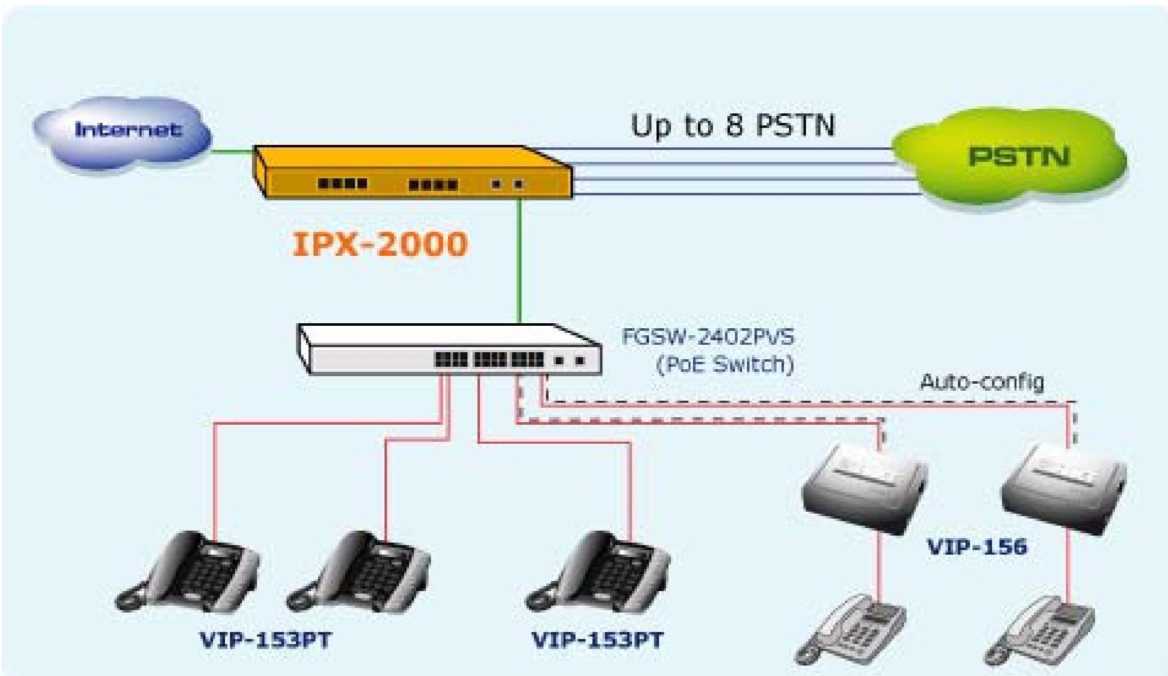

Figure 1 IPX-2000 network assumption

Finally, this network system leaved us five concerns, namely, availability, compatibility, QoS, control over voice service and voice and data integration. Those concerns leaded us to the start of the second half of the project.

2. Five concerns

In the first part of the study, we summarized five basic concerns that required us to consider in building a VOIP network system. In this part, we will take a look at how those concerns will affect the system and how we can address them.

2.1 Availability

We are building a telephone system in the company, so the first and the most important thing we need to consider is the telephone communication availability. We summarized four conditions and listed them in term of four questions. We then found the answers to them one by one.

With the new system:

- **Are we able to receive the phone calls from PSTN?**

  Under the consideration that all the business partners and customers may be unaware of the change we are making and that the system we are building is under a transitional period, we decide to keep the original PSTN network and maintain the use of analog phones. That means all calls originating from the PSTN will be received by the original analog PBX system. Therefore, in this period, we do not need to be able to receive PSTN calls using our IPBX system.

- **Are we able to make a call to PSTN?**

  According to the IPX-2000 user's manual, it supports the call from IP phone to PSTN. But calls will go through the PSTN subscriber lines but not the internet. IPX-2000 supports analog telephone direct connection, so we can also use analog telephone to make a call to the PSTN.

- **Are we able to make a telephone call through internet?**

  This is how VOIP works. With IP phones, we can make telephone calls through internet using our new IPBX system. Calls from analog phone to internet are also available but we need to have a telephone adapter.

- **Are we able to receive phone calls from internet?**
  Internet telephone and the analog telephone with adapter can both receive calls from the internet.

Therefore, we can conclude by using a simple table to represent the communication availability and flexibility for the new system that is going to build in Compucon House.

	Internet	*PSTN*
Call	IP Phone Analog Phone(With adapter)	IP Phone Analog Phone
Receive Call	IP Phone Analog Phone(With adapter)	Not available

Table 1 Communication availability

2.2 Compatibility

Compatibility is another important issue we have to think about. Here we need to address the questions like “Can we make phone calls using our VOIP system to all the other telephones?”, “Can the outside world recognize our devices?” These questions lead us to think about the devices we are using, whether they are compatible with other devices used inside the company or the devices outside the world. Other thinks such as the cables, connectors, protocol are also important things we need to look into.

