The University of Auckland
NEW ZEALAND

v1.0

Voice Over IP

BTech Project

Semester 1 Report

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Abstract

Voice over Internet Protocol (VOIP) is a technology to transmit voice conversations over a data network using the Internet Protocol. We can simply call it internet telephone. Therefore, if we have got an Internet connection, we will be able to use the same connection to place the long distance free telephone calls. This process works by using already available VOIP software or devices to make phone calls over the Internet, essentially circumventing phone companies and their service charges.

VOIP is not a new technology. Actually, it is quite mature. There are a lot of companies around the world who are providing VOIP services now. Many businesses now have owned VOIP devices or are currently using the VOIP as their main telephone systems. This VOIP project requires us to design and setup a VOIP system in Compucon House, a median size company in New Zealand. An IP PABX system is required to be built which integrates voice and the existing data network to provide a better service to the company.

This report is a summary of work done during the first part of the project. It will then become a guide for the work in the next period. The first part of the report introduces the basic technology and concept involved in this project. The second part lists some requirements and restrictions for the project. After that, we will take a look at how we the system should look like and how we are going to build the system. Then we come to a conclusion to this report.

1. Project Brief

This section briefly introduces the definition and the scope of the project which may lead us through the research process and then help us understand the things we are going to develop.

1.1 Project description

This 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, system is required to be installed and tested in the second part.

1.2 Project scope

Our main task during the first part of this project is to research on the VOIP topic, get familiar with the technology and so as to have a basic understanding of the system we are going to build. The following is a list of the research scope that may guide us through the research process. However the research will not be limited to these, we may need more information in order to understand and build the system.

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.

2. Project Deliverable

The deliverable of the project includes planning, installing and testing and the commissioning documentation.

In the first part (Semester 1) of the project, we are required to plan the project and make research on the VOIP technology and get an idea of the overall structure of the system. In the second part (Semester 2) of the project, we are required to design and install the VOIP system in Compucon House.

The final deliverable will be a VOIP system that can work parallel with the existing PBX system using the devices provided. To be more detailed, Planet IPX-2000 should be used as a main device for the VOIP that connects the IP phones to the internet and ideally to the PSTN. Eventually, the company can reduce some cost from the use of original telephone system.

3. VOIP Introduction

The purpose of this section is to give a brief description of the technology and concept used in this project. We will take a look at some related examples as well in this section.

3.1 What is VOIP

The online definition for VOIP from Wikipedia is:
Voice over Internet Protocol (also called VoIP, IP Telephony, Internet telephony, and Broadband Phone) is the routing of voice conversations over the Internet or through any other IP-based network. The voice data flows over a general-purpose packet-switched network, instead of dedicated legacy circuit-switched telephony transmission lines. [1]

3.2 How is it used

As we can see, the voice data will travel through the internet work rather than the telephone lines. Therefore, if we have an internet connection, we should be able to use the VOIP phone. 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 the most common way of using VoIP. This is a way that enables you to connect your telephone to the internet work by connecting it to your computer. An adapter is required for the in this process. This adapter turns the analog signals into digital format and sends it out to the internet. It is very simple to setup a system like this. An adapter is connected between a telephone which should be supported by the adapter and the computer which should have the VOIP software installed. The computer should also be connected to the internet. Then you are able to make telephone calls.

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. Therefore, instead of connecting the phone to the wall connector, we need to connect the IP phone to the router or other network devices so as to connect it to the internet. Here we do not have to connect the IP phone to the computer, which means we do not need to install any software for the VOIP system. As the development of the wireless phone, the IP wireless phone will be available in the near future which will be another exciting technology.

Computer-to-computer call is a simple and cheap way to make VOIP calls. These calls are entirely free, the only thing you need is to install the software which can be found for free on the internet. If you have an internet connection, a microphone or a speaker, you can have make VOIP phone calls with the other one who is also using the software anywhere in the world.

