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Analysis of Telephone System of a University Campus and Design of a Converged VoIP System

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Abstract

VoIP(Voice over IP), so called Internet phone, provides several unique advantages, for instance, inexpensive long distance call service than the existing wire telephone network and various multimedia services from Internet network added to voice call service. This has been widely discussed and has been a research topic throughout the developed countries in the recent years. The paper is intended to describe and analyze the feasibility of the results of the VoIP system designed if implemented in a campus network like Dhaka University telephone network. Thus the convergence strategy of an IP PABX system with the existing wire telephone network has been discussed. While taking this into consideration, upgrading method of the existing system using modern network appliances has been described and at the same time new network topology connecting the present IP network with the telephone network has been designed.

Keywords: VoIP, IP telephony / PABX system, Network convergence, Dhaka University

1. Introduction

Voice over Internet Protocol (VoIP) is a multifaceted topic. There is no single template that can be followed when contemplating the use of this technology to supplant a traditional phone system. The technology itself is fairly generic and somewhat mature, however the impact to the external environment, business conditions surrounding the actual requirements, capital funding plan and other integral factors complicate the analysis. Implementing VoIP at Dhaka University, is much the same as elsewhere except for unique characteristics endemic to the University. It is however, these unique characteristics that warrant a closer look since capital investment will be significant and once a solution is installed, it will most likely remain long beyond its' intended useful life.

There has been much discussion about VoIP phone systems within commercial and public environments. *"Although progressing rapidly, Internet telephony still has some problems with reliability and sound quality, due primarily to limitations bothin Internet bandwidth and current compression technology".* Some companies have adopted use of VoIP systems as a replacement for traditional phone systems, Public Branch eXchanges (PBXs) and Centrex type services. Most of this however, is on private intranets where bandwidth is available and predictable. Many larger organizations have adopted hybrid solutions due to geographic vagaries, economics or administrative manageability.

Within the confines of the University, there are several factors that differentiate this

environment from that of a corporate one. Although these don't affect the technology trend or the economics associated with VoIP deployment, they do influence the nature and type of deployment campus wide. This document highlights many key areas that need to be addressed when considering large-scale deployment of VoIP as well as some special considerations relative to the University.

The results are not surprising when all influences are considered. If we analyze the business need and then factor in the appropriate variables, the strategy seems straightforward. Some of the variables to consider are cost, the existing infrastructure, implementation practicality, maintenance, operability and the University topology.

2. Background

2.1 Telephone System

In order to get a telephone call to travel from one place to another, it must pass through the telephone network. This network consists of many different parts, operated by many different companies, but are inter-connected using common signaling methods. Physical components required for telephone networks (figure-1.1)are: Transmission Facilities, Local Loop, IOF - Interoffice facilities, Switching Systems, Customer Premise Equipment (CPE).

Central Offices, signaling between different telephone systems, different methods of transmission, and the use of tandems (transfer) in the network are discussed below:

On each telephone call, a talking path must be set up between the calling and the called telephone. The method of making this connection, known as switching, has progressed from the simplest of hand operated switches through the more complex manual systems to the present switching systems. In telephone switching systems, phone calls are "switched" meaning they cross through a switching matrix to route calls from an origination point to a destination point.

Local "central offices" are where the end users are located. This is the most common switching system in the telephone network. Long Distance "tandem" switches are where long-haul long distance calls are switched to connect local central office switches throughout the world.

PSTN (Public Switched Telephone Network) is the traditional telephone network that provides POTS (Plain Old Telephone Service); that is, the network that anyone could access by circuit- switch connection. This connection is the dedicated service that can guarantee the reliable, accessible and quality. PSTN converts voice into electrical signals and transmitted through the circuit-switched network.

2.2 Existing PABX System of Dhaka University Campus

Currently, Dhaka University is using Mitel SX-2000 LIGHT PABX system The Mitel SX-2000 LIGHT system is an advanced, fiber-distributed telephone system that is designed for larger organizations or for networked telecommunications environments. The distributed architecture separates the control node from peripheral, application, and network access nodes and links them by multimode fiber optic cable.

The SX-2000 LIGHT system can be configured as a multi-cabinet, control redundant system.

The redundant main control cabinet can support up to eleven expanded peripheral nodes located up to 8.7 miles (14 kilometers) away. The redundant main control cabinet also supports up to five DSU cabinets or Network Services Units (NSU). So, this installation is used in Dhaka University campus network.

The SX-2000 LIGHT system (figure 1) consists of a redundant main control cabinet and associated peripheral cabinets. Fiber optic cables connect the peripheral cabinets to the main control cabinet. Copper cables from the extensions terminate at the peripheral cabinet. Installers do not have to route the extension cables between many floors or run the cables off-premises to a centrally-located system.

Depending on the Fiber Interface Module (FIM) that is used, the peripheral cabinets can be located up to 0.62 miles (1 km), 1.9 miles (3 km), or 8.7 miles (14 km) from the main control cabinet. This versatility allows system resources such as lines, trunks, and digital service applications to be physically distributed among several remote locations.

The topology consists of the local loop, circuit-switched telephone network with the central node being the local central office. In this topology, 11 peripheral nodes have a dedicated point-to-point link to a central node. And another 4 peripheral nodes are extended through extenders with 4 of the main peripheral nodes. If one node wants to send data to another, it sends to the central node, which then relays the data to the destination node.

In this network, optical fibers are used to connect the main 11 peripheral nodes to the control node and copper wires have been implemented for all the other connections.

The positions of the nodes and the distribution boxes are given below:

1. Control node : 1 – Administrative Building
2. Peripheral nodes : Total – 15

a. Business Studies –3 nodes
2 distribution boxes

b. ISWR – 1 node

1 distribution box

c. Curzon Hall – 2 nodes

1 distribution box

d. Khondokar Mokarram Hussain Building – 2 nodes

1 distribution box

e. Arts Building – 2 nodes

1 distribution box

f. Register Building – 4 nodes

3 distribution boxes

g. Science Annex – 1 node

1 distribution box

The schematic diagram of the campus PABX system topology is given in figure 2.

3. Literature Review

3.1 What is IP PABX?

Over time, the PABX has grown to incorporate all sorts of advanced features such as voicemail, unified messaging, auto attendant (IVR), automatic call distribution (ACD), call queuing, branch office support, telecommuters, softphones, CTI (integration with the PC), and more. With the advent of IP, the acronym PBX morphed into its latest incarnation, the IP PABX. An IP PABX is a PABX that supports packet-based transport protocols - commonly referred to as "VoIP". The most popular current protocol is SIP, which stands for "Session Initiation Protocol". Then, as the IP PABX began to rise in market share an even new label appeared called the "Hybrid IP PABX" A Hybrid IP PABX is an IP PABX that, along with VoIP, also supports legacy and analog switching protocols such as TDM. A Hybrid IP PABX will typically also support analog phones to complement its support of IP Phones.

That is, with VoIP, a PABX is still used to distribute calls throughout an organization, but the PABX must support the Internet Protocol being used. In actual practice, such an IP PABX is a computer server. Once received by the IP PABX and inside the organization, the call may be carried through a data network or via digital or analog communication lines to recipient endpoints. If calls are distributed inside the organization on a data network, the need for a dedicated communications circuit is eliminated and the calls can be carried on the internal LAN as data packets.

3.2 What is VoIP?

Voice over Internet Protocol (VoIP) is a new way of communicating. It requires the use of the internet and the technology that allows information to travel between users. Before I delve into how VoIP works it is important to understand how the plain old telephone system (POTS) works.When you make a telephone call your voice travels in its analog form from your telephone to a telephone company switch. At the switch it is determined if your call is passed to another switch or if it can be routed to its destination. Using the POTS system your phone call travels the same path along physical wires based upon where the phone call will terminate.

VoIP requires an individual to have broadband Internet access. The most common of which is either through the use of a cable modem or DSL (direct subscriber line). Data travels the Internet in packets. In order for your voice to travel the Internet, it must be converted from analog to digital. Once your voice has been converted to digital packets it can be transferred via the Internet. But before your voice can enter the Internet and travel to its destination some information has to be added to it so that it meets IP (Internet Protocol) standards.

It mentioned earlier that data travels in packets, that is your voice is broken up into several different packets that must reach your destination and be reassembled in the right order and converted from its digital form back to an analog form so that the recipient can hear your voice. So each packet of data, which contains a portion of your phone call, has a minimum of 160 bits added to it so it can reach its destination [30]. In addition to the IP protocol it also requires the use of real-time protocol (RTP). "It provides timing information that allows the receiver to reconstruct the original timing of the transmitted material in a way that identifies the content being sent, provides security, and notifies the overriding application of lost data." RTP is usually used in conjunction with unreliable datagram protocol (UDP). UDP allows for information such as voice and video to be sent without waiting for acknowledgement of it being received. "It is useful in cases where one sender wants to send the same information to multiple receivers and is not too worried if some pieces get lost along the way." VoIP requires the use of these three protocols in order to function. For this reason broadband access is needed. The fastest modem dial up connection using POTS can only achieve 53Kbps (kilobits per second) whereas cable modems and DSL can achieve speeds of 10 – 100 Mbps (megabits per second). The higher speed is needed in order for an actual conversation to take place over the Internet without significant delay.

3.3 What are the technology requirements?

To get started with VoIP service there are a few basic items you need: Broadband Internet access, computer and software if you plan to use the computer as your phone, adapter box if you plan on using your regular house phone. However, for a call center there are more things to consider.

You still need broadband Internet access but supporting hardware becomes more complex. First, a decision is need to determine how much bandwidth is required to support adequate VoIP telephone service. Then you need to determine the amount of bandwidth you have on your Internet access. Then you can determine how many phones can be supported by a single broadband connection. In order to share the broadband connection an intelligent router will be needed to split the connection between all of the phones.

Westminster college in Salt Lake City, Utah changed their phone system over to VoIP. Based upon their article, Implementing campus-wide voice over Internet protocol (VoIP) phone systems at small colleges, they installed servers, switches, routers, phone hardware and power switches. They upgraded to a 1-gigabit backbone for Internet. They set over 1000 phones on this system.

Washington University in St.Louis, changed their phone system over to VoIP. Based upon their article, Designing VOIP in Campus Network, they installed servers, switches and routers had been recommended to use instead of hub and fiber optic instead of copper wire in backbone network. In the backbone network, Gigabit Ethernet had been implemented and mostly Fast Ethernet for other networks. On one link, since 80 Mbps maximum capacity was used at that time, another 12,166 lines could be used in Campus with G.729.

Rutgers, The State University of New Jersey, VoIP technology was implemented. Their environment was distributed and decentralized which subsequently fostered the growth of various systems. Each department or school within the University was given money from the Central Administration for telephone services. In some cases the department or school used those funds to purchase a more state-of-art system that enables them to pool telephone numbers thus reducing their monthly expenditures. However, this approach had resulted in over 350 different types of telephone systems distributed across the University. This model was becoming almost impossible to manage given the many different types of systems, voice mail, system software, etc.

3.4 Quality of Service

Quality of service (QoS) is something that can be different for each and every individual. For some quality can refer to how well the voice sounds over the phone, for others it may refer to the amount of noise they hear in the background, echoes, reaching an individual are all issues that go into quality of service. Another side of quality can refer to security of a conversation as well reliability. A study in India showed that the majority of people preferred the lower cost of service over responsiveness, value added services, reliability and voice quality.

Security or privacy of phone calls is another issue for QoS. This becomes exceptionally important for law enforcement officials. There are many differences between security of measures of public switched telephone networks (PSTN) and VoIP. Sicker and Lookabaugh [28]in their paper titled VoIP Security: Not an Afterthought used table 1 to show the differences between security measures.

3.5 Comparison of Costs

Which type of phone service is cheaper? PSTN or VoIP. Costs are more involved than a simple phone bill at the end of the month. Costs include hardware requirements, training costs, switch over costs, potential and loss of business in transition. Different companies will have different costs for telephone service based upon whether they are working on the international, national or local level. The company CISCO provides a lot of hardware for VoIP phone service and claims on their web site that companies have saved millions of dollars by using their technology.

4. Methodology

4.1 Analysis of Telephone System

Present telephone system in University of Dhaka, Mitel SX-2000 LIGHT system is analyzed by it's control cabinet, peripheral cabinet, links and power system as follows:

4.1.1 Control Cabinets

The Control Redundant SX-2000 LIGHT system supports applications that require up to 3000 lines. The redundant main control cabinet provides full back-up, including independent power supplies, so that system operation will not be affected if a main control component fails. When the system switches to the alternate main control, calls in progress are not dropped and callers are unaware of the system event. The redundant main control cabinet of this network has supported eleven expanded peripheral nodes.

Control Cards used in control cabinet

Main Controller Card

The standard Main Controller card for the SX-2000 LIGHT is the MC III E. In many features of this card, it is more important that it provides circuit switch matrix to establish voice and data paths from one peripheral device to another.

Circuit Switch Matrix Card

The Circuit Switch Matrix card is required for the control redundant SX-2000 LIGHT system. The Circuit Switch Matrix card increases the Main Controller card's circuit switch matrix size from 24 X 24 circuit-switched links to a 48 X 48 non-blocking link matrix.

Control Resource Card (CRC)

The Control Resource card (CRC) provides additional circuitry in the control node to support distributed system architecture. The CRC supports functionality between the Fiber Interface Modules and the Main Controller card. The CRC is non-redundant in all configurations; if power is removed from a CRC the System Fail Transfer becomes active at all peripheral nodes.

4.1.2 Peripheral Cabinets

Each peripheral cabinet holds up to 12 Peripheral Interface Cards and provides up to 192 ONS or DNI ports. By purchasing the Peripheral Node Expansion feature package, a slave cabinet can be added that expands the node up to a total of 384 ports and 22 Peripheral Interface cards (the number of voice channels remains the same).

Control Cards Used in Peripheral cabinet

Peripheral Resource Card (PRC)

The Peripheral Resource card (PRC) provides miscellaneous circuitry for distributed systems. The PRC is installed in all peripheral nodes. The PRC provides:

- System Fail Transfer contact closure
- Single ended to balanced conversion of FIM to DSU signals
- Terminal port multiplexer.

Peripheