The Feasibility study of IP telephony system replacement for AIA Thailand

By

Supawan Jantacha

Submitted in Partial Fulfillment of the Requirement for the Degree of

Master of Science

in Technology Management
Assumption University

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Master Project Approval

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Academic Year	2/2003

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ACKNOWLEDGEMENTS

I would like to take this opportunity to thank several people who have helped to make this project completed.

I express the most sincere appreciation and thanks to my advisor, A. Tanawat Ruangteprat, for his consistent support, guidance, and encouragement in the preparation and writing throughout the fulfillment of this project.

Furthermore, I would like to express appreciation to Dr. Soonthorn, Dr.Wisanut and Dr. Anusorn, all of my instructors in Technology Management Program for providing their knowledge and experiences through the course study as well as their valuable suggestion and helpful comment on the project. Also, Dr. Tibrat for her suggestion about writing project guidline.

I would like to thank Khun Surachai L., my boss for his understanding and assistance throughout this project, as well as Khun Chaiwat C. and Khun Sirisak B., for the Data Center and Network communication section, for providing the technological information and other related information to this project.

Finally, I wish to express the special appreciation to my family, my colleagues and friends for their understanding, support, encouragement and great sacrifices during the creation of this project.

ABSTRACT

This project is to study the feasibility to replace existing telephone communication system with IP telephony for AIA (Thailand).

The corporate strategy of AIA needs to improve customers', agents' and staffs' satisfaction on service to top quality, and to remain company as the forefront of technology. The focus on communication system – Network Services and Telephone Services is important to support for all their needs. The management has an expectation that the company's communication system will be more flexible, available, easy and manageable.

And nowadays, Voice over IP (VoIP) technology is quickly becoming a viable and acceptable technique for transporting voice. What was once thought to be an economical approach for incorporating this platform has drastically shifted.

The contribution to the project, the organization can be benefited in terms of reducing cost (calling cost), and quality of voice communication. It also enhances the employee efficiency.

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I. CHAPTER 1: INTRODUCTION

American International Assurance Company Limited, or AIA, is a life insurer in Southeast Asia and has been serving the region for more than 70 years. With its regional head office based in Hong Kong, AIA has branch offices and affiliated companies located in countries and jurisdictions including China, Hong Kong, Macau, Malaysia, Thailand, Singapore, Brunei, Guam, Indonesia, India, Sri Lanka, Taiwan, the Philippines, Japan, Korea, Vietnam, Australia and New Zealand.

AIA was founded in 1919 in Shanghai, China. At that time, the firm was called "International Assurance Company", or INTASCO. AIA in Thailand was established in 1938 and developed over a three-year period before the company's operations were temporarily suspended at the advent of South East Asia's involvement in World War II. At that time the Japanese authorities imposed restrictions on many companies, prohibiting them from engaging in business activities. AIA therefore suspended its operations. After the war, AIA head office was relocated from Shanghai to Hong Kong in 1947. The company recommenced its operations in Thailand in 1949.

In Thailand, AIA have 9 branch offices in Bangkok and 8 branch offices distributed around the country, i.e. Chiang Mai, Khon Kaen, Nakhon Ratchasima, Suphanburi, Chonburi, Surat Thani and Songkhla.

1.1 Background of the Project

As the corporate strategy of AIA needs to improve customers', agents' and staffs' satisfaction on service to top quality, priority and to remain company as the forefront of technology particularly in the area of customer service, one thing that is focused is the communication system – Network Services and Telephone Services, which are important gear behind technology that provide support for all their needs. The management section expects that the communication system will be more flexible, available, easy and manageable.

1.1.1 AIA's Telephone System Overview

All branch offices – 17 offices: 9 located in Bangkok and 8 located in other provinces – can call to each other via Private Branch Exchange (PBX) or router features with the main PBX at head office building linking to each branch office over Tie-lines or Leased-line, except 2 branch offices in Bangkok that need to dial out to Public Switches Telephone Network (PSTN). AIA has more than 1,000 telephone numbers, and around 10,000 extension numbers, over the country.

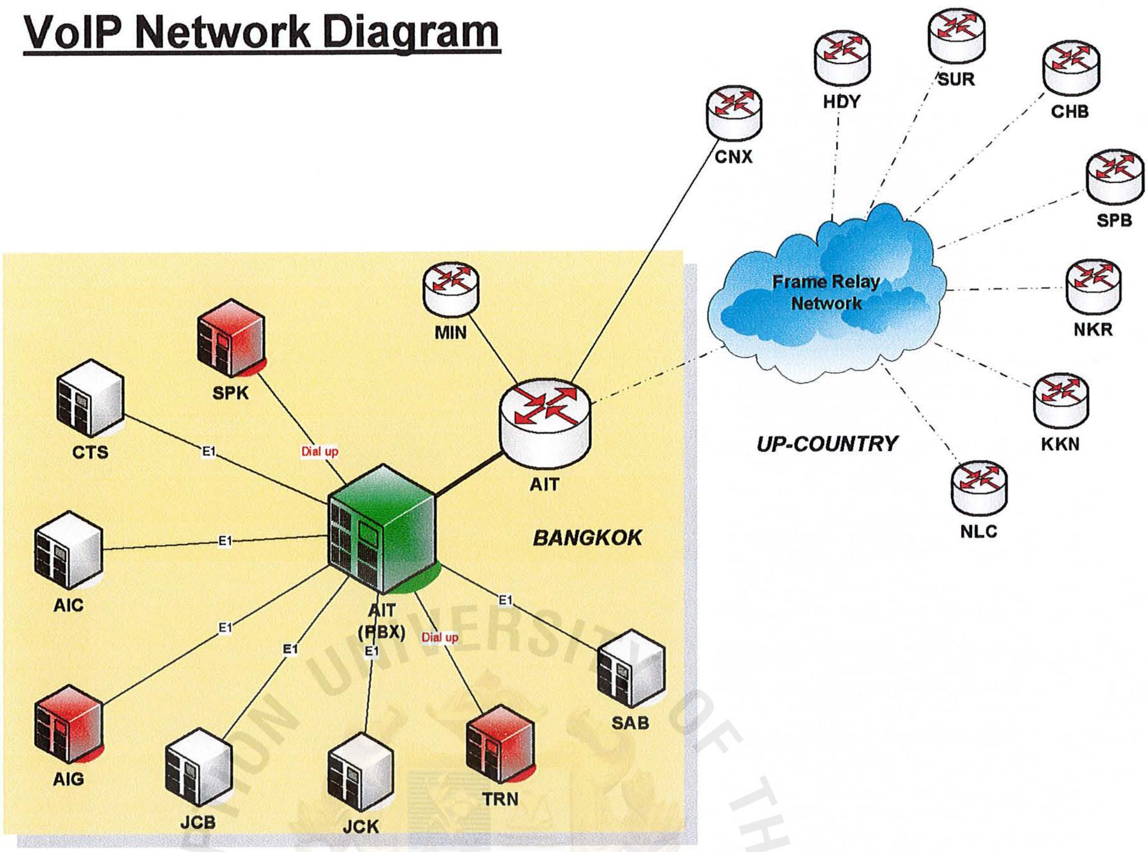


Figure 1-1 AIA's Telephone System Overview

1.1.2 Evolution of IP Technology

Over the past few years there has been incredible growth of next-generation telecommunication which is a testament to the maturity of VoIP technology. A majority of voice and fax traffic has moved from circuit switched networks to the Internet. Voice quality between a circuit switched call and an Internet based call is now indistinguishable – Many Tier One carriers are passing voice over the Internet without even realizing it.

VoIP has introduced vast cost savings; it has also introduced unprecedented opportunities in the telephony marketplace. The time has come for telecommunication to make the leap into the burgeoning VoIP marketplace. [http://www.connexions.inter change.co.uk/]

Internet Protocol (IP) Technology is a platform which is evolving and providing a foundation for incorporating other technologies in an intelligent and effective way. The technology allows transporting voice over a legacy PBX, thus incorporating a Converged or Hybrid environment as well as over the utilizing pure IP.

And nowadays, Voice over IP (VoIP) technology is quickly becoming a viable and acceptable technique for transporting voice. What was once thought to be an economical approach for incorporating this platform has drastically shifted.

Today, telephone systems are based upon the same or similar technology that is used for data processing systems. New service and cost saving options are often dependent upon current telephone system technology being in place. Consequently, there is a need to review these systems and to construct a strategic approach. There are many potential benefits to companies and the corporation from developing a uniform approach wherever possible.

Connecting PBX with a VoIP solution is a simple method for introducing VoIP to an organization. Many corporations currently use tie lines, connecting PBX to offices that are in different geographic locations. This method of toll bypass is used to escape long distance charge that would be applied had the calls between the offices crossed the PSTN. [Configuring Cisco Voice over IP, Second Edition, 2002; P.416]

1.2 Problems Statement

Telephone Service Department is responsible for planning, budgeting, managing, subject to supervisory review and approval; moreover, supporting all Business Divisions on telephone service matter is one of its responsibilities.

PBX is one of the most critical automated functions within a commercial organization and the most recent technology, when used together with other elements of the IT infrastructure, offers some of the greatest opportunities for graining competitive edge, improving service and reducing cost. But, most companies initially invested in legacy PBX installed at all branch buildings, some of which are more than ten years of age, which provided the latest telecommunication technology.

After a PBX survey it has been found that most of PBXs such as ROLM IBM 9751 PBX and OKI 1200 which are installed since 1993, become very obsolete and the company is holding a highly maintenance cost of them.

Today, PBXs are based upon the same or similar technology that is used for data processing systems. New service and cost saving options are often dependent upon current telephone system technology being in place. Consequently, there is a need to review these systems and to construct a strategic approach. There are many potential benefits to companies and the corporation from developing a uniform approach wherever possible.

To sum up, problems about the telephone service in American International Assurance Company Limited are as follows:

1. High cost of rental expense for communication.
2. Redundancy of wiring cable of voice and data communication.
3. Central (network) and some of network PBXs are not IP-PBX, not supporting to IP telephony.

1.3 Project Objectives

According to management strategy on Data Network, Telephone Service and widespread deployment of IP network trendiness, the company would like to exploit a new way to maximize investments by integrating voice and data network. Therefore, if the company would like to replace or implement any equipment of voice communication IP technology must be focused. This deals with replacing PBX with IP-PBX at head office. Thus, objectives of this project are as follows:

- To study the feasibility of replacement the existing PBX system infrastructures with IP-PBX.
- To design conceptual diagrams to implement the IP-PBX that meets the company requirement and is compatible with the existing infrastructure of telephone system.
- To study further major communicative applications in order to be transformed from “Voice-only” to “Converged” (Voice and Data).

1.4 Scope of Study

The scope of this project is to study the replacement of PBX with IP Telephony. Enterprises should look at tangible benefits rather than simple cost savings by understanding the needs of the business units.

This project is targeted at information gathering and a complete assessment of the telephone system to ensure that business is using feasible and scalable solutions today as well as being prepared for next steps.

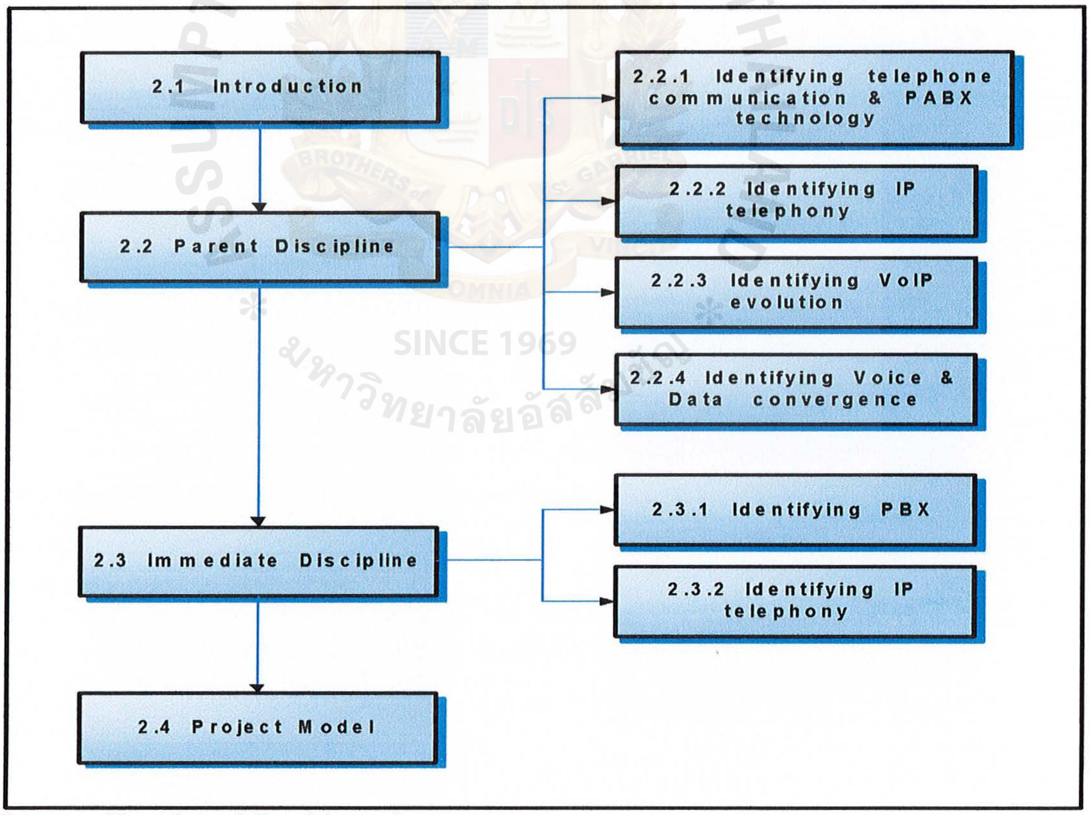
The assistant is gathering information by distributing a questionnaire to each of the participating vendors. Additional evidence and supplemental information are solicited from printed sources, textbooks, journals, articles, etc.

II. CHAPTER 2: LITERATURE REVIEW

2.1 Background of Literature Review

This chapter will review articles, texts, white papers, websites and similar sources of reference to provide an overview of current knowledge on both traditional telephone system and IP technology concept. Furthermore, the success and best practice of the replacement or implement of IP-PBX will be provided in this chapter in order to provide a clear picture based on the current situation.

Figure 2-1 Project roadmap



Source: Developed for this project

2.2 Parent Discipline

This project is organized as follows. First provide the reader with a basic technical understanding of Telephone communication and PBX technology. And then discuss about IP Telephony and Voice over IP technology evolution that can identify voice and data convergence.

2.2.1 Identifying telephone communication and PBX technology

This section will provide information about the history of telephone, the information related to land-line based phone systems and fixed telephony, and PBX.

History: Antonio Meucci who called it electrophones invented the telephone around 1860. The first recorded public demonstration of Meucci's invention took place in 1860, and there was a description of it published in New York's Italian language newspaper.

In 1861 Philipp Reis presented a machine for electronic voice transmission. Elisha Gray independently invented it and demonstrated it in 1874, but two hours before he submitted his patent announcement, Alexander Graham Bell submitted a patent (although his proposed design did not work). As a result, Alexander Graham Bell was usually credited with the invention.

The very early construction of the telephone was based on sound transportation through air rather than generated electric signals from speech. According to a letter in the Peking Gazette, in 968, the Chinese inventor Kung-Foo-Whing invented the thumtsein, which probably transported the speech through pipes. Even the early inventions made by Meucci et al transported the sound through pipes.

The modern handset came into existence when a Swedish lineman tied a microphone and earphone to a stick so he could keep a hand free. The folding portable phone was an intentional copy of the fictional futuristic communicators used in the television shows of Star Trek.

The history of additional inventions and improvements of the electrical telephone includes the carbon microphone (later replaced by the electrets microphone now used in almost all telephone transmitters), the manual switchboard, the rotary dial, the automatic telephone exchange, the computerized telephone switch, Touch Tone® dialing (DTMF), the digitization of sound using different coding techniques including pulse code modulation or PCM (which is also used for .WAV files and compact discs).

Newer systems include IP telephony, ISDN, DSL, cell phone (mobile) systems, digital cell phone systems, cordless telephones and the third generation cell phone systems that promise to allow high-speed packet data transfer.

The industry is divided into telephone equipment manufacturers and telephone network operators (Telcos). Operating companies often hold a national monopoly. In the United States, the Bell System was vertically integrated. It fully or partially owned the telephone companies that provided service to about 80% of the telephones in the country and also owned Western Electric, which manufactured or purchased virtually all the equipment and supplies used by the local telephone companies. The Bell System divested itself of the local telephone companies in 1984 in order to settle an antitrust suit brought against it by the United States Department of Justice. (Oxley, Michael G., 1993.)

Land-line based phone systems and fixed telephony: The network that connects most phones together is known as the PSTN, which stands for Public Switched Telephone Network.

Fixed phone lines are usually copper wire lines, which form a circuit between the subscriber and a subscriber-line interface. The SLI is nondescript street furniture, usually a box on the ground, or a silver can on a telephone pole. The SLI provides dial tone, and converts voice and dialed numbers to digital signals, which are sent on a few wires to the exchange. SLIs were invented so that central offices could be smaller, thus less expensive, while giving better service. Some recent installations may use optical fiber to connect an SLI to the exchange. Some old installations may connect subscribers directly to an exchange without using SLIs.

An analog phone's twisted pair is self-contained, self-testing and designed to fail safely. It typically modulates incoming and outgoing conversations on the same pair of wires, and biases the lines at 48 volts DC to power the telephone. The power is provided from multiply redundant power systems at the exchange- most phones will continue to work in a power-failure. Furthermore, the dial tone is presented only when the SLI's, and exchange's computer believe the network is up.

An analog line uses frequencies of 0-3.5 kHz, with frequencies higher than this filtered at the SLI before it is converted to digital samples. The analog speech signals are carried over the digital backbone network as a stream of digitally encoded samples at a sample rate of 8 kHz (8,000 numeric samples representing sound-pressure per second). The frequencies on the copper above 4 kHz can be utilized for DSL connections.

A line is a single voice communications circuit between the subscriber and the central switching office. A trunk is a single circuit between an SLI or central offices and may be analog or digital and is transmitted via copper, microwave, or fiber optics. A trunk group is a grouping of identical trunk circuits between two specific central offices.

Larger companies and organizations often employ a PABX (Private Automatic Branch Exchange). This is a telephone switch that defines its own local phone number range, which is commonly embedded in a public local phone number range. Some of the largest companies now even have their own internal telephone networks across the country, or even throughout the world, with limited gateways into the PSTN.

Most PSTN systems use analog communication between individual phones and the local switch. If digital communication is used for an individual phone, the system used is usually ISDN (Integrated Services Digital Network).

PBX: From Wikipedia, the free encyclopedia stated that Private Branch Exchange (also called PBX or Private Business Exchange) is a telephone-switching center that is owned by a private business, compared to one that is owned by the common carrier or

Telephone Company. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.

In industrial countries, most companies (with more than around 10 employees) need their employees to be able call each other, call outside phone numbers (the public telephone network or PSTN), and receive calls from outside.

For companies with multiple physical locations, PBXs are sometimes interconnected by so called trunk lines.

PBXs are distinguished from smaller "key systems" by the fact that external lines are not normally indicated and selectable from an individual extension. From a user's point of view selecting a line and dialing the external number make calls on a key system; calls on a PBX are made by dialing 9 (or 0 in some systems) then the external number.

Another alternative is to connect all the telephone sets to the PSTN, but the major disadvantages are that every extension requires its own line (usually with a monthly recurring line charge); also, to-"internal" calls would be dialed externally, and charged for.

Finally, most local phone companies offer Centrex or "Virtual PBX" service in which each extension has a trunk line connected to the telephone company's Central Office, where software on the CO switch enables PBX-like functionality.

Functionally, the PBX performs three main duties: first establishing connections (circuits) between the telephone sets of two users. Note that fax, modems and many communication devices can often be connected to the PBX (although the PBX may degrade line quality for modems). Therefore telephone sets are referred to as extensions. Second, maintaining such connections as long as the users require them. Last, providing information for the Accounting Department (e.g. metering calls). There are many PBX manufacturers. Some of the most common include Avaya (Lucent and AT&T), Siemens AG (including Rolm), NEC, Nortel, Toshiba, Fujitsu, Vodavi, Mitel, and Ericsson.

PBXs offer many capabilities. Normally each manufacturer may have a different name for each capability. Here is a short list of common capabilities:

- Direct Dialing (DDD or DDI), also called Direct Inward Dialing (DID)
- Customized Abbreviated dialing (Speed Dialing)
- Follow-me
- Call forwarding on absence
- Call forwarding on busy
- Call transfer
- Music on hold
- Automatic ring back
- Night service
- Call distribution (Automatic Call Distributor, fixed sequences, etc.)

The extension interface can be:

- Proprietary: the manufacture has defined a protocol. One can only connect the manufacturer's sets on the PBX.
- Standard Interfaces: any device supporting the standard can be connected. The most common digital standard for fixed devices is ISDN.

Potential links between PBX (Trunk lines) can also use proprietary protocols, but if several manufactures are on site, the use of a standard protocol is required. Most used standard protocols are QSIG and Digital Private Network Signalling System (DPNSS).

2.2.2 Identifying of IP Telephony

IP Telephony, also called Internet Telephony, IP Telephony is the technology that makes it possible to have a telephone conversation where the signal is carried over the Internet or a dedicated network in Internet Protocol (IP) packets, instead of over dedicated voice transmission lines. (Wikipedia, 2003) This allows the elimination of circuit switching and the associated waste of bandwidth. Instead, packet switching is used, where IP packets with voice data are sent over the network only when data needs to be sent, i.e. when a caller is talking.

Its advantages over traditional telephony include to lower costs per call, especially for long-distance calls and to lower infrastructure costs: once IP infrastructure is installed, no or little additional telephony infrastructure is needed.

The protocols used to carry the signal over the IP network are commonly referred to as Voice over IP, or VoIP protocols. VoIP protocols include the heavyweight H.323, which also provides videoconferencing and data capability, MGCP (Media Gateway Control Protocol), SGCP (Simple Gateway Control Protocol), and SIP (Session Initiation Protocol), a protocol to initiate VoIP connections.

Next, the following topics are technology overview, hardware overview, VoIP signaling standards, VoIP Constraints/Issues, transport technologies, product integration profile, and international presence and support.

Technology Overview: Due to the relative newness of VoIP technology the organization is opting to preserve current PBX investment and add a VoIP gateway that allows for conversion of analog/digital signals into that of the IP format. The legacy PBX continues to have an ongoing proven track record in providing extremely high reliability thus reducing the likelihood of telephony service failure to a minimum. Due to this fact the migration onto pure IP is taking a phased approach thus allowing for the preservation of circuit switched telephony.

In addition, depreciation of the existing equipment and feature functionality will pay a role as well as who will have ultimate responsibility for the platform. A converged network will require technical knowledge and cooperation that will encompass both the voice and data folks within an organization.

At this point in time it is essential to point out that there are in fact many ways to describe and/or refer to the adoption/deployment (i.e. VoIP, IP Telephony, Toll-Bypass, etc.) of an IP environment. The following are brief explanation of three primary methods in transporting voice when the IP technology is utilized.

1. Converged/Hybrid Environment:

This method looks to utilize a combination of TDM (Time Division Multiplex) and IP technology. Transmission and conversion are accomplished either through an externally attached gateway or internal to a PBX/switch.

As a technology this platform allows for the transmission of a voice call to be converted from either analog or digital signal into an IP packet thus being transported via the WAN/VPN network. In essence the call bypasses the PSTN thus allowing for cost saving due to utilization of the existing infrastructure. In addition this method provides the reuse of the existing telephony equipment.

2. Toll Bypass:

One of the oldest methods in transporting voice traffic over the IP/data network is through the utilization of a gateway. This method allows for any existing switch platform, the opportunity to bypass the PSTN or private lines and reutilize what is available over the WAN infrastructure. In essence, business units that have or will have a WAN connection, can utilize this methodology as a cost saving measure. For those areas/countries that do not have the rates associated with the US this method of transport has been well received within the AIG Corporation.

3. IP to IP:

A complete and true approach utilizes the IP network for the transport of voice traffic as well as IP end devices. This approach requires that an IP device (i.e. IP phone, Soft Phone) be utilize to initiate a call, which is then placed on the data network for transport. The call can be transported either over the public or private network (i.e. VPN/ISP, etc) thus having a direct effect on security and QoS.

VoIP is an emerging technology/platform that looks to provide immense flexibility over non-homogenous environments thus allowing for greater utilization of resources and applications.

Within this environment we find the most optimal locations as being those, which are referred to as a Greenfield. Greenfield, synonymous with newness, tends to offer the most benefit when it comes to implement a converged voice and data network.

The IP based technology allows and supports higher bandwidth applications thus providing for multimedia communications, such as desktop video, conferencing and wireless applications. In essence this technology is quickly becoming adapted as a major platform and standard for delivering voice applications.

The following are some of the key elements of concern when considering a VoIP application:

- Supported protocol (i.e. H.323, SIP)
- Supported power to the teleset/phone (i.e. IEEE 802.1af or IEEE 802.3af).
- Reducing Latency via RTP for Voice Traffic by upgrading to a faster voice encoder (i.e. 160 bits or 20 bytes of digitized information is transported in a 64 byte datagram)
- Firewalls and VoIP products purchased from multiple vendors thus causing a mixed environment to have issues in that streaming traffic (i.e. voice & video), blocked via the network address translation (NAT).
- Delayed delivery of packets (latency) and/or dropped packets should be avoided through the use of multiple queues due to buffer utilization.
- Optimizing switches and routers for VoIP traffic though the use and support of IEEE 802.1p/Q prioritization using classes of service, which permits multiple hardware queues/classes of service (i.e. utilization of Layer 2&3).
- Application needs to be based on an open-architectural design so as to provide maximum flexibility.
- Is the Internet Telephony Service Provider (ITSPs) subject to federal regulations for the considered site(s)?
- Will access charges be imposed on VoIP calls?
- How will the carriers guarantee performance (i.e. SLA) in the absence of standard?
- Security:
    - Performs a network risk assessment and identifies potential trouble spots such as remote users and unused ports.

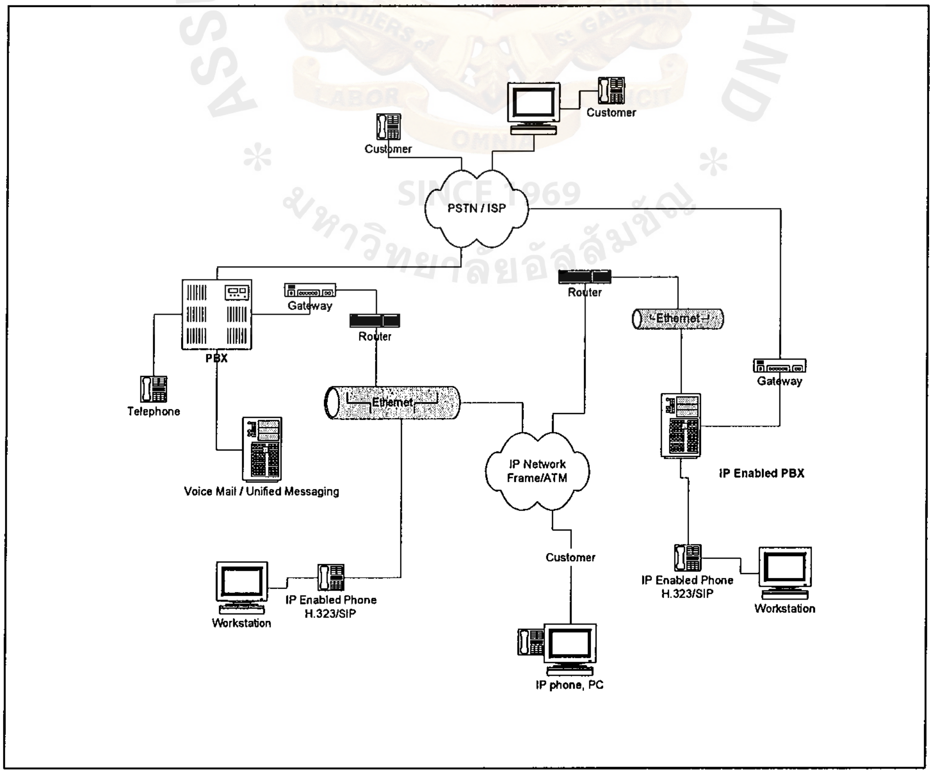
- Determines service required and disables all those not needed.
- Segments IP voice from IP data networks on a single network using technologies such as virtual local area networks (VLANs), access control and state firewalls.
- Uses encrypted tunnels or virtual private network (VPN) tunneling for remote locations.
    - Utilizes vendor's inherent encryption for voice related traffic.
- Disables all unused ports from the system.
- Turns off features that permit automatic registration of phones by end users
- Limits network administration access to a few individuals.
- Secures phones in public places with passwords and log-in numbers.
- Consistently updates system with latest security scanners.
- Monitors the network constantly for all suspicious traffic.

- Per minute rate charge offered by the carrier supersede that of additional bandwidth.
- Delivery of call control related information (i.e. via UDP & TCP)
- Operating system of platform
- Redundancy/resiliency
- Growth potential
- Support of TDD/TTY capability
- Maximum end devices supported including that associated with ACD agents
- DSP's supported for TDM to IP conversion
- Busy hour call completion (BHCC) of proposed platform

The following are some of the key benefits to keep in mind when considering a VoIP application:

- A VoIP platform allows for efficiency through centralized management in handing the complexities of an environment.
- Converged networks allow for the elimination of dual cabling thus reducing costs.
- Ability to share and support a multimedia environment (i.e. voice, video, fax, chat, etc.)
- Ability to share and support multiple applications (i.e. telecommuting, etc) and scalability.
- Elimination of costly point-to-point tie lines.

Figure 2-2 VoIP Configuration Diagram



Source: Adopted from The Lippis Enterprises, Inc., 2002

Hardware Overview: Hardware that will be described is server, gateway, gatekeeper, switching, and phone.

***Server*:** As the brains of the operation the server has a responsibility of maintaining the software configuration of the platform utilized. This can range from complex network routing algorithms to simplistic tasks such as allowing a station/phone the ability to conference.

***Gateway*:** An essential device for encoding calls that needs to be placed over an alternate path/transmission median for completion of a voice call is the gateway. This in essence converts a circuit switched call into IP packets (i.e. visa versa) for transporting over data facilities. In addition the gateway manages the QoS to ensure that voice traffic has priority so as to deliver toll quality service.

Gatekeeper: The function of this component can be found in the utilization and control of subnet bandwidth management when utilizing H.323 protocol. The primary task is to perform call addressing, administration control and locating other gateways. In addition the gatekeeper maintains a registry of devices with the network during startup and requests made.

Switching: This is utilized to connect your IP devices to and allows traffic to pass either over layer 2 or 3 in conjunction with the OSI model (Open System Interconnection). Layer 2 essentially provides discrete LAN segments to one or more users on each switched port. Layer 3 is designed to offer more complex routing capabilities and switching at the network layer. These devices (i.e. layer 3 switching) create virtual circuits for transmission of data while handing functions such as routing, addressing, error handing, congestion control, etc. all at wire speed. Noted, Cisco provides layer 2 switching.

Phone: This area/instrument is very much like that of the old legacy system in that they are expensive and proprietary for most parts. The phone can take on an actual physical appearance or be part of what is commonly referred to as a Soft phone. This does however exclude the analog devices being utilized thus providing an inexpensive universal alternative.

VoIP Signaling Standards: The lack of standard remains a major challenge for Voice over Internet Protocol, since multiple standards make it difficult to achieve interoperability among vendor products. Without a fully supported industry wide standard, many manufacturers have developed different and incompatible VoIP products that essentially make their products proprietary.

Currently, the four leading standards that support VoIP are H.323, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP) and H.248.

H.323 is the older standard that supports the majority of installed systems. Each protocol offers some merits but also has limitations. Due to current constraints, many vendors modify these protocols beyond the Internet Engineering Task Force (IETF) specifications, rendering them proprietary.

The following provide bit more details, with regards to strengths and weaknesses as it relates to each:

- ***H.323*** is an earlier protocol developed by the International Telecommunication Union (ITU) and is defined as a set of standards for sending voice and data communications over a packet-switched network. It was originally designed to support video and is considered bulky (i.e. resource intensive) and complex for supporting voice traffic. Due to its extensive overhead, it also causes problems with firewalls. Other limitations include lack of scalability and quality of service (QoS) guarantees.

- ***H.450*** is a standard and subset of H.323 allowing for generic, basic phone feature (i.e. transfer, hold, etc.) functionality. This standard (i.e. determined by the ITU) provides compatibility among disparate systems.

- ***SIP*** and SIP+ had their beginnings in the IETF and are gaining widespread adoption, which are primarily due to the inherent efficiencies. More scalable than H.323, they are considered more suitable for voice applications.

Another benefit is that SIP is considered easier to develop applications, since many programmers find it similar to HTML. Also, SIP is more flexible than H.323 and works in conjunction with Resource Reservation Protocol (RSVP), Real-Time Transport Protocol (RTP), STSP, Session Announcement Protocol (SAP) and Session Description Protocol today are the only provider offering/supporting the SIP protocol (i.e. WCOM, which utilizes SIP, AT&T utilizes H.323) to support their platform.

Carriers are frantically positioning themselves so as to provide sufficient capacity and capability required for handing such a service.

Various providers such as WorldCom have to enable the conversion from TDM to IP packets as well as transportation facilities for such traffic. The issue with this approach is found to be with tat of the transportation facilities for such traffic. The issue with this approach is found to be with that of the carriers who have not yet incorporated a true Hybrid switching platform that will accommodate the anticipated IP traffic.

Carriers have however started the transition so as to utilize the tremendous amount of fiber laid over the years in anticipation that all traffic (i.e. IP, analog, digital) will be transported as IP and traverse the network in the making. It should come as no surprise that such a transition will and I taking place thus allowing for efficiencies derived through a common global infrastructure. As with the other platforms this solution is the best fitted for those Greenfield environments, which are small to medium in size.

The following are some of the economics/key benefits associated with to this solution:

- Minimal layout of capitol for startup.
- No IT staff required to manage or administer the platform.
- Risk associated with out dated or non-standard technology is eliminated.
- Possible savings when significant traffic is incurred between facilities.

- Ability for incorporating multimedia capability (i.e. video, data, voice).
- Ability to leverage applications (i.e. voice mail, etc.)
- Reduced expenditures associated with wiring of facility.
- International facilities can leverage the platform and bypass expensive interstate rates.

For those sites that are purely domestic there is very little in the way of savings when utilized for voice traffic. When comparing that associated with the pricing of a traditional circuit switched environment as opposed to IP while taking into consideration the increased bandwidth required, savings become nominal if any at all. Existing deep discounts provided by the carriers have eliminated the savings, which was once thought to be associated with VoIP.

VoIP Constraints/Issues: A goal in VoIP telephony is added to the calling capabilities associated with that of the legacy PBX while utilizing the IP based network. In doing so a connection to the public switched telephone network (PSTN) as well as the private voice networks is required so as to enable and allow for completion to those not serviced/connected by the IP world.

One of the major hurdles encountered when invoking this platform is found in the ability to maintain voice quality standards (QoS). At this point in time the endeavor at hand is a task that presents many concerns bearing the fact traffic which will share an environment once reserved for data alone.

The following are some key areas of concern focus when the deployment of a VoIP Platform is considered:

- Voice quality (QoS) – In order for VoIP to be successful the voice quality it provides must at least equal that of the PSTN. The following are three primary factors affecting the voice quality.
    - **Delay** – Concerns itself with the amount of time it takes for speech to exit the speaker's mouth and reach the listener's ear. Delay can be introduced at several locations within the transmission, such as during packetization (i.e. placing of the packets) on the network, as well as the network itself. It cannot exceed 250 milliseconds or distortion will exist thus having a direct effect on speech.
    - **Jitter** – Deals with the variation in inter-packet arrival time. If packets are received out of order, they must be held long enough for the slowest packet to be placed in the correct order so they may be played in the correct sequence.
    - **Packet Loss** – Looks at IP networks, which do not provide a guarantee of delivery much less correct delivery order. When dealing with time sensitive transmissions, like voice, the normal TCP based packet retransmission schemes are not suitable.
- Interoperability, having a direct effect on efficiency, is one of the key areas of concern and at present is classified as an open issue. The reason behind this as of this moment is due to the lack of a single standard being adopted (i.e. H.323, SIP, etc.) by the various boards (i.e. IEEE, ITU, etc).

- Bandwidth management is and will remain a major challenge, as the technology platform becomes an accepted, effective means for organizations to conduct business.
- Security is an area that deserves unprecedented attention due to the obvious associated with any financial organization. The following are some recommended precautionary measures:
    - o All connections between network elements should be encrypted; Encryption should be utilized on the last mile if the public network is utilized. It would be unfair to place the onus on the carrier since they have not provide the service when your home or business phone is utilized.
    - o Verification of the network architecture should go through rigorous testing that is ongoing and essential when the current environment is altered.
- SLA's (Service Level Agreement) among the carriers will have a direct correlation on the traffic and QoS incurred. Delay in a conversation encountered by jitter and latency of packet delivery has a direct effect on quality of service.

Transport Technologies: Without a means to effectively/efficiently transport voice traffic over the IP network in a secure way, makes this a futile effort. Fortunately there are many carrier provider services offered both public and private than can accommodate transporting VoIP traffic.

In theory, any data network protocol could support digitized voice such as Asynchronous Transfer Mode (ATM), which was designed specifically to handle multi-media traffic. Today however, most of the attention is focused on Frame Relay, the public Internet and/or VPN networks.

Frame Relay: It is a Layer 2 protocol and as such does not guarantee end-to-end frame/packet delivery. There is no specific provision to support the notion of "priority" traffic and congestion control is achieved by discarding frames/packets.

Frame provides a flexible and efficient data transport mechanism that includes lowered costs for bandwidth in tying together multi-protocol networks/devices.

IP (Internet Protocol) – Public, IP is a Layer 3 protocol and is primarily connectionless or otherwise known as no dedicated end-to-end path. With regard to packet delivery, public IP is a "best effort" protocol.

Again, it is incumbent on the terminal devices to manage and control packet delivery, retransmitting data as necessary.

VoIP takes digitized voice, and divides it into packets that are then sent with other packets across the public IP packet-switched network. At the receiving end, the voice packets are re-assembled and hopefully arrive as a normal sounding voice mail.

IP (Internet Protocol)-Private: IP is a Layer 3 protocol and is primarily connectionless or otherwise known as no dedicated end-to-end path.

With regard to packet delivery, private IP is a "best effort" protocol. Again, it is incumbent on the terminal device to manage to control packet delivery, re-transmitting data as necessary. However, private IP has the ability to provide different levels of service through the use of MPLS and TOS markets in IP packets.

VoIP in essence takes digitized voice, and divides it into packets that are marked as priority and then sent with other packets across the private IP packet-switched network. At the receiving end, the voice packets are re-assembled and arrive as a normal sounding voice call.

Asynchronous Transfer Mode (ATM): ATM is a packet or switching technology based on handing fixed-size cells with each cell composed of "overhead" bytes as well as information or payload bytes.

The ATM scheme was developed specifically to effectively handle multi-media traffic- voice, data and video. ATM is aimed at using relatively high-speed links such as DS3 (i.e. 28 t-1's), OC3 (i.e. 3-DS3's) and higher, and in general is more costly to deploy than either Frame Relay or IP.

Virtual Private Network (VPN): VPN (Virtual Private Network) is an IP based service offered by the various carries, which is normally supported by either a T1 or a Frame relay access connection. Today, because of the converged networking capability this data centric service is now supporting and carrying voice over the same facility. The concept behind this service for handing voice traffic is found within the ability to convert analog or digital traffic into an IP packet via SIP and or H.323 thus leveraging a single service provider for voice and data.

1. Public VPN utilized much like the private in that virtual connections are mapped for transport but differ in that they service a variety of customers thus providing at least in the way of QOS.
2. Private VPN provides mapped virtual connections (i.e. within the carriers network) to service your organizations traffic, offering an extremely secure environment.

Product Integration Profile: This sub-section outlines the findings regarding the technical environment and the standards necessary to operate in an open environment with AIG's pre determined architectural components (i.e. operation systems, databases).

Due to the extensive dependence of interfacing with the data infrastructure alignment to the standard operating systems, supported desktops as well as providing support for database integration are and will remain as essential and pertinent to the platform.

The report paper by AIG which is recommended to evaluate by criteria and compiled into "Evaluation matrices" which provide an understanding of how individual products satisfy company business as it relates to administrative capability while taking into voice and concerned technology such as Call Center are taken into consideration.

A) Evaluation Categories

These evaluation categories include:

- *Vendor's Company* – this category ranks the company's performance, leadership, client references and their ability to support our company administrative.
- *Feature/ Functionality* – this category ranks the product's business functionality and extensibility as compared to a legacy PBX.
- *Product Integration Profile* – this category rank the product's ability to integrate with other packages/applications
- *Architecture* – this category ranks the product's architecture taking into consideration and requirements. This includes operating systems, supported protocol (i.e. SIP, H.323), databases, customization programming interfaces, multi-language, etc.
- *International Presence* – this category rank the company and product architecture's ability to support multiple languages and their ability to support international sites directly and/or through third-party vendor relationships.
- *Operational Ease* – This category ranks the product's ability in utilization on a daily basis (i.e. administration).
- *Installation* – this category ranks the product's required components and time frames associated with installation.

B) Requirements Rating Scales

The table 2-1 and table 2-2 display raking definitions for numerically scored Business and Technical Features and ranking definitions for company and International Support, respectively.

Table 2-1: Business and Technical Features Ranking Definitions

SCORE	RATING	DESCRIPTION
NA	Not Acceptable	Failure to meet one or more critical mandatory requirements (average score <= 1; or any score equal to 0)
NC	Not Capable	Current features are not capable of meeting requirements, with no anticipated timeframe for delivery (average score >= 1; but less than 2)
SC	Requires Significant	The product requires significant change by the vendor and/or the requirement is targeted for release in at least 12 months (average score >= 2 but less than 3)
CM	Can Meet Requirement	The product can meet requirements with acceptable modifications or is targeted for release in < months (average score >4)

Source: Adopted from an AIG report

Table 2-2: Company and Technical Feature Ranking Definitions

SCORE	RATING	DESCRIPTION
NA	Not Acceptable	The company or its support has a record of poor performance, a risky position, lack sufficient leadership, insufficient references, or cannot support international needs.
NC	Not Capable	The company or its support have a record of mediocre performance, an unfavorable market position, exhibit poor leadership, neutral references, or can support limited international needs.
SC	Requires Significant	The company has a record of fair performance, an average position, average references, or fair support for international needs.
CM	Can Meet Requirement	The company and/or its support have a record of good performance, a good company position, leadership and a good reference could provide full support for international needs.

Source: Adopted from an AIG report

The Table 2-3 provides the specific numerical ratings. The lower end of the scale zero (0) is the least desirable. The top end of the scale five (5) is the most desirable.

Table 2-3: Detail Numerical Ranking Scale

Rating	Description of Rating
0-	Not Acceptable
2-	Meets requirement with significant limitations; require customization by the vendor
5-	Meets requirement with product extensions that could be done by AIG

Source: Adopted from an AIG report

C) <u>Weighting Scale</u>

The following chart is a brief explanation of how totals were derived as well as the importance associated with each particular area. As noted below "Implementation/Installation" was given the highest percentage weight of 30% due to the fact that this area associated with the platform tends to be the most critical in determining the ultimate provider of a VoIP solution.

Sources of information for an AIG report, which allowed for these rating criteria, were derived from R.F.I. responses, hands on testing, interviews, and questionnaire, along with printed sources, journals and the Internet.

Table 2-4: Weighting Scale

Questions	A set of questions focused on the primary categories (i.e. Operational Ease, Supported Protocol(s), Feature Functionality, Implementation/Installation and International Presence) as related to the application were presented to each of the various vendors for comments.
Weight	Referenced as a percentage, it depicts the amount of value placed on each category (i.e. Feature Functionality, Implementation/Installation, Operational Ease, Supported Protocol(s) and International Presence) as it relates to the application. The percentages are derived as a direct result of interviews with the various vendors (i.e. Avaya, Siemens & Cisco), which include hand on testing and information derived from printed source (i.e. journals, news articles, etc.) including the internet.
Average	A number was derived and given to each of the vendors which represents how they scored as it relates to the area in question (Feature Functionality, Implementation/Installation, Operational Ease and International Presence).
Section Weight	Is the average divided by the percentage given to each critical category (i.e. Feature Functionality, Implementation/Installation, Operational Ease and International Presence)?

Source: Adopted by an AIG report

International Presence and Support: This sub-section outlines the findings regarding the service and support offered through the international environment with focus on AIG's primary locations. These areas as listed are either serviced through a third party vendor or the present organization.

In the event this is no support or presence in these locations, a rating of a "0" will be given. Where support through a partnership is derived a "2" will be inserted. A "5" will depict the vendor has an office within the region.

Out of the vendors, this scoring item was critical due to the fact that a large majority of company business is accomplished overseas. In addition languages supported for the various platforms in conjunction with the countries were taken into consideration.

Table 2-5: Evaluation by Country

Country	Cisco (Call Manager)	Avaya (S8700)	Siemens (HiPath 5000)
China	5	5	5
South Africa	5	5	5
Taiwan	5	5	5
United Kingdom	5	5	5
Japan	5	5	5
Singapore	5	5	5
Australia	5	5	5
South America	5	5	5
Total	40	40	40
Average	5.0	5.0	5.0

Source: Adopted from AIG Report

1. Cisco:

Cisco Systems is the leading global provider of data networking solutions. Cisco has expanded its market place that was once only associated with data functionality/capabilities (i.e. hardware and software).

Clients of Cisco now benefit from internetworking solutions that enable efficient exchange of information, whether that is voice or data specific thus providing cost savings and process efficiencies. Cisco offers an extensive list of third party providers for sales and service support in all countries that do not have direct presence.

Languages Supported: English

2. Avaya:

Avaya, although primarily a US based organization, offers a strong presence internationally with support either obtained by distributors or a direct or a direct presence. Avaya continues to bring forth their unprecedented knowledge, expertise and flexibility within the voice-related products offered.

Languages Supported: English, Persian French, Spanish, Italian, Portuguese, German

3. Siemens:

Siemens is a European based organization providing sales and support service across the globe through either direct support or a third party provider. As a global organization they continue to expand their business significantly abroad as well as within the US.

Languages Supported: English, German

2.2.3 Identifying VoIP evolution.

VoIP is changing the way organizations are looking to conduct business as well as a way to improve operational efficiencies for new and existing architectures. The IP based technology platform is rapidly altering networks and providing what is perceived as the foundation for facilities/environments to be built upon.

The ability to utilize existing infrastructure and the Internet as a channel for providing service is a reality. Cost benefits are being realized through the utilization of shared applications (i.e. voice mail, ACD, etc) as well as reduced carrier costs. And in addition the platform has proved itself invaluable as another means in providing disaster recovery.

As VoIP technology more becomes a mainstreamer and a standard for a protocol (i.e. H.323, SIP, etc) is obtained, business will begin to embrace the platform while reaping the benefits. The reality that both applications (i.e. voice & data) can coexist while adding tremendous benefit to an existing legacy system or a new switch platform altogether is becoming a reality.

Going forward small to medium-sized enterprise facilities will realize the true benefit and positive impact on the business. Greenfield locations will in fact be ideal in realizing and capitalizing on this technology followed by those requiring uplifts of the existing environment.

Additionally and prior to the technology taking hold some essential elements must be in place, which in fact is based on the theory of a true open architecture and that of an adopted standard (i.e. H.323, SIP, etc).

Businesses should not be converted to IP Telephony just to save money, [Garter advises]. The IT analyst suggests that companies should wait until the productivity benefit delivered by voice and data convergence can be evaluated.

As of now, Voice over Internet Protocol, VoIP, is not as widely accepted as at first projected, but the future has never looked better. According to Frost & Sullivan, VoIP traffic is expected to account for more than 75 percent of all voice traffic by 2007 (http://www.imakenews.com/rcwmirus). IDC, a research and consulting firm, projects that worldwide retail IP telephony services revenues will increase to $19.5 billion in 2005 from $0.4 billion in 2000 (http://www.cisco.com/). And, with broadband households in the U.S. spending an average of $102 in local and long-distance voice services, the potential for VoIP success is tremendous.

The constant products is making further enhancements to its VoIP technology. Many companies plan to introduce SIP, Session Initiation Protocol, into their VoIP call systems (http://www.eweek.com/). SIP is a standard protocol used to establish route, and terminate connections. It serves as an alternative to proprietary vendor protocols in VoIP and PBX systems. Michael Gentry, the senior technical director and chief engineer at the U.S. Army Signal Command, states that it is "simple and nonproprietary and more flexible, so [they] believe SIP is the future." SIP is used in order to provide more enterprise support, and focus less on support from service providers. Enhancements, such as SIP, ensure much more to come from the market leaders in VoIP. With more research and lower costs, the technology for various types of data to be exchanged simultaneously and dynamically seems to have a very promising future.

While there is not many technologies emerging that might affect the success of VoIP, there are a few reasons the technology might not bode well with many companies.

Some believe that the additional costs are much greater than most businesses are able to afford, as well as the fact that the technical expertise needed adds additional costs. One alternative technology is TDMoIP (Time Division Multiplexing over Internet Protocol), which is a transport technology that expands T1, E1, T3, E3, serial data or analog voice circuits transparently across an Internet Protocol. When used for voice, these circuits are transparent to signaling and provide superior voice quality and lower latency than VoIP. TDMoIP gateways allow for a variety of solutions for both enterprise networks and service providers. TDMoIP is one of few other technologies that enable data to be transmitted by Internet Protocol, and it is believed that most of these will lose out to the more advanced, researched, and popular VoIP.

2.2.4 Identifying Voice and Data converged.

Converged networks are allowing the global environment the ability to enhance their business while providing another form of redundancy. No longer is the adoption of the technology driven by cost savings alone; in fact, there are newly realized capabilities promoting the platform. The following are some of the key benefits being derived:

- Sharing of applications (i.e. voice mail/unified messaging, ACD, etc) among a multitude of site thus reducing administration and cost.
- An additional means for providing disaster recovery capability.
- Enhanced telecommuting capabilities offered to the user environment at minimal cost to organization thus providing a transparency to the voice and data networks.
- Cost reduction as previously derived on local and long distance calling fees.

- Inter-office trunk over the corporate WAN/Internet (i.e. toll by-pass) thus replacing tie/private voice trunks between companies owned PBX's.
- Simplification and consolidation: An infrastructure that is integrated to support all forms of communication allows for standardization and reduced equipment and support costs.
- Providing the means for a centralized directory via LDAP.
- Setting the stage for transporting video as well as Voice and data.
- Additional channel for conducting business.
- Ability to links and share information through disparate systems/platforms.
- Reduced costs derived from consolidated management and administration.

The need for additional intelligent multimedia channels that provide service into and out of the organization has taken on a new and significant role in that flexible, remote, cost effective access has become an essential ingredient for success. The work place due to these advancements in technology provides tremendous growth and opportunity within the organization thus having a direct effect on the client base and employee.

This new paradigm does not come easily; there are many factors in today's environment that must be in place to provide this new ability so as to insure security, integrity and quality of signal. This new channel creates a challenge for the vendors and organizations in that proper planning of the architecture must be in place in order to accommodate service effectively.

The technology although touts a cost-effective mechanism for passing voice, data and video is essential that when looking to deploy the platform that call control is related to, information is effectively passed and prioritized (i.e. via UDP & TCP) thus allowing for valid real time information. This information is and will be critical when the utilization of ACD and other applications requiring such pertinent information is considered.

Note: Call Centers are extremely dependent on state information (busy, idle, wrap-up, call in queue, etc.,) for running the supervisory, reporting and agent client applications. With environment a missed state change message means they (and the application) have no way of knowing things are different. Therefore finding agents showed as busy when they are not, calls in queue when they've been already answered, and similar reporting and management issues. In some cases this will be due to the missed the state change message entirely, in other situations because the packets arrived out of order and the client application had no way to verify and reorder the data contained therein.

2.3 Immediate discipline

This section will describe the key is in the implementation IP-PBX and factors to ensure success which went through an extensive evaluations process to choose which system supports and offers to the company's requirements.

2.3.1 How to implement IP-PBX system?

If the system is implemented properly, that nearly all of the major IP-PBX solutions do what they claim. Some features, capacities and underlying architectures differ, but these differences should not result in unhappy customers or replacement of one IP-PBX solution with another. It's true that second implementations often work out better, but only because lessons were learned the first time around resulting in adjustments during the second try.[17]

In most cases, companies tend to choose a system for its features or because they are dazzled by a manufacturer's strong marketing force; then, they go out and look for the cheapest vendor to deliver the solution. This is an inefficient strategy which can backfire both in terms of cost in the long-term as well as in user satisfaction and business operations. While it is important to ensure that the system you choose has all the features you need today, using features as the primary decision-driver is not a wise way to go. The reason is that in the IP-PBX market, much like many other technology sectors, there is a seesaw effect among the solutions. That is, though one manufacturer may offer more features at a given time, another will soon leapfrog the competition with a new release of code, which offers more -- only to be surpassed by yet another manufacturer shortly thereafter.

Mahdavi, Faramarz (2003) stated that the key is in implementation highlighted three key factors below to ensure success. An extensive evaluation process is to choose systems which support and are offered to customers among the factors most critical as follows:

1. ***The financial stability of the manufacturer*:** We want to make sure that the company we choose is financially sound and it is highly likely to remain in business for many years to come.

2. ***High level of domestic and international support*:** Every company claims to have excellent support. We need to make sure that the degree of support, pricing of support and level of support are far superior to others.

Overall pricing of gear, licensing and maintenance: To get a practical grasp of pricing of a telephony solution, you must look beyond the upfront capital costs. In our evaluation, we do not only consider the upfront costs, but also the cost to maintain the system including application licensing, annual maintenance and administration or personnel. One factor that buyers often ignore is the investment of necessary staff to administer the system they are buying. The more complex the system will be, the more senior IT Admin you will need.

3. ***The underpinning architecture and operating system*:** A collective decision to stay away from IP-PBX systems that run on the Windows operating system is. Windows is not necessarily a bad OS for IP-PBX, but we favor real-time operating systems made for mission-critical applications. We also insist on solutions that offer an open architecture and adhere to industry standard protocols.

2.3.2 Preparing Convergence Networking

The intent of this white paper is to provide senior managers with the necessary strategic and financial justifications to make effective decisions regarding investments in IP Communications.

IP Communications — also known as "convergence" — refers to the integration of data, voice, and video solutions onto a single, Internet Protocol (IP) based network. Often perceived as a technology of the future, the products and solutions that companies needed to deploy IP Communications exist today. In fact, a recent Phillips InfoTech study has found that 44 percent of enterprises are already in the process of migrating to IP telephony and that 12 percent of all voice lines shipped this year will be IP station lines. IP Communications is now a viable technology, which is causing many IT managers to reconsider their current network strategies. As a result, PBX sales have declined by 25 percent in the last two years and 62 percent of voice and data decision makers have reported postponing their investments in PBX technology in anticipation of migrating to IP Communications solutions.

Next, the following topics are how to implement IP telephony, network assessment, outside vendors, selecting an outside vendor, and ongoing service.

How to Implement IP Telephony: Many companies have been interested in IP telephony due to the advantages it offers. However, the steps involved in implementing the technology are daunting, especially given companies' fear that the resulting system will not adequately replace their existing telephone networks. Therefore, it is important to undertake a well-thought-out plan of action and seek the appropriate outside help.

***Network Assessment*:** Network assessment is an important phase that must precede the implementation of an IP telephony system. Many companies look for outside help in this key area.

However, the assessment is not a standalone, single step that merely precedes the "real" implementation. Ideally, it is an ongoing process, taking place before, during, and after the installation. The goal is not only to determine the current capabilities of the network to handle voice but to figure out what improvements are needed.

There are numerous reasons to undertake a network assessment. For example, network upgrades will almost certainly be necessary to handle the new capabilities and traffic. Large capital expenditures may be necessary to transition voice traffic, especially when investing in specialized hardware; cost savings will be realized over time, offsetting these initial investments. Because of the expense involved, it is important to determine specific needs beforehand.

Network assessment is a key step in addressing call quality. This issue has dogged IP telephony and, in turn, has affected user perception to some degree, slowing down enterprise adoption. Though it may come cheaper than a traditional phone call, an IP-enabled call is worthless if call quality is so low that it inhibits the end user's ability to conduct business. Cost savings will also be erased if IT managers constantly have to micromanage and fix the system to get it to maintain performance.

Voice over IP (VoIP) call quality is the ultimate indication of converged network success; the network must be thoroughly tested to make sure that human speech is intelligible, happens in real time, and occurs on a consistent basis. Call-quality assessment can draw from a wide variety of fields of study, from electrical engineering to human auditory psychology.

Security and risk mitigation are also key concerns for all networks, (but) especially for ones that involve new technologies and new ways of communicating. IP telephony opens up new possible access points. The assessment will help companies anticipate and block security gaps. In most cases, IP telephony security will be integrated with an end-to-end Internet virtual private network (VPN) solution. Security must be fully tested, including the use of software agents or "white hat" hackers to probe for security holes.

The network assessment phase can help companies avoid buying outdated or closed technologies. The proprietary nature of some of the technologies on the market can be a problem, though the current generation of products is mostly moving beyond proprietary standards. The assessment phase also helps companies plan for added capacity, deal with traffic spikes, and otherwise anticipate real-world growth and problems.

***Outside Vendors*:** As with any new system, all companies should consider how much of their IP telephony services they want to implement themselves and how much they want to rely on an outside vendor. Besides the initial network assessment phase, most vendors offer a wide variety of related services, such as network design, integration, managed data services, equipment installation, and software development.

Core competency is the classic reason for outsourcing, especially when it comes to communications networks. For many companies, working with specialized vendors, especially those that offer service guarantees can be a far more attractive option than hiring internal staff.

Cost may also be a good reason for choosing an outside vendor. IP telephony can be cheaper for a vendor to operate and manage. Because IP telephony is based on software running on basic servers, an outside vendor that can provide simplified browser-based call-control systems can improve administration and control of an IP telephony system.

***Selecting an Outside Vendor*:** When selecting an outside vendor to help with a premises-based, IP telephony implementation, companies should evaluate technology, skills, and services capabilities. The choice of vendor — and services — should be made with an eye toward cost-effectiveness and network efficiency.

As previously noted, many companies look for outside help during the network assessment stage. Skilled vendors use the latest technology assessment tools. These tools can often help companies avoid problems, such as those potentially caused by calls passing through heterogeneous environments, such as from IP to the public switched telephone network (PSTN), that is, the regular phone system.

Access to the PSTN will make IP telephony far more useful for most companies. Companies should also look at what technologies the vendor is using. IP telephony has a history of proprietary solutions, but the market is moving toward more open, standardized implementations. Going with a proprietary solution can lock an enterprise into a relationship with a single vendor while cutting it off from potential improvements in IP telephony technology.

Finally, the enterprise must look at the skill set the vendor provides. It may desire ongoing help from the vendor in several areas. Having an ongoing relationship with the vendor that performed the original network assessment and installation can help with troubleshooting and add-on services.

Ongoing Services: Even after enterprise implements IP telephony, ongoing services can make it easier to monitor and manage:

- Training will be necessary for all staff. Technical staff must learn to implement and maintain the system, while other employees must learn how to use it.
- Testing and analysis tools periodically verify that the system is maintaining call quality as it scales to deal with an increase in volume.
- IP network management services offer predictable pricing and predictable service levels, allowing enterprises to plan ahead more effectively.
- Proactive recommendations for upgrades, including an ongoing upgrade schedule and assessment of new technologies.

Vendors of business communications solutions are preparing new generations of products and services that will move beyond the proprietary standards of the past, making it easier for companies to install and manage IP telephony inside the enterprise.

2.4 The Project Model

After review of the literature, some researches remind that to implement or covert to IP Telephony for company business does just not save money and suggests that company should wait until the productivity benefit delivered by voice and data convergence is evaluated.

The conclusion of research paper indicating about deploying an IP- PBX system is clearly there, and to effectively implement it, companies need to make sure that they have a converged network infrastructure, a converged IT organization and a converged IP-PBX vendor.

2.4.1 Linkage between parent/immediate discipline and project objective

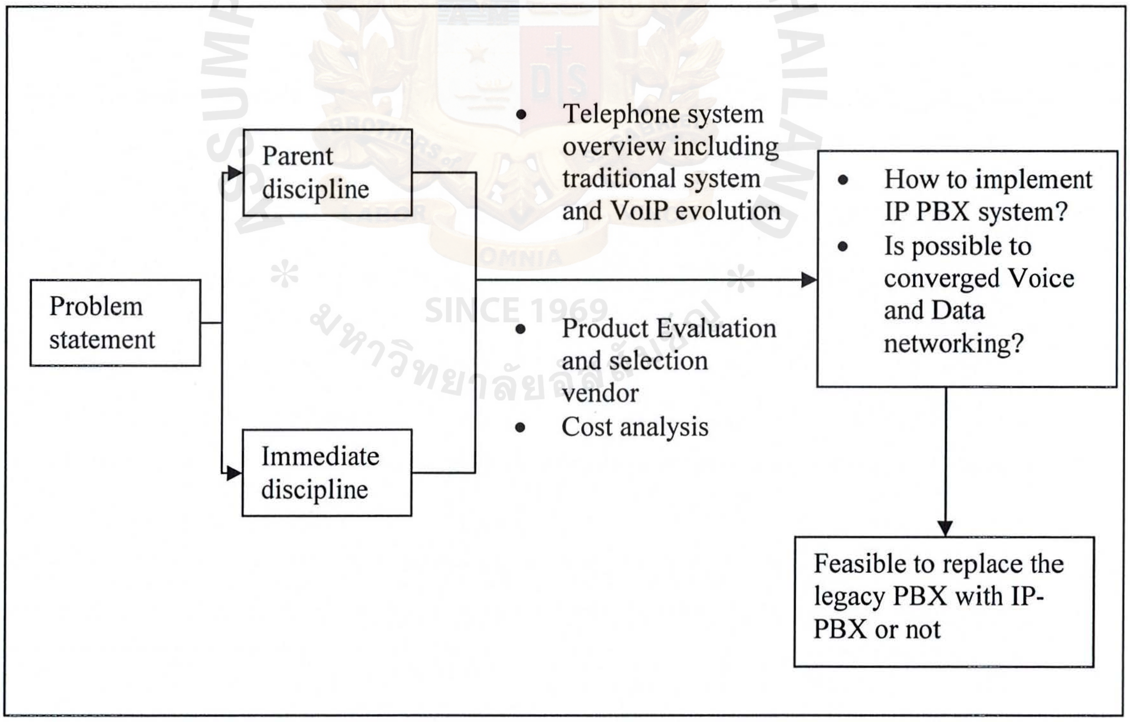
Unfortunately, this factor is often ignored (more) prominent by vendors. It is difficult for vendors to effectively communicate to their potential customers the fact that required changes in the infrastructure which are not really part of the IP-PBX cost, so they ignore it.

On the other hand, buyers view costs in infrastructure changes as only required due to an IP-PBX deployment, so for all practical purposes it is effectively an IP-PBX cost. Those who are responsible on IT should thus view network infrastructure that it needs to be designed right – regardless of voice.

In a properly designed network, not only will you be able to deploy voice, but you will have a far enhanced network for all related applications. Too much emphasis has been placed on cost as the driving factor towards convergence, whereas the emphasis should be placed on resiliency, performance, reliability and ease of management.

The project model will to be associated with the trials completed as well as the intensive research conducted within the company thus enabling the organization to provide what has been determined as an optimal IP telephony solution. The selected vendors have the company's commitment and understanding of what is required by an enterprise organization in providing an intelligent, robust, secure platform that allows for true optimization and efficiency.

Figure 2-3 Overview of project model



Source: Developed for this project

Figure 2-3 explores that Telephone system overview which includes tradition PBX system and VoIP evolution, and Production Evaluation, selection vendor to severed the implementation that is to analyse the cost in order to find out the feasibility to replace the legacy PBX with IP-PBX.

CHAPTER 3: METHODOLOGY

3.1 Project Design

This chapter presents the methodology and the design of this project. Methodology is one of the important parts for doing the project because it can help us to conduct the project and find out the appropriate solution, analyse and evaluate the technology evolution of telephone communication system.

3.1.1 The method of Data collection

Gathering information is done in several ways. In a business that has detailed processes and detailed design documents, reviewing the existing documents could provide most of the information that is needed. In most instances, interviewing key stakeholders and doing researches on infrastructure are required.

This section is targeted at information gathering and a complete assessment of the telephone systems to ensure that business units are using feasible and scalable solutions today as well as being prepared for their next step.

As other IT resources are focused on, the telephone systems will become a part of capacity planning exercise. In terms of time and simplification of the data collection process will be communication study and follow up over the phone, e-mail and face-to-face interview.

The purpose of this section is to gather data from IT and PBX engineering staffs in support of the telecommunications infrastructure and user in each area by interviewing them (see more detail in appendix A). Moreover, the purpose is to gather data from vendors who retain in support of the telecommunications infrastructure.

3.1.2 Propose information for each area

This section focuses on existing infrastructures of voice and data communication networking and analysing initial cost, as well as finding the investing cost of Hardware/Software that will be replaced with IP-PBX which includes the following.

- Voice – The purpose of this section is to understand voice infrastructure including key areas such as usage of trunk lines, cell phones, calling cards, voice service traffic patterns and detailed rates.
- Data – The purpose of this section is to understand about data infrastructure including such key areas as network topology (LAN), inventory, VPN (Viral Private Network) and web hosting.
- Documents – Documents such as vendor contracts that can demonstrate areas of focus within telecommunications infrastructure will be followed up with the phone conversation to validate our understanding of the submitted data, if needed.

3.2 Project Process

3.2.1 Compiling the Information

Once the information has been gathered, it should be consolidated and checked to verify content accuracy and overall intent. Then, this information is placed into a logical format for making a decision. This can summarize a business case of this project which findings with invoice for equipment needs that has extensive voice and data traffic analysis and capital expenditures versus to total savings for this project should include the following:

- Definition of business driver and reasons to go to a new solution
- Explanation of any foreign technical terms; keeping the explanations simple
- Summarized findings of the gathered information
- High-level drawings of current and proposed design
- Definitions of equipment and process needed for the deployment of the new solution
- Definition of acceptance criteria

3.2.2 Designing

The design, focuses on finding out the key factor to implement and design a new platform of IP telephony and selected product which are compatible to voice communication and company requirement as follows;

Basic PBX requirements:

- Analog or Digital extensions interface
- The PBX must support appropriate signaling protocol and must be able to forward the called party number to existing system.

- The PBX must be configurable to allow making calls to PBX extensions and PSTN numbers.
- Remaining existing features of telephone communication.

3.2.3 Selecting Products

The focus of this paper will be based on the three primary Products that installed existing communications infrastructure (Cisco, Avaya and Siemens) and their current platforms, which offer an alternative diverse solution for transporting voice.

3.2.4 Testing

The testing is to determine feature functionality such as security, stability flexibility, ease to integrations and management including quality of service.

3.2.5 Evaluation

The evaluation is on evaluating products and PBX's vendors, functionality, measurement and financial analysis.

3.2.5.1 Product evaluation criteria

This sub-section outlines the findings regarding the technical environment and the standards necessary to operate in an open environment with pre determined architectural components (i.e. operating system, database).

Due to the extensive dependence of interfacing with the data infrastructure alignment to the standard operating systems, supported desktops as well as support for database integration are and will remain as essential and pertinent to the platform.

Table 3-1: Operating System

	Cisco (Call Manager)	**Avaya (S8700)**	**Siemens (HiPath 5000)**
OS Server, NT, 2000& Sun Solaris(UNIX)	2	5	5
DBMS, MSQL, Oracle & Sybase	2 (MSQL)	2(Via ODBC)	1(Informix)
OS Desktop; Win 95, 98, NT, 2000 & XP	5	5	5
Total	9	**12**	11
Average	3.0	**4.0**	3.67

Source: Adopted by an AIG report

3.2.5.2 Functionality/ Features Scores

This sub-section focuses on the functionality offered as it relates to the components/modules utilized by VoIP applications. These evaluated features are outlined below, focusing on the functional characteristics of the products and their custom development effort, which includes flexibility within their product line.

Table 3-2: Functionality and feature

Functionality/Feature	**Cisco (Call manager)**	**Avaya (S8700)**	**Siemens (HiPath 5000)**
Support E911 functionality	2	5	5
Standard support of 3&6 way conferencing	5	5	5
Application supports multiple languages	2	2	5
Supports abbreviated dialing	2	5	5
Provides granularity reporting for application and station usage (i.e. out of the box reports).	5	5	5
Ability to support analog station sets	5	5	5
Inherent Auto Attendant capability	0	5	0
Supports Auto callback	0	5	5
Supports Call forwarding	5	5	5
Supports Call park	5	5	5
Supports ANI (i.e. caller ID)	5	5	5
Supports intercom (i.e. boss, secretary, group)	5	5	5
Supports bridged call appearances	5	5	5
Supports H.323 &/or SIP	2 (H.323)	2 (H.323)	2 (H.323)
Save number redial	5	5	5
Supports Softphone	5	5	5
Supports station hunting	0	5	5
Supports programmable soft keys	5	5	5
Supports attendant console	2 (PC based)	2 (PC based)	2 (PC based)
Supports IP conference phone (i.e. Polycom)	5	2	2
Supports IP Fax capability	0	5	2
Supports music on hold& announcements	5	5	5
Supports flexible LCR/ARS routing	5	5	5
Supports call pickup groups	5	5	5
Supports of telecommuting	5	5	5
Total number of features supported/offered	2	5	4
TOTAL	92	**118**	110
AVERAGE	3.54	**4.54**	4.23

Source: Adopted by an AIG report

From Table 3-2 that although a feature may be supported, it is an option and there will be a 0 placed next to a number as well as provided with a lower score.

3.2.5.3 Features and Fax Support:

1. Cisco:

As a pure IP solution provider their fax support is currently derived via an additional gateway that supports a variety of analog devices. As for speed, Cisco providers speed capability up to 19.2 KBPS. Total features derived from the Call Manager platform as of today are range in the area of 50.

2. Avaya:

Avaya offering both a pure IP solution and a converged environment continues to retain the robust feature capabilities normally associated with legacy platform, which are approximately 450. As for their fax capability, it is supported through analog as well as IP utilizing T.38 and it groups 3 standards. At present, supported speed of analog and IP fax is at 33.6kbps.

3. Siemens:

Siemens, as well, offers both a pure IP solution and a converged environment. Siemens also continues to retain a robust system much like their legacy platform. As for their feature functionality their IP solution appears to be a subset of their TDM switch offering approximately 250.

As for their fax capability, which is derived externally through an AP1140 (access point/station side gateway) providing support of analog devices, such as fax, modems, etc up to a speed of 33.6kbps.

3.2.5.4 Implementation/Installation

This sub-section focuses on Installation of the application as it relates to the system components and their flexibility in accommodating a range of operating system /workstation configurations as well as database capability, redundancy and the ability to integrate with other applications.

VoIP as an application requires flexibility within the system to accommodate a multitude of configurations. These evaluated components are outlined below, focusing on the functional characteristics when the product is deployed into the facility(s).

Table 3-3: Implementation and Installation

Implementation/Installation	**Cisco (Call Manager)**	**Avaya (S8700)**	**Siemens (HiPath 5000)**
System is Open DataBase Connectivity compatible with the ability to write to Oracle, Sybase, Informix, etc?	4	4	1 (Informix only)
Inherent maintenance tools provided for fault isolation?	5	5	5
Security feature associated with application?	2	5 (encrypted)	2
Application Program Interface (API) for CTI/ Middleware providers (i.e. Genesys &GeoTel) and CRM (i.e. Point &Siebel)	4	5	5
Support of multiple operating systems for servers. (i.e. Windows 95, 98, 2000, NT, XP & UNIX)	2	5	5
Technical support/maintenance offered for product.	5	5	5
Client workstations/soft phone support windows 95, 98, 2000, NT & UNIX	5	5	2
Power to stations adheres to IEEE 802.1AF and/or IEEE 802.3AF standards.	2 (proprietary)	2 (802.1AF)	2 (802.3AF)
Supports networking capability (i.e. QSIG, etc.)	2 (AMIS)	5 (AMIS,DCS , QSIG)	2 (QSIG)
Route patterns/facility restriction capability	5	5	4

Application/Platform is LDAP compliant	5	5	5
Platform offers power fail capability (i.e. CO trunks)	0	5	5
Gateway supports T1/E1 (ISDN/PRI) & Analog trunks	5	5	5
Audio compression (vocoders) support of 50, 100&150 stations.	5 (1)	5 (1)	5 (1)
Platform supports multiple messaging platform (i.e Notes and Exchange)	2	2	2
Grade of service overflow (i.e. PSTN)	0	5	2
Total	66	88	74
Average	**3.47**	**1.63**	**3.89**

Source: Adopted from an AIG report

3.2.5.5 Redundancy/Survivability:

This sub-section focuses on vulnerability of the application/platform as it relates to redundancy for remote facilities and their dependencies on the hardware/software in place so as to retain voice communications. Consistency and reliability of the platform along with the retention of feature functionality is a need that must be inherent in designing a voice environment when an IP based switch is utilized.

System components have been identified at a high level so that they may be considered when opting to deploy an IP based solution. VoIP as a platform will be dependent o multiple components to accommodate what is now provided in the way of capability. The evaluated area is outlined below, focusing on the true capability when survivability is looked at.

1. Cisco:

The following are the needed requirements relating to the Cisco's platform in providing survivability to a remote office as well as pointing out where a single point of failure might exist.

- Router/gateway (single point of failure)
    - Router provides feature functionality stored in memory (SRST) for remote sites
- Call Manager (central site) (two required for redundancy)
    - Operating system being windows based
    - RAID
    - Power
- Ethernet switch (two required for redundancy)
    - Power modules (two required for redundancy)

All sites are designed with two routers to be placed as a standard with the option for each business unit to decline whether or not the need is present.

2. Avaya:

The following are the needed requirements relating to the Avaya platform in providing survivability to a remote office as well as pointing out where a single point of failure might exist.

- Router
- Ethernet switch (two required for redundancy)
  - Power modules (two required for redundancy)
- IP600/S8100 (central site)
  - RAID
  - Power
- G700 (remote office support supplied with internal gateway and redundant processors and power).
- 8700 (remote office support supplied with internal gateway with sleeping processors and power).

3. Siemens:

The following are the needed requirements relateing to the Siemens platform in providing survivability to a remote office as well as pointing out where a single point of failure might exist.

- Router
- Ethernet switch (two required for redundancy)
  - Power modules (two required for redundancy)
- HighPath 5500 (central site)
  - RAID
  - Power

- HighPath 5300 (remote site support with intelligence)
    - RG 21 gateway (single point of failure)
- HighPath 3000/5000 (remote office support)
    - Solution is provided via hardware and/or software
    - Local gateway is imbedded within the platform (i.e. 3000/5000)

Note: Siemens does not provide media encryption thus limiting security capability.

Integrated Messaging:

1. Cisco:

Messaging platforms that Cisco currently supports and is integrated with are the following:

- Microsoft Outlook/Exchange version 5.5
- Lotus Notes is currently in development

2. Avaya:

Messaging platforms that Siemens currently supports and is integrated with are the following:

- Microsoft Outlook/Exchange version 5.5
- Lotus Notes due to be released on 8/02

3. Siemens:

Messaging platforms that Siemens currently supports and is integrated with are the following:

- Microsoft Outlook/Exchange version 5.5
- Lotus Notes
- Netscape

CHAPTER 4: ANALYSIS AND DISCUSSION

4.1 Analysis Accepted Product

From the three Products, the following final scoring is a mandatory criterion of their ability to provide and support these components thus having a direct effect on maintaining the existing investment.

The final scores have taken the major categories into account, representing the key components, elements and features associated with level of importance.

Table 4-1: Vendors Evaluation Result

	Question Break Down	Weight	Cisco Average	Cisco Section Weight	Avaya Average	Avaya Section Weight	Siemens Average	Siemens Section Weight
Feature Functionality	26	25%	3.54	0.885	4.54	1.135	4.23	1.057
Implementation/ Installation	19	30%	3.47	1.041	4.63	1.389	3.89	1.167
Operational Ease	8	25%	3.5	.875	4.8	1.2	3.7	.925
International Presence	8	20%	5.0	1.0	5.0	1.0	5.0	.10
Total Weighted Average			3.81		4.72		4.14	
Final Product Rating			NC		CM		CM	

Source: Adopted by AIG Report

4.2 Conceptual designing Network diagram

This section is designed and analysed from Chapter Three as the basic requirements can be designing two scenarios of conceptual network diagramming which is differently from media communication.

4.2.1 System Requirement

1. Upcountry areas: All PABXs (7 locations). As our business in UPC grows like calls volume at each location, those PABXs with basic features can no longer keep-up with new requirements.
2. Bangkok Area: AIG (over 10 years old system). AIG's ROLM PABX with RAD MUX requires high MA costs from single qualified vendor.
3. Bangkok Area: SPK (over 10 years old system). SPK's OKI PABX provides limited features and capabilities.
4. All other locations in Bangkok area (AIT/AIB, AIC, JCB, CTS, SCT andJCK). Upgrade IP processor is suggested. IP phones should be launched to a group of high-end users at AIT/AIB as a pilot test before expanding to other areas in the future.

4.2.2 Solution

Having reviewed the system configuration requirement of all branches in Thailand, this project proposes on preliminary PBX migrations.

4.2.2.1 Bangkok

1. Replacing the ROLM and OKI with voice gateway that sufficient to handle the traffic based on the existing system capacity if there is no Server redundancy requirement and used to cater for the traditional analog/digital voice cards.

2. Upgrading all system in BKK with the latest version with IP enabled so that IP Trunk can link all systems up.

3. Deploying gateways in CTS, SCT and JCK, these gateways are and connected to other systems with the T1 trunk. Under such circumstances, all Pabx systems in BKK are networked by IP trunk.

4.2.2.2 Upcountry

Replace the rest PABX systems with Local Survivable Processor. (LSP may not be required if the site is not considered a critical location)

4.3 Implementation Issue

As one would imagine, there are several aspects to implement packetized voice networks. From a technical perspective, differences among the various technologies – Frame Relay, IP and ATM – can significant. The balance of this paper will address some of the implementation issues, which must be addressed and expanded upon.

Network Design Considerations

The following areas surround the various standards being implemented and the recommended protocols, as well as other key items needing serious consideration when implementation issues such a platform are considered.

- Support of a protocol that has been deemed by the various committees as a standard.
- Quality of Service (QoS) targets/standards.
- Overall bandwidth requirement for all service types.
- The transmission speed of the network infrastructure.
- Physical topology and physical media of the network infrastructure.
- Minimum and maximum packet sizes.
- Traffic measurement and engineering (call volume, stations to be supported, applications (i.e. ACD, voice mail, etc), etc).
- Jitter, the variation in arrival time between packets.
- Latency (i.e. QoS), or the delay from the signal source to the signal destination through the network.
- Intelligent switches
- Routers ability to prioritize voice traffic.
- Security associated (i.e. Firewall and platform related)
- Infrastructure cabling (i.e. Cat 5 or higher)
- Resilience/redundancy
- Internet access

Note: Cisco's Call Manager requires their routers to be in place in order to offer feature functionality.

- No hubs with in the network only intelligent switches.
- Utilization of both voice and data networks (i.e. IPand PSTN)
- Security provided for the network and application/technology

4.4 Transportations and Cost analysis

The replacement IP-PBX system must analyse technology transporting, traditional tied line telephone, VoIP tied line, equipment cost and benefit.

Table 4-2: Transporting VoIP versus VoFR

Issues	Traditional	VoIP	VoFR
Technique	TDM	IP Package	FR Cell
Bandwidth	64Kbps	24 Kbps	15-16 Kbps
IP Phone	Can Not	Can	Can not
E1/T1	Need	Do not Need	Need
Internet Connection	No.	Yes	No.
FXO and FXS (4W E&M)	Yes	No	YES

Source: Developed for this project

Table 4-3: Traditional Tie-Line Telephony Cost

Location	**Brand**	**Channels**	**Tie-Line Cost**
AI Tower	*Avaya*	*N/A*	-
To Jewelry	*Avaya*	*60 Ch.*	*20,000*
To AIC (Sathorn)	*Avaya*	*90 Ch.*	*30,000*
To CharterSquare	*Siemens*	*30 Ch.*	*20,000*
To Sathorn City	*Siemens*	*N/A*	-
To JC. Kelvin	*Siemens*	*30 Ch.*	*20,000*
To AIG Bld.	*Rolm*	*90 Ch.*	*25,000*
To Supakarn	*Oki*	*N/A*	-
To S&A	*Avaya*	*60 Ch.*	*20,000*
TOTAL			***135,000***

Source: Developed for this project.

Table 4-4: VoIP Tie-Line Telephony Cost

Location	**Brand**	**Channels**	**VoIP Tie-Line Cost (Est.)**
AI Tower	*Avaya*	*N/A*	-
To Jewelry	*Avaya*	*60 Ch.*	*10,000*
To AIC (Sathorn)	*Avaya*	*90 Ch.*	*15,000*
To CharterSquare	*Siemens*	*30 Ch.*	*10,000*
To Sathorn City	*Siemens*	*N/A*	-
To JC. Kelvin	*Siemens*	*30 Ch.*	*10,000*
To AIG Bld.	*Rolm*	*90 Ch.*	*15,000*
To Supakarn	*Oki*	*N/A*	-
To S&A	*Avaya*	*60 Ch.*	*12,000*
TOTAL			***72,000***

Source: Developed for this project

Table 4-5: Equipment Costs (Estimate.)

Location	**Brand**	**Method**	**Cost (Estimate)**
AI Tower	*Avaya*	*Upgrade*	*3.2 Mil.*
To Jewelry	*Avaya*	*Upgrade*	*0.8 Mil.*
To AIC (Sathorn)	*Avaya*	*Upgrade*	*0.8 Mil.*
To CharterSquare	*Siemens*	*Add-on*	*0.4 Mil.*
To Sathorn City	*Siemens*	*Add-on*	*0.4 Mil.*
To JC. Kelvin	*Siemens*	*Add-on*	*0.4 Mil.*
To AIG Bld.	*Rolm*	-	*N/A*
To Supakarn	*Oki*	-	*N/A*
To S&A	*Avaya*	*Add-on*	*0.4 Mil.*
TOTAL			***6.4 Mil.***

Source: Developed for this project

Table 4-6: Tie-Line Only Break Even Points

Year	Traditional (Annual)	IP/FR Network (Annual)	Annual Save	Equipment Cost	Account Save	Account Equipment Cost
-	1,620,000	864,000	756,000	6,400,000	756,000	6,400,000
1	1,620,000	864,000	756,000	192,000	1,512,000	6,592,000
2	1,782,000	777,600	1,004,400	192,000	2,516,400	6,784,000
3	1,960,200	699,840	1,260,360	192,000	3,776,760	6,976,000
4	2,156,220	629,856	1,526,364	192,000	5,303,124	7,168,000
5	2,371,842	566,870	1,804,972	192,000	7,108,096	7,360,000
6	2,609,026	510,183	2,098,843	192,000	9,206,938	7,552,000
7	2,609,026	510,183	2,098,843	192,000	11,305,781	7,744,000
8	2,609,026	510,183	2,098,843	192,000	13,404,624	7,936,000
9	2,609,026	510,183	2,098,843	192,000	15,503,467	8,128,000

Source: Developed for this project

CHAPTER 5: CONCLUSION

5.1 Summary of the Result

According to Appendix C, the break-even for investment this VoIP is in Year 5. From the feasibility study, it shows that the technology is feasible to implement. The technology of implementing it consists of following:.

- New Technology for voice transporting
- Change in Old Fashion Equipments
- IP Phone – Soft phone and Hard Phone in Anywhere (upcountry or oversea)
- SIP Protocol
- Enhancing Productivities – web, Unified Messaging
- Single Network Management
- Big Step over the competitors
- Reducing cost on multi-site call
- Increasing application on telephony
- Increasing productivities on employees
- Simplifying Cabling system
- Simplifying the management

To increase the efficiency of IP-PBX, it requires budget, knowledge of users, engineering specialization, and vendors.

5.2 Limitations of the project

The project limits to the feasibility study. It cannot reach to testing stage since the current network does not yet support in terms of security.

5.3 Contributions of the project

The contribution of the project is to an organization. The organization can be benefited in terms of cost (calling cost reduced), and quality of voice communication. It also enhances the employee efficiency.

5.4 Recommendations and future research

The future research for project is to test the system. The test result will provide the improvement for stage of the implementation.

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APPENDIX A: INTERVIEW DATA

The questionnaires are being requested to provide further information:

<u>Interviewing Admin. Service Manager:</u>

Q: Having conducted the exercises of Global PBX Review early this year, we understand that there are some obsolete PBX systems are still used in our regional offices and the manufacturers no longer support those PBX systems. Such as:-

- **Malaysia - Lucent G3siv4 installed at Penang**
- **Thailand - OKI 1200 installed at SPK. IBM Rolm 9751 in AIG Tower**
- **Singapore - IBM Rolm installed at Tanjong Pagar, Rolm 8000 in Changi.**

Under such circumstances, I would like to know if there is any strategic plan being contemplated to replace or upgrade those systems in our country?

A: We are considering replacing the 2 PBX systems. We are working with R/E (Khun Thitirat) to review users' requirements at the 2 sites in order to prepare a business case for local's and HO's consideration so that we can propose it with 2004 budgeting season accordingly. It is expected that a draft business plan would be done by the Q1 of 2004.

Q: Apart from the legacy PBX system, has ATDC studied the feasibility of using the IP telephony system for AIG offices before? What are the market trends of VoIP in Thailand?

A: For upcountry locations, we have already been using VoIP to save long distance costs.

For Bangkok offices, we need to consider router upgrade, as these routers have no VoIP hardware.

Interviewing Network Manager

Q: Is there any VoIP deployed in your offices for toll bypass and IPT? If yes, please provide details of site, size & application etc.

A: Yes, Router's FXS interfaces connecting directly to PBX's CO lines.

(Referring to figure 1-1)

Q: Do you have any plan to deploy VoIP in the near future including toll bypass and IP Telephone?

A: For upcountry locations, we have already been using VoIP to save long distance costs. There are no airtime costs for calls made within Bangkok area (3 baht per call and no air time). In Bangkok offices, we need to consider router upgrade, as these routers have no VoIP hardware.

Q: Is there any benefit in using VoIP in our offices?

A: Our upcountry SC-CNX is one example to deploy VoIP (voice over IP) using router via second data-com line (back up line) to save cost on long distance calls. Based on statistics, on average, each of 33 staff will make 2.5 calls to AIA HO and other AIA locations. Average duration of each call is 4 minutes.

- Total number of calls made = 2.5 x 33 = 83/day or 83 x 4 = 332 minutes/day
- Total minutes made by staff = 22 x 332 = **7,304** minutes/month

- In addition, agents also make use of our VoIP to access IVR system = 300 minutes/day or
- 22 x 300 = **6,600** minutes/month
- Total minutes = **13,904** minutes/month
- Below is monthly cost comparison table:

	cost (baht)	cost (US$)	remark
AIA VoIP	43,924	976	monthly datacom cost (DataNet)
public long distance	250,272	5,562	18 baht/minute
Y-tel 1234 (public VoIP)	111,232	2,472	8 baht/minute

- The cost saving on monthly long distance calls as well as pay back for router h/w cost.

Q: Please provide the narrative and market share of VoIP for respective vendors in your country. i.e. Avaya, Cisco, Siemens and Alcatel .

A: Cisco router is AIG standard data network equipment. For IP PABX it is not yet implemented.

Q: Have you contacted the vendors before for VoIP deployment?

A: We are working with our vendors if they have any information on this at all. In general, each vendor usually, each party, would keep its figure if confidential, so we might be able to get some from our certified PBX vendors: AVAYA & SIEMENS

Q: Have you contacted the vendors before for VoIP deployment?

A: Yes, there have been presentations, especially from AVAYA's vendors on this from time to time. Actually, it has been mentioned and brought up more and more often, especially in the past 3 months. As discussed, we would definitely put a great deal of consideration to IPT and VoIP concept to our future PBX's development. This point has also been raised by AIGDC. We have shared our network structure, plan and position with him. There have been many useful suggestions from the meeting concerning future projects related to PBX with IPT and VoIP implementation. AIGDC also shared with us that a universal IP standard has been set, and agreed by major vendors. That team would kindly forward more information on this to us so that we can conduct a study with our vendors in Thailand accordingly.

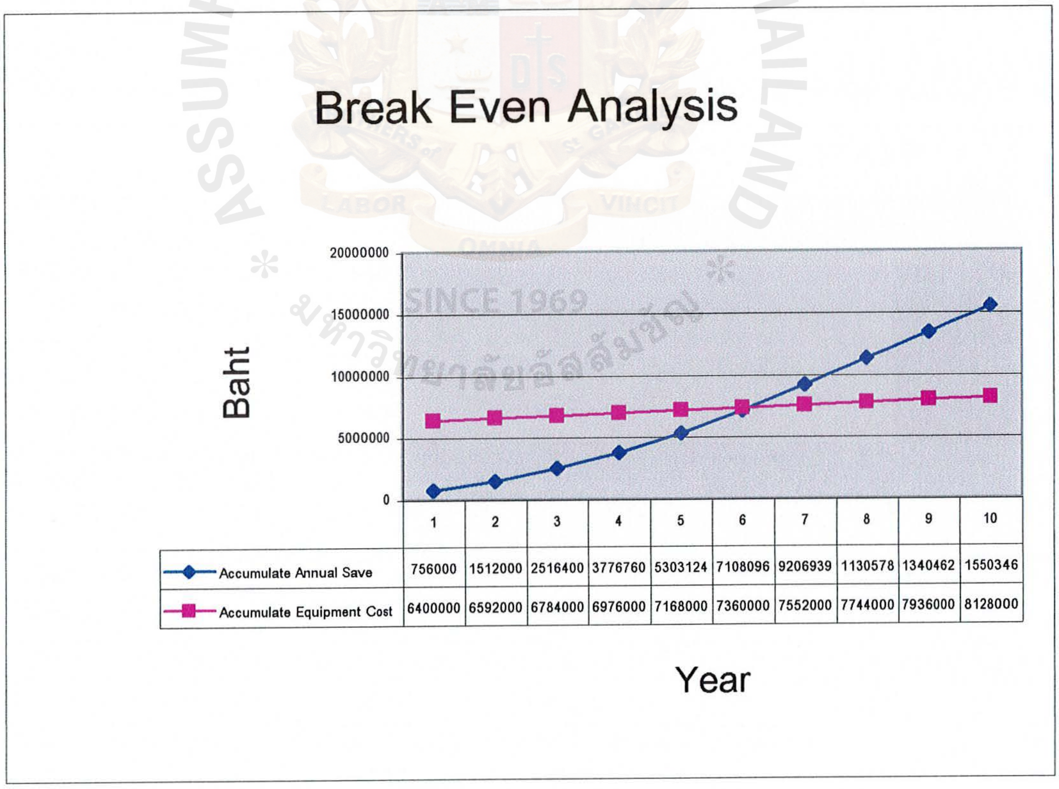
APPENDIX B: VoIP CONCEPTUAL DIAGRAM

APPENDIX C: BREAK EVEN POINT

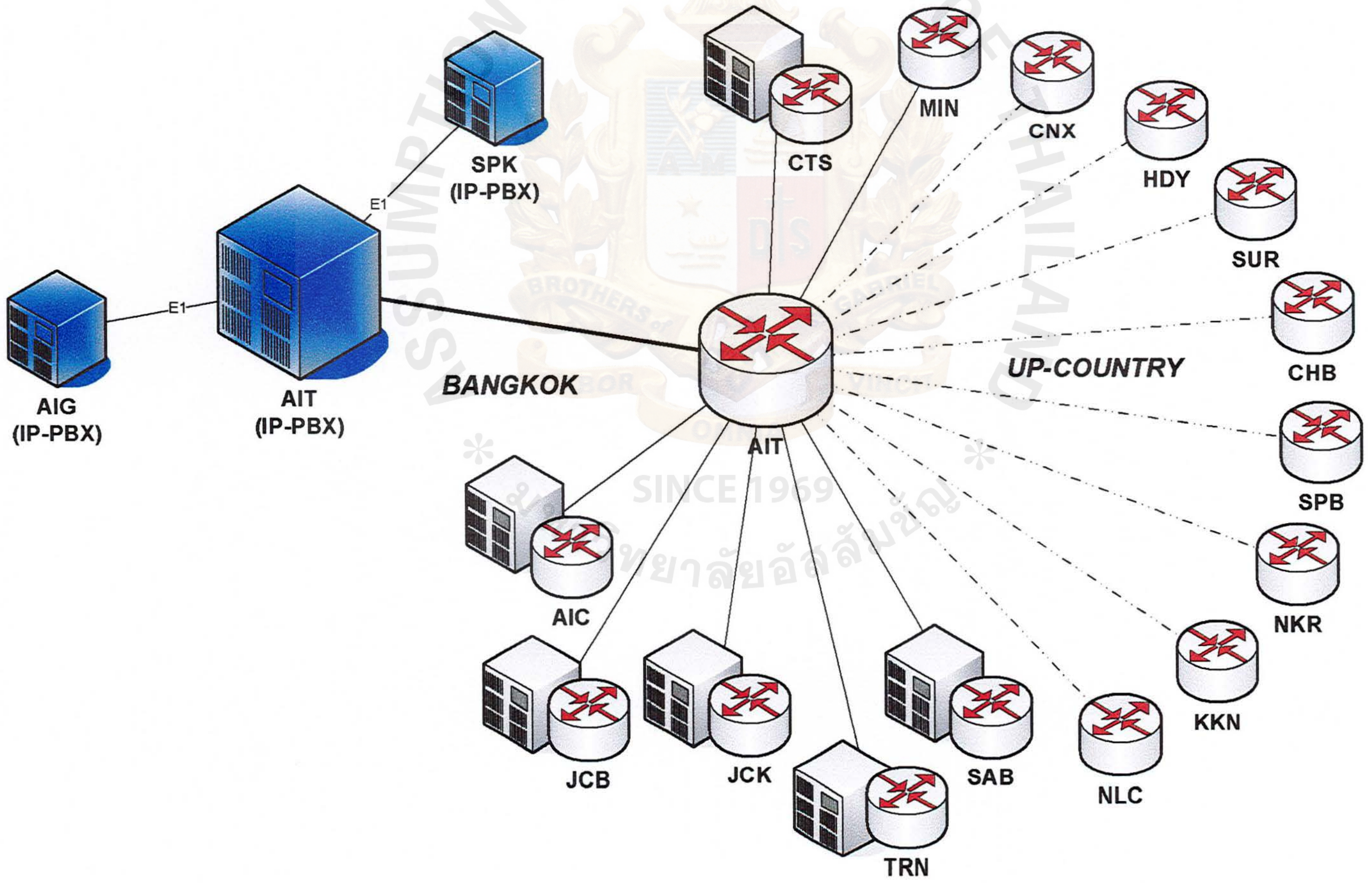
Tie-line Break Even Analysis

	A	B	A-B = C	D	C	D
Year	Traditional (Annual)	Network (Annual)	Annual Save	Equipment Cost	*Accumulate Annual Save*	*Accumulate Equipment Cost*
-	1,620,000	864,000	756,000	6,400,000	756,000	6,400,000
1	1,620,000	864,000	756,000	192,000	1,512,000	6,592,000
2	1,782,000	777,600	1,004,400	192,000	2,516,400	6,784,000
3	1,960,200	699,840	1,260,360	192,000	3,776,760	6,976,000
4	2,156,220	629,856	1,526,364	192,000	5,303,124	7,168,000
5	2,371,842	566,870	1,804,972	192,000	7,108,096	7,360,000
6	2,609,026	510,183	2,098,843	192,000	9,206,939	7,552,000
7	2,609,026	510,183	2,098,843	192,000	11,305,782	7,744,000
8	2,609,026	510,183	2,098,843	192,000	13,404,625	7,936,000
9	2,609,026	510,183	2,098,843	192,000	15,503,468	8,128,000

Assumption: Maintenance cost 3% per year



<u>**VoIP Network Conceptual Diagram (Scenario I)**</u>



<u>**VoIP Network Conceptual Diagram (Scenario II)**</u>