3.3 What are the advantages

Cost:

The first advantage resulted from the use VOIP is probably the reduction of cost. In general, phone service via VOIP is free or costs less than the traditional PSTN. The integration of data and voice saves cost spent on using and maintaining two different networks. If users have existing under-utilized network capacity, they can use VOIP at no additional cost. VOIP to VOIP phone calls on any provider are typically free, while VOIP to PSTN calls generally requires the VoIP user to pay a little money. Free VOIP to PSTN services are rare. A notable provider is VOIP User. [2]

Functionality:

VOIP facilitates some features that are difficult to achieve for the original telephone network.

- VOIP uses such phones that can be seen as a network device, and it is identified by the network address (IP address). Therefore, no matter where you are around the world, the phone call should be able to reach you without problem.
- Free VOIP phone services are provided by some organization in some countries.
- VOIP phone are integrated with other services available on the internet.

Mobility:

VoIP allows users to travel anywhere in the world and still make and receive phone calls.

- Phone numbers are no longer restricted by the locations. VOIP calls are automatically routed to the destination.
- Users of Instant Messenger based VoIP services like Skype, Gizmo Project or Yahoo! Messenger can also travel anywhere in the world and make and receive phone calls.
- Video conversation, message data or file exchange are in parallel with audio conferencing and information passing.

3.4 What do we need to consider

Voice Quality:

The most important thing related with a telephone call system is the voice quality. 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?”

Availability:

Availability is the proportion of time that a service is available for use. [3] In telephone system, availability means the probability that a call can be established successfully on first attempt, excluding user factors like people busy or no pick-up. [4] We may also use the following formula to represent the availability:

$$\text{Availability} = \frac{\#\text{ of successful calls}}{\#\text{ of first call attempts}}$$

According to the research done by Wenyu Jiang and Henning Schulzrinne from Columbia University, VoIP has a 98% (about 7.3 days unavailable per year) net availability, which is still some steps away from what the PSTN offers today (three to four 9’s), but already comparable to the availability of mobile telephone networks (around 97% to 99%). This leads us to think about how we can increase the availability of VOIP phone calls.

Compatibility:

Compatibility is another important issue we have to think about. We will often ask 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.

3.5 Example of VOIP

There are a lot of VOIP providers in the world. The most famous one is Skype. Skype is a free VOIP solution. It is software that allows user to make telephone calls over the internet.

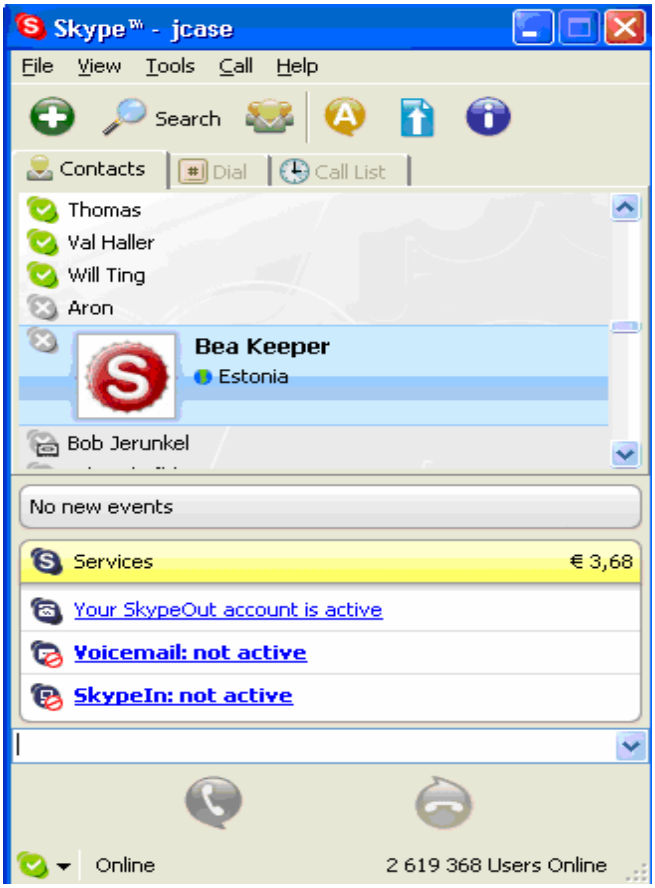


Figure 1 Skype interface

Skype provides several ways for telephone communication. The skype-to-skype call is entirely free. Setup the software in your computer, connect yourself to the other person in the other side of the world who is also using Skype, you are now able to talk to the other person using your speaker. However, you need to buy some credits if you want to make a phone call from Skype to a landline telephone, but it is mucher cheaper than the PSTN system if you are making a long distance call.