We studied two other known brands in the purpose of finding some compatibility issues and to see how well IPX-2000 can work compared to the others.

Panasonic KX-TDA200

Figure 2 Panasonic KX-TDA200

- A flexible system that easily configured from 8 extensions up to 256 with DXDP, allowing the system to grow with your company.
- Up to 192 ports and max of 128 wireless telephones
- Expand the system by plugging in additional expansion cards without buying a whole new system
- Telephone compatibility: Works with Panasonic KX-T7000KX-T7400, 7600, 7720, 7731, 7750 series as well as KX-NT136, KX-NT265 and SLTsSupport wireless telephones
- USB port for computer configuration
- Voice codec: G.711, G.729a

[3]

Avaya IP406 V2 Office

Figure 3 Avaya IP 406 V2 Office

- An all-in-one solution specially designed to meet the communications challenges facing small and medium sized businesses.

- Provides a combination of up to 190 analogue and digital extensions, with capacity for 8 analogue trunks or 3 digital trunks (72 T1/PRI channels and 90 E1 channels).
- Allow analogue extension
- Eight 10/100 Mbps LAN Switched ports (layer-2).
- X.21/V35 WAN interface Supports both soft and hard phones
- 4 analogue ports
- Only supports Avaya IP phones

[4]

IPX-2000

- WAN port allows connection to a broadband modem or a WAN router
- LAN port allows connection to LAN switch
- FXO ports allow connection to FXS jacks on the wall or analog PBX
- USB ports allow connection to USB hard drive
- Works with various IP phones (desktop, WiFi, Bluetooth, and DECT), VOIP gateways, and analog telephone adapters
- Provides Internet access to all LAN devices through Network Address Translation (NAT)
- TCP/IP, NAT, DHCP, HTTPs, DNS protocols are used for Internet sharing
- HTTP and Web browser are used for configuration and provisioning
- Voice codec support: G.711, G.726, GSM, G.723.1 (5.3, 6.3kbps), G.729A (8kbps)

As we can see, all three systems have their own features and they all work very well in a VOIP system. And as you can see, IPX-2000 shows a full compatibility with the outside world: single device with various phone types of phone support, compete voice codec, simple web configuration, various technology support etc.

It provides similar compatibility and less restriction then the Panasonic and Avaya systems.

2.3 QoS

QoS (Quality of Service) is one of the most important factors that affects the telecommunication. Voice is transmitted over the internet after its conversion and encoding. Packets sometimes don't reach the destination or get delayed en route. Either of these can occur when there's congestion on the IP network. Generally, all VOIP products use the User Datagram Protocol (UDP) and the Real-Time Protocol (RTP), over IP. This means that the voice packets that are lost aren't retransmitted, whereas most IP "data" packets use the Transmission Control Protocol (TCP), which detects and arranges for retransmission of lost packets. Therefore, we have to face the questions such as "How much traffic will voice add to the existing network?", "Would it cause a problem?" and probably "How can we provide voice quality to the VOIP system we are going to build?"

Voice quality is influenced by several factors such as choice of codec, delay, packet loss, jitter etc. Here we only studied the two major factors, latency (delay) and jitter. So what are latency and jitter?

- *Latency* – Delay for the packet delivery
- *Jitter* - Variation in delay for packet delivery [5]

After a while of study, we found that there are solutions we can use to reduce those effects.

- *Resource reservation* - to make sure that the VOIP call has the bandwidth needed allocated from point to point before the conversation takes place. Only works on local side.
- *Prioritization* - Here, the end point suggest a priority on the packets and each router decides if it will honour this request or not.

IPX-2000's QoS feature allows user to have some control over the voice quality through resource reservation.

Figure 4 IPX-2000's QoS service

Above is a screenshot of the VOIP QoS service configuration. Input the reserved bandwidth for VOIP communication. The uplink VOIP reserved could be, say 192 out of 256 kbps to allow 2 concurrent G.711 calls

2.4 Control over voice service

IPX-2000 provides call control and media relay services to SIP clients and applications. After we studied the IP-2000 product specification we listed the following basic features provided by IPX-2000.