3.6 Future of VOIP

Analyses show that there will be at least 10 years before we can switch fully from the original telephone system to VOIP system. Telecom says it will replace the analog phone system entirely with the IP based phone system by the year 2010 which is around 4 years time. And also, a report by the Forrester Research Group predicts that by the end of 2006, nearly 5 million U.S. households will be using VOIP phone service. [5] So as we can see, the future of VOIP is very solid and will be a definite solution after the analog telephony.

4. Possible Solutions

This section introduces several possible ways of setting up a telephone system for Compucon House. We will compare them and try to find a best solution for the company.

4.1 Current PSTN

One possible way is to stay with the original telephone system. Figure 2 shows the Telecom international call rate for businesses. Compucon House has branches in Australia, Canada and Hong Kong. The cost to make a call to Australia and Canada is around 50 cents per minute, and is about $1.30 per minutes to Hong Kong.

Most frequently called countries	Peak 8am - 6pm Mon - Fri	Off-peak 6pm - 8am Mon - Fri, weekends, & Stat Hols.
Australia	$0.49	$0.42
Canada	$0.52	$0.48
Ireland	$0.52	$0.48
UK	$0.52	$0.48
USA	$0.52	$0.48
Denmark	$1.15	$0.88
Finland	$1.15	$0.88
France	$1.15	$0.88
Germany	$1.15	$0.88
Italy	$1.15	$0.88
Netherlands	$1.15	$0.88
Norway	$1.15	$0.88
Spain	$1.15	$0.88
Switzerland	$1.15	$0.88
Argentina	$2.07	$1.67
China	$1.77	$1.30
Cook Islands	$1.35	$0.79
Croatia	$1.88	$1.57
Czech Republic	$1.88	$1.37
Fiji	$1.35	$0.79
Greece	$1.65	$1.19
Hong Kong	$1.29	$0.79
Hungary	$1.88	$0.92
Indonesia	$1.75	$1.37
Japan	$1.26	$0.70
Malaysia	$1.68	$0.88
Mexico	$1.65	$1.19
New Caledonia	$1.35	$0.79
Papua New Guinea	$1.35	$0.79
Philippines	$1.73	$0.88
Poland	$1.88	$1.37
Russia	$1.88	$1.57
Singapore	$1.26	$0.74
South Africa	$1.88	$0.92
South Korea	$1.68	$0.88
Taiwan	$1.30	$0.88
Thailand	$1.75	$1.32
Turkey	$1.65	$1.19
Vietnam	$2.07	$1.67
Western Samoa	$1.35	$0.79

Figure 2 Telecom international call rate

4.2 VOIP service provider

The second way is to purchase services from third party that provides VOIP services.

Skype

Skype immediately comes into our mind.

1. Skype-to-skype calls are entirely free at all time.

2. Skype-In: With SkypeIn, you can get your own, regular phone number. So if your friends who aren't using Skype want to call you by dialing a regular number, you can still receive the call in Skype. No matter where you are.

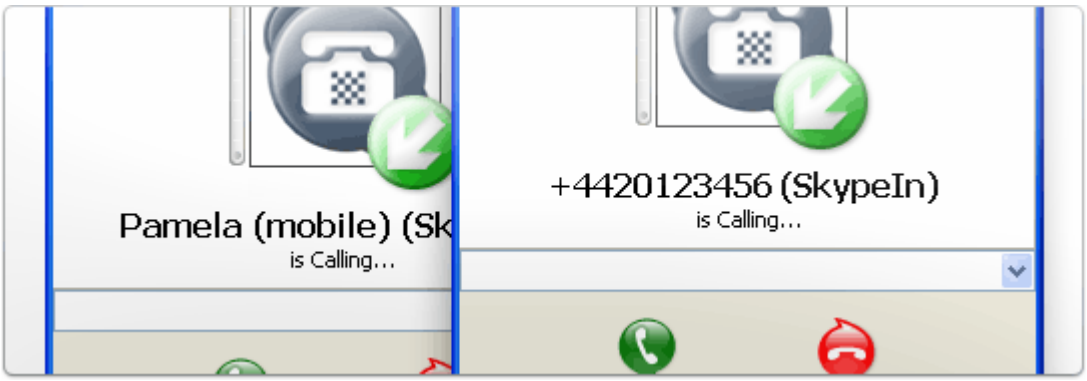


Figure 3 Skype-In

3. Skype-Out: A low cost way to make calls from Skype to friends who still use those traditional landlines or mobile phones. That means calling anyone, anywhere in the world at local rates.

Skype-Out is the only thing that requires the some charges.

€0.017/min≈$0.034(3c) to most popular destination

Sipserve

Sipserve is a New Zealand company that provides IP telephone services.

Sipserve offers the following services:

- Free SIP to SIP calling between any two devices connected to the Sipserve network
- Free SIP to PSTN calling to any free calling number in the USA (i.e 1 800) and soon to NZ (0800 and 0508)
- Very low cost SIP to PSTN calling anywhere in the world
- Real Auckland numbers mapped to your SIP device or softphone $10+gst per month with unlimited in bound calls.

Figure 4 shows the prepay rate for calls to different places

Destination	Rate NZD
Domestic	
New Zealand - Auckland	0.05
New Zealand - National	0.10
New Zealand - Mobile	0.45
International	
Australia	0.12
Australia - Mobile	0.50
China	0.17
Germany	0.12
Germany - Mobile	0.50
Hong Kong	0.12
Poland	0.18
Poland - Mobile	0.55
Thailand	0.32
UK	0.10
UK - Mobile	0.50
USA	0.10

Figure 4 Sipserve prepay rate

iTalk

iTALK is a VOIP plan offered by Slingshot New Zealand.

iTALK is an innovative new service that allows any broadband user to make and receive calls over the internet at heavily discounted rates. Making calls with iTALK is as easy as using your normal phone, plus it also comes with a range of additional free features to save you even more. [6]

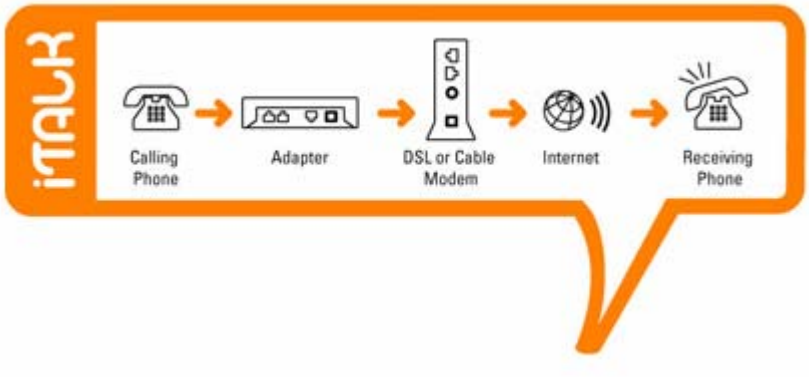


Figure 5 iTalk

With iTALK, you can either connect your telephone to your broadband connection using the iTALK Voice Box (coming soon) or use a special VoIP phone ($99.95). When making calls using iTALK you simply pick up the phone, and use it just like you do today. You can be up and running within minutes of signing up.

iTalk provides the following services:

- 5c per minute anywhere in New Zealand
- 5c per minute to over 40 international destinations
- 30c per minute to NZ mobiles
- Unlimited free calling to other iTALK users
- Free Voicemail and other features such as Call Waiting
- A second phone number or phone line

And the monthly charge is $9.95

4.3 Hardware Support

It is also possible for a company to build the VOIP system without any service providers but using some existing devices.

Cisco VOIP routers

Figure 6 shows the basic structure of VOIP network using VOIP router

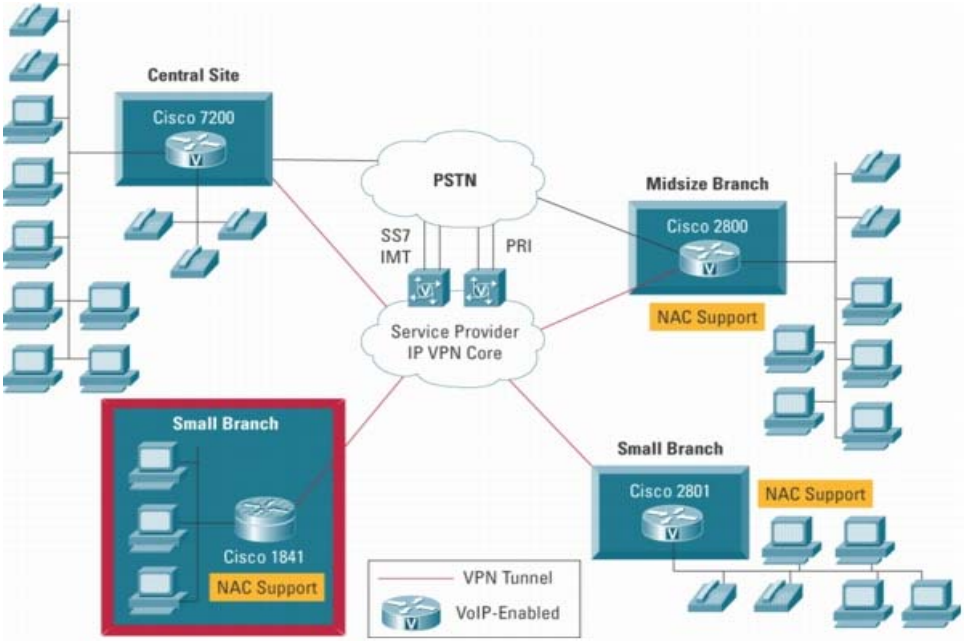


Figure 6 Cisco VOIP network

As we can see, the Cisco 2800 Series can meet the IP Communications needs of small-to-medium sized business and enterprise branch offices while concurrently delivering an industry-leading level of security within a single routing platform.

According to the research done by Qiao Ma last year, the Cisco 2800 Series architecture has been designed specifically to meet the expanding requirements of enterprise branch offices and small-to-medium-sized businesses for today's and future applications. The Cisco 2800 Series provides the broadest range of connectivity options in the industry combined with leading-edge availability and reliability features. In addition, Cisco IOS Software provides support for a complete suite of transport protocols, Quality-of-Service (QoS) tools, and advanced security and voice applications.

Cisco 2800 also provides a feature-rich telephony system, such as IP phone support, integrated call processing, integrated voice mail and broad range of voice interface etc. Please refer to Qiao Ma’s project report done in the last year for more detail.

Planet IPX-2000

PLANET IPX-2000 is the next generation voice communication platform for the small to medium enterprise. Designed as an open, scalable, and highly reliable telephony solution, the IPX-2000 is able to accept 200 extension registrations, and effectively scales from under 100 users to as many as 200 in a standard rack-mountable unit. The PLANET IPX-2000 is also designed to operate on a variety of VOIP applications. It provides centralized call control, auto-attendant, voice conferencing, PSTN access and digital and IP-based communications.

The IPX-2000 integrates up to 8 calls via the IPX- FXO (Foreign eXchange Office, FXO) module to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints. Moreover, the IPX-2000 also integrates telephony call processing, call control, voice mail, and a widely PBX application programming interface into a highly scalable architecture designed to support both traditional circuit-based and the Internet telephony service within a distributed enterprise communications network.

With the IPX-2000, standard SIP phones can be easily integrated in your office, plus the auto-config feature, you may integrate our IP phone VIP-153T/VIP-153PT, and the analog telephone adapter - VIP-156 to build up the VoIP network deployment in minutes.

5. Our Choice

Let us review the existing options.