- SIP Registrar
- SIP Outbound Proxy with media relay
- SIP Gateway (FXO)
- SIP PBX for extension calls
- Auto attendant
- Voice mail
- Meet-me conference

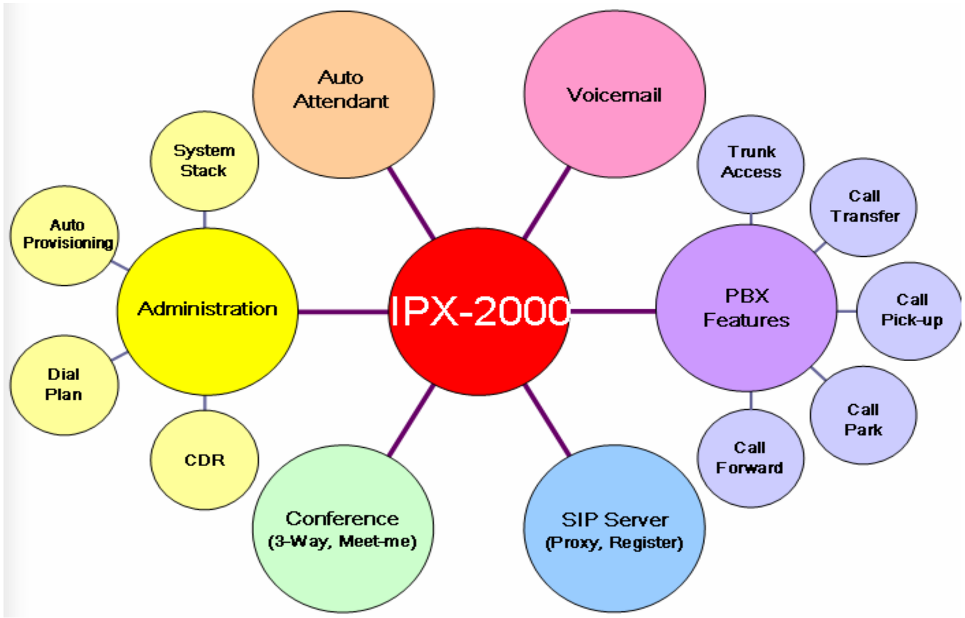

Figure 5 IPX-2000 Voice service control

Then we had a detailed study with user manual and summarized the IPX-2000 voice services in a graph shown as figure 5. We also found that the control over voice services is manageable and user friendly.

2.5 Voice and data integration

- IPX-2000 not only provides IP PBX services but also provides Internet access to all LAN devices
- IPX-2000 has a built-in suite of voice application for supplementary services and therefore no special-purpose hardware is required
  - Data and voice network share infrastructure
  - Lower cost than having two different infrastructures
  - Data and voice integration
- Web-browser interface to the data network configuration and voice service provisioning
  - Manageability of both networks

3. Standards and protocols

This section works with various standard and protocols used by IPX-2000. In this part, we studied some technical standards involved in VOIP in the purpose of gaining some technical and networking knowledge so as to help us install and maintain the IPX-2000 system.

3.1 Call control

In VOIP, the digital signal processor (DSP) segments the voice signal into frames and stores them in voice packets. These voice packets are transported using IP in compliance with a call control protocol for transmitting multimedia (voice, video, fax and data) across a network. There are many call control protocol available in the world: H.323 (ITU), MGCP (level 3, Bellcore, Cisco, Nortel), MEGACO/H.GCP (IETF), SIP (IETF), T.38 (ITU), SIGTRAN (IETF), Skinny (Cisco) etc. [6] We have studied 3 of them, namely, Megaco, MGCP and SIP.

Megaco – The Media Gateway Control Protocol is a signalling protocol, used between a Media Gateway and a Media Gateway Controller (also known as a Call Agent or a Soft Switch) in a VOIP network. It defines the necessary signalling mechanism to allow a Media Gateway Controller (Call agent) to control gateways in order to support voice/fax calls between PSTN-IP or IP-IP networks. This protocol is defined by IETF RFC 3525 and was the result of joint work of IETF and ITU. It is also known as H.248. H.248 is the name given to it by the ITU, Megaco is the IETF name. [7]

MGCP - Media Gateway Control Protocol is used for controlling telephony gateways from external call control elements called media gateway controllers or call agents. A telephony gateway is a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks. MGCP assumes a call control architecture where the call control intelligence is outside the gateways and handled by external call control elements. The MGCP assumes that these call control elements, or Call Agents, will synchronize with each other to send coherent commands to the gateways under their control. MGCP is, in essence, a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents.

SIP - The Session Initiation Protocol is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences". It was originally designed by Henning Schulzrinne (Columbia University) and is specified in the RFC 3261 of the IETF SIP Working Group. In November 2000, SIP was accepted as a 3GPP signaling protocol and permanent element of the IMS architecture. It is widely used as signaling protocol for Voice over IP, along with H.323

The motivating goal for SIP was to provide a signaling and call setup protocol for IP-based communications that can support a superset of the call processing functions and features present in the public switched telephone network (PSTN). Being a client-server base protocol, SIP provides the necessary protocol mechanisms so that the end user systems and proxy servers can provide various services. SIP is known as the most suitable protocol used for VOIP nowadays and the market for SIP devices (IP phones, VOIP routers, etc) are growing rapidly.

IPX-2000 uses SIP 2.0 for call control:

- IPX-2000 is an embedded call-processing server communicating with client stations with Session Initiation Protocol (SIP).

* IPX-2000 supports all hardware endpoints that use SIP and RTP for communication.
* Compatible with other IP-PBX devices that use SIP 2.0

3.2 Voice codec

Codec is short for coder/decoder. In telecommunications, it represents a standard for encoding or decoding a signal. For example, telephone companies use codecs to convert binary signals transmitted on their digital networks to analog signals converted on their analog networks. [10]

Following is a list of codec that IPX-2000 has implemented:

* G.711 - The G.711 is the codec that the PSTN uses. It is quite large for travel across the Internet, enabling 64Kbps of voice. We use this codec for PSTN compatibility
* G.726 - covering the transmission of voice at rates of 16, 24, 32, and 40 kbps.
* GSM - most popular for standard mobile phones in the world
* G.723.1 – operates at 5.3, 6.3kbps
* G.729A – operates at 8Kbps

3.3 Protocols

IPX-2000 works with various internet protocols:

* **TCP/IP**

  TCP/IP suits are used for data transmission through the internet
* **DHCP**

  Allow the use of DHCP to gain IP address

  Supports configuration of DHCP service

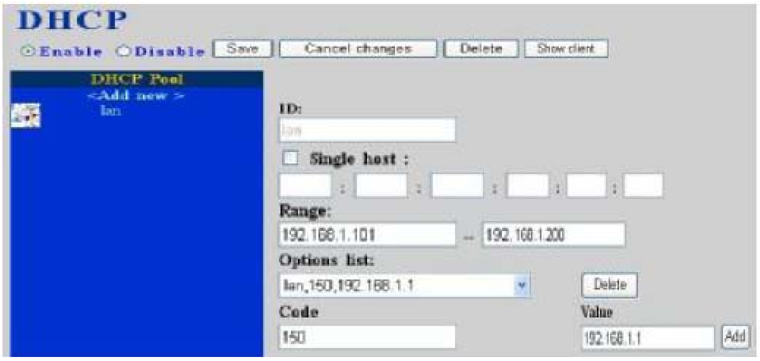

Figure 6 IPX-2000 DHCP service configuration

- **NAT**

  IPX-2000 provides Internet access to all LAN devices through Network Address Translation (NAT)

- **HTTPs**

  IPX-2000 uses web-browsable interface for system configuration

  HTTPs ensures a secure remote access to the configuration interface

- **DNS**

  Dynamic WAN IP address causes difficulty for inbound connections from remote clients or IP PBX systems.

  Adopt domain names provided by DynDNS

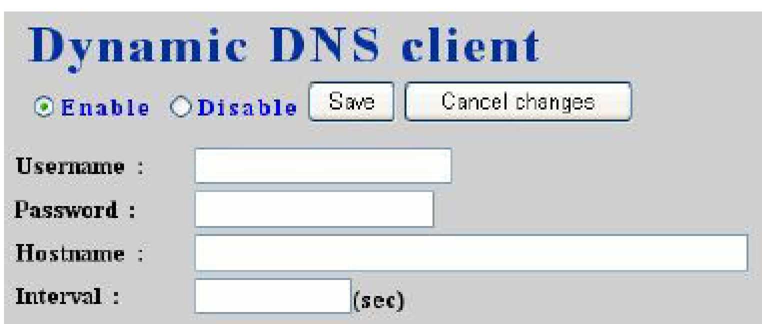

Figure 7 IPX-2000 Dynamic DNS service configuration

3.4 Connections

IPX-2000 supports three types of WAN connection

- **Static IP** - used in the static IP network environment
- **DHCP** – used when there exists a DHCP server to allocate IP address

- **PPPoE** - used when IPX-2000 is used as a gateway to the internet
- Choose the right connection type according to the network environment used by the company

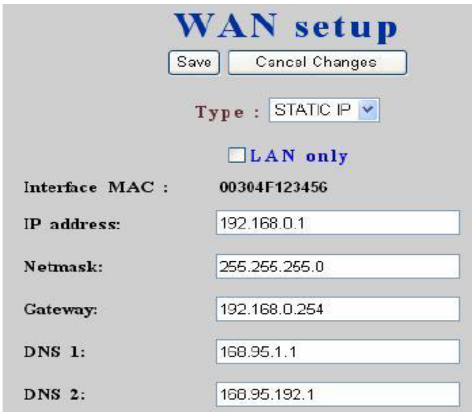

Figure 8 IPX-2000 WAN setup

3.5 Management

Network configuration

IPX-2000 uses a web based network configuration system. It comes with a web-browsable interface to the data network configuration and voice service provisioning, which brings the manageability of both networks together to facilitate administration locally or remotely.

Internet sharing

IPX-2000 uses HTTP and HTTPs for internet access and sharing.

4. Network Design

After the study of the relevant devices and technologies, we had gained some information and knowledge on building the VOIP network using IPX-2000. We started to look at the actual installation and commissioning process in Compucon House. In this part, we had a prototype design for the IPX-2000 network.

4.1 Current network

Before we started to design the new VOIP system, we needed to have an understanding of the current network system in Compucon House.

Telephone system

At present, there is an analogue phone network with an analogue PABX taking 8 trunk lines in and supporting 16 staff extensions.

Data network

There is a separate Ethernet data network connecting a server farm to 16 staff desktops. That is, the data and voice networks are totally independent and separate.

Having had a look at the existing system, we need to have a review of the requirement: We are going to add IPX-2000 and VOIP system to the data network, and phase out the analogue PABX network.

4.2 Analogue PABX

Since the analogue PABX is going to be phased out from the company, it is better to have the new VOIP system works independently from the analogue PABX. Therefore, for the ease of implementation, we listed the following proposal:

- No need to consider how the original telephone network works
- No connection needed between the data network and the analogue PABX
- No configuration needed for the original PABX
- Therefore we still get same analogue telephone service as before

4.3 Network requirement

Before we design the new network, we need to identify the requirements.

Hardware requirement

Following is a list of devices we need for the IPX-2000 network.

- IP PBX

  IPX-2000
- IP phones

  Different brands of IP phones that supports SIP 2.0 can be used
- Analogue telephones

  Normal analogue telephones

  Connected to the IP PBX through adapters
- Telephone adapter

  Used to connect analogue telephone to the IP PBX
- Switch for telephones

  Used to connect the analogue and IP phones to the IP PBX

Cabling

- Network cables

  Used for network devices connection

  Need to identify the type of cables (cross over or straight, cat5 or cat3 etc)

  Normally Cat 5 100Base-T cables are used
- Telephone cables

  Used for telephone or PSTN connection
- Connectors

  Need to identify the connectors used for network cables and telephone cables

  Normally RJ-45 for network cables and RJ-11 for telephone cables
- Splitters

Trunk lines are connected to both analogue PABX and IP PBX

Need splitters to separate lines for IP PBX connection

- Location and layout

Identify how and where hardware is located

Identify how the cables are installed in the office

Identify the length and the amount of cables we need

4.4 Network topology

After we had identified the requirements, we drew up a network topology graph including the current network and the new VOIP system.

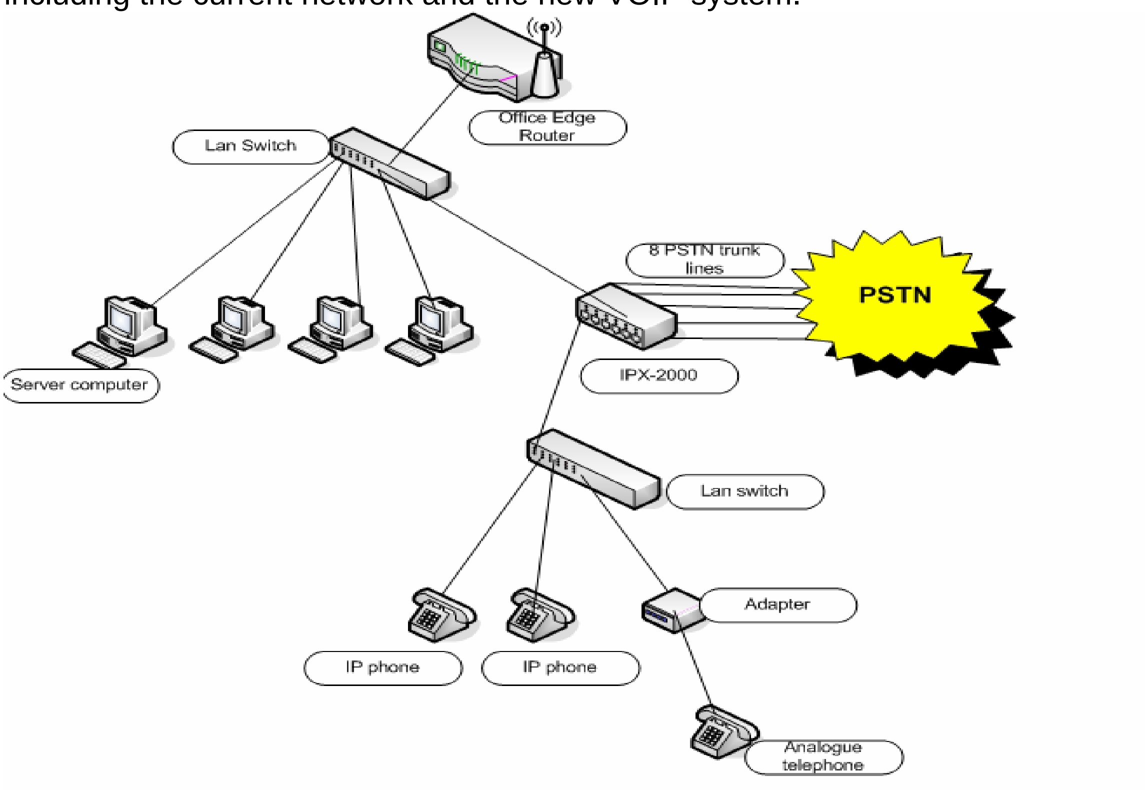

Figure 9 Network topology graph

- IPX-2000 is connected to the office LAN switch through the WAN port
- 8 PSTN trunk lines are connected to the 8 analogue FXO ports
- Another LAN switch is connected to IPX-2000 used for telephone connection

- IP phones are connected to the telephone LAN switch
- Analogue phones are connected to the telephone switch through telephone adapters

Note:

- Standard telephone can be connected to the analogue telephone adapter (Planet VIP-156) to make VOIP calls.
- Wireless connection is not an ideal way for VOIP communication because it raises a large amount of jitter problems. Replacing the wireless connection with a solid line connection will be a better solution.
- Purchasing a Frame Relay service from the ISP is another solution to reduce the delay and jitter problem and improve the voice quality
- Originally PSTN trunk lines are connected to the analogue PABX, we need some splitters to split those lines so that we can connect them also to IPX-2000
- This separate network requires new cabling structure

Figure 10 Telephone line splitter

5. Implementation issues

Before the real installation and implementation of the new system, we studied several implementation concerns.

5.1 Loading on Internet

The load generated by the new voice system on the internet is important since the company's bandwidth is limited. We need to make sure that the company has enough bandwidth for the staff to use data network and make internet telephone communications. So we need to understand what the bandwidth requirement for the voice communication is.

Basically, the amount of bandwidth required to carry voice over an IP network is dependent upon a number of factors. Among the most important are:

- Voice CODEC and Sample period
- IP header
- Transmission medium

Voice CODEC

- We have already learned that codec is used to transfer the analogue voice signal into digital stream
- The codec determines the actual amount of bandwidth that the voice data will occupy.
- Codec also determines the rate at which the voice is sampled.
- Two characteristics of codec are:
  The number of bits produced per second
  The sample period – how often the samples are transmitted

IP header

- The term 'IP header' is used to refer to the combined IP, UDP and RTP information placed in the packet.
- RTP is the first, or innermost, layer added. This is 12 octets.
- UDP adds 8 octets, and routes the data to the correct destination port.

- IP adds 20 octets, and is responsible for delivering the data to the destination host.
- In total, the IP/UDP/RTP headers add a fixed 40 octets to the payload.

Transmission medium

- Transmission of IP over other mediums will result in different overhead calculations

Suppose we are using G.711 codec

- G.711 samples at 20ms
- This generates 50 frames of data per second. G.711 transmits 64,000 bits per second so each frame will contain 64,000 ÷ 50 = 1,280 bits or 160 octets.
- The IP header adds 40 octets. This means 200 octets, or 1,600 bits sent 50 times a second
- The above calculation results to 80,000 bits per second. This is the bandwidth needed to transport the Voice over IP only. It does not take into account the physical transmission medium.
- Fixed IP overhead 40 octets, fixed Ethernet overhead 38 octets.
- In summary we can have: Bandwidth required is (160 + 40 + 38) x 50 x 8 = 95,200 bps

As we can see, the bandwidth required for a G.711 call is about 95.2kbps

Therefore, we need to be able to allocate about 96 kbps bandwidth for a single VOIP call through the PSTN (PSTN uses G.711)

5.2 Bandwidth control

So how do we control the bandwidth usage of the VOIP system?