Stay with the original telephone system:

- High cost
- Old technology

VOIP service provider

- Third party involved
- Extra fees
- Using other's technology
- May provide relatively high QoS

Hardware support

- Company's own VOIP system
- No third party involved
- An exploitation of new technology
- Relatively low cost
- May have restriction on the way of calling
- May face technical problem

5.1 Why IPX-2000

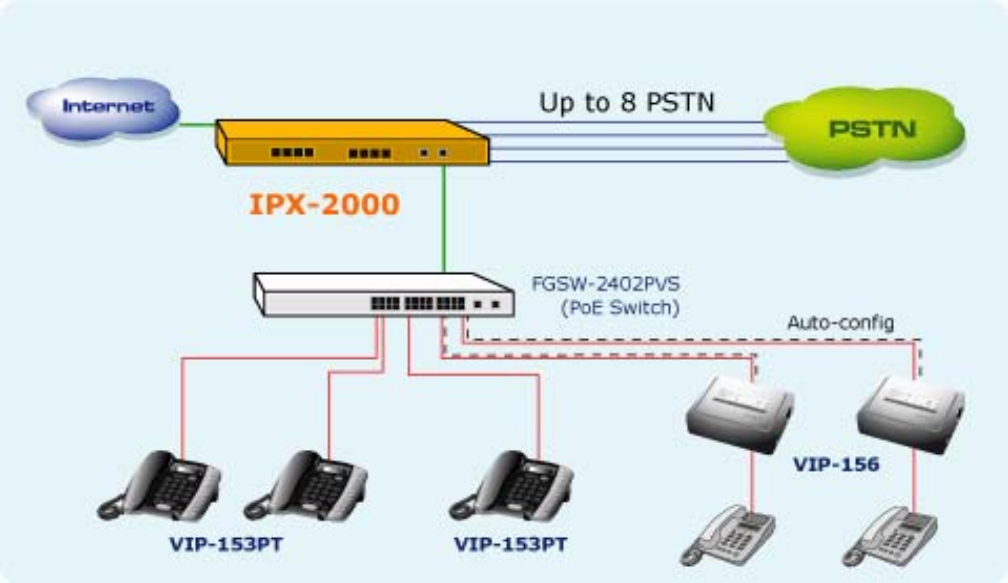
Being a technology based company, Compucon House decides to use IPX-2000 to build the company's own VOIP system. There are several reasons for that:

- Telecom will replace the analog phone system with internet phone system by 2010. It is not worthwhile to buy VOIP services from other companies.
- The 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.

- IPX-2000 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.

5.2 The big picture



As we have mentioned in the project deliverable, the final VOIP system we are building is independent from the original analog PBX system. It works parallel with the original PBX system. Therefore, we do not need to consider the structure of the original PBX system when we design the new system.

5.3 Ways of communication

First thing we need to consider when we build an IPBX telephone system is the communication flexibility between 4 components involved in this system, that are formed by analog phone, IP phone with PSTN and internet.

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 have the following table to represents the communication for the new telephone system in the Compucon House.

	Internet	PSTN
Call	IP Phone Analog Phone(With adapter)	IP Phone Analog Phone
Receive Call	IP Phone Analog Phone(With adapter)	No available

Table 1 Communication availability

6. Project assessment

This project assessment is the summary of the opinions of activities which have been done during the first part of 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 part of the project, student got to understand the basic technology involved in this project and is able to draw a big picture of how the system works. During this period, student did some research on the definition, usability, availability, and consideration on the VOIP topics. After that, student studied and compared different solutions that may be suitable for the company, which leads to a final solution that will then be implemented during the second part of the project. Student has also drawn out a draft picture of the final system, and listed out some consideration and important issues that must be taken care off during the next part.

Project is processed by the student followed the instruction of the project supervisor. And an initial project plan also helps the student in many ways to

achieve the final deliverable. However, the project plan needs to be updated as the project goes.

7. Conclusion

This report is basically a summary of work done during the first part of the project which will then become a guide for the next part of the project. The project definition and scope is defined in the first part of the report, which is followed by the deliverable of the project. The following sections introduce the VOIP technologies together with some examples. Possible solutions for the company are given and compared in the following section in the report. And at last, some ideas of the final solution are given.

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