Concurrent calls

- Suppose the upload limit is 512kbps
- Suppose a telephone call takes 90kbps bandwidth, then a maximum of 5 calls can be made at the same time through the internet
- However, there is only a 512-5*90=62kbps upload bandwidth left for the other network data
- It is better to reserve a certain amount of bandwidth for the VOIP communication purpose.
- Suppose we only allow 2 people make VOIP call at the same time, so we can reserve at least 180kbps bandwidth for VOIP calls
- Therefore, we firstly need to work out the bandwidth requirement for the data network without VOIP, then we can have an estimate of bandwidth for VOIP communication and work out the number of concurrent calls we can have.
- Note that the PSTN calls does not take any internet bandwidth

Bandwidth allocation

- Data usage

  Work out the average and the peak usage of the bandwidth for the data network

  Work out the percentage of usage of each application

  Bandwidth usage can be recorded using bandwidth monitor or management tools
- Voice usage

  Work out the average bandwidth left for VOIP communication. Based on this bandwidth:

  For a certain CODEC, work out the bandwidth requirement for a single VOIP call

  Estimate a number of concurrent calls we could have

Work out the total bandwidth needed

- Allocate bandwidth

  IPX-2000's QoS service

  Buy the bandwidth management equipment (Optional)

Bandwidth control using bandwidth manager

Bandwidth Manager is a product of Planet that can is used for bandwidth management of the VOIP system. By deploying Bandwidth Manager, the network administrator can allocate bandwidth based on business priorities by giving VOIP traffic guarantee bandwidth and/or a higher priority. [11]

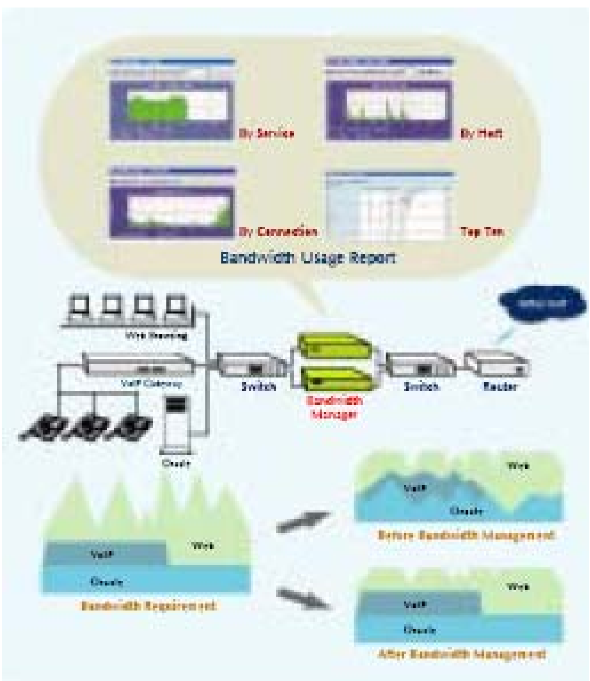

Figure 11 VOIP network with Bandwidth Manager

6. Installation

Finally we came to the installation stage. This part will be used as guidance in the real installation process. We firstly had a review of installation requirements. After that, we listed sets of possible installation steps.

6.1 Installation requirement

Original PABX

- As we stated in the previous session, the installation of the new system is independent of the analogue PABX.
- The only change to the analogue system is that the original 8 trunk lines from the PABX are needed to be connected to the PSTN through splitters so that the VOIP gateway can also be connected to the PSTN
- Analogue PABX will be used during the installation process
- Analogue PABX can be placed at the original place
- Cabling and the infrastructure of the analogue PABX does not need to be changed
- Keep the original PABX during the transition period. It will not only serves as a backup for the VOIP system but is also used for the comparison of two system

Main devices

- 20-port Ethernet LAN switch for the VOIP phones
- VOIP router (IPX-2000)
- n x Analogue telephone adapter (optional)
- 2 x Planet bandwidth manager (optional)

New Cabling

- New cabling is required for the separate network as stated in session 8
- If we assume we have 16 staffs, we need around:
- 16 wall plates with jacks
- 16 network cables (100BaseTX, straight through, RJ-45) to connect the wall jacks to the new network switch
- 16 extra network cables to connect VOIP phones to those wall jacks
- 1 network cable to connect the switch to the VOIP gateway (IPX-2000)
- 1 network cable to connect the gateway to the original network switch
- 16 telephone lines (RJ-11) for PSTN trunk connection
- 8 telephone line splitters

6.2 Hardware Installation

IPX-2000

- Connect IPX-2000 to the office LAN switch through the WAN port
- Select routing protocol in IPX-2000
- Add the route of IPX-2000 to the office edge router
- Connect 8 PSTN trunk lines to IPX-2000
- Require disconnection of PSTN trunk to the analogue PABX
- Connect PSTN trunk lines to both analogue PABX and IPX-2000 through splitters
- Setup IPX-2000 according to user manual section 3
- Connect a LAN switch to IPX-2000

Cabling according to the cabling design

- Install new wall jacks and wall plates if needed
- Connect existing or new wall jacks to the IP phone switch, wiring depends on the cabling design

IP-Phones

- Connect 1 IP phone to wall jack for IP phone
- Add user to IPX-2000 corresponding to the IP phone
- Assign an IP address to the IP phone, do basic setup according to the product user manual
- Modify information in IPX-2000
- Do the similar steps for the other IP phones.
- If telephone adapter is used instead of the IP-phone, connect the adapter to the switch
- Connect phone to adapter
- Assign IP address and setup the adapter
- Modify information in IPX-2000

6.3 Network configuration

- Static IP
  Assign a static IP address to IPX-2000 that can be seen from the outside
- Routing protocol
  Setup routing protocol and routes for IPX-2000
  Refer to IPX-2000 user manual section 2.2 and 2.3
  Add new route of IPX-2000 to the office edge router
- Allocate IP addresses for IP phones and devices in the new LAN
  IPX-2000 uses static LAN routing, therefore sub-netting and classless addresses can be used
  Assign IP address one by one to the devices
- Other configuration
  Time zone
  DynDNS
  QoS

- Refer to IPX-2000 user manual section 2 and 3 for more configuration detail

6.4 IPX-2000 service configuration

- Figure 12 shows the IPX-2000 basic configuration steps. Configuration detail can be found in IPX-2000 user manual section 2 – section 3.17

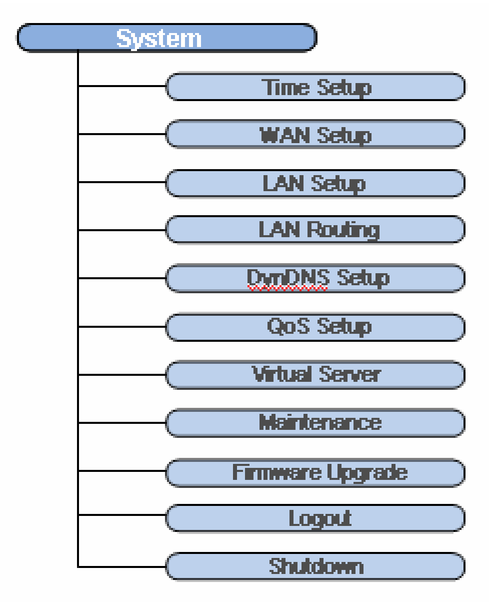

Figure 12 IPX-2000 system configurations steps

- Perform the advanced feature setup following the instructions in IPX-2000 user manual section 3.17
- Setup IP phones according to VIP-103PT/VIP-153PT/VIP-153T user manual
- Setup telephone adapter according to VIP-156 user manual

7. Pilot and test

This section listed a set tests needed to be done after the system installation which is used as guidance to the real test process.

- Connectivity test

  Verify IPX-2000 to office edge router connectivity

  Verify IPX-2000 to PSTN connectivity

  Verify connectivity between IPX-2000 to each of the wall jacks

  Verify IP phone to IPX-2000 connectivity
- Network configuration testing

  Test basic routing between office edge router and IPX-2000

  Test routing between office edge router IP phone
- Basic call testing

  Test in-site calls

  Test call to PSTN

  Test call to other VOIP site (If possible)
- PBX Feature testing

  Basic feature test (call forward, extension control)

  Advanced feature test (auto attendant, music on hold, meet-me conference)

  Test IP phone features
- QoS testing

  Test QoS for inner call and outgoing call

  Load test for single call, multi calls, voice conference

  Test traffic generation and effects on data network

  Test IPX-2000 bandwidth reservation

  Test call congestion

8. Project assessment and conclusion

This assessment is a summary of the opinions of activities which have been done during the project. During the development of the project, project manager organized meeting regularly to check the progresses that the students have made. During the meeting students gave opinions, shared ideas and comments and gained information from the project supervisor, and the other students.

In the first semester, we got some understanding of basic idea about the VOIP technology, we studied and compared some solutions of building a telephony system, and finally we came up with a solution with and a big picture of how system should work. At the beginning of this semester, we studied some concerns and issues related to the VOIP system we are going to build. After that, we researched and studied some relevant technologies, standards and protocols in the purpose of gaining some knowledge which may help us to design the system. We started the design phase of the project by listing sets of hardware requirements. We then completed a network topology graph based on the hardware requirements. After that, we started to think about the actual installation for the system. Based on the network topology that had been worked out, we listed sets of steps including the hardware installation and software configuration. Finally, we worked out a brief test plan associated to the installation. The project manager and I both agreed that the actual installation stage will take a lot of time. Therefore it will be stretched to November or December and conducted by the company's staff.

In conclusion, this is full year project had helped me gain not only so much new knowledge, but also lots of research and communication skills. Being a real life industry project, it had also given me a real good experience. At the same time, it also helped me understand there are still some skills that I need to improve, such as the time management, information gathering and filtering, and presentation skill.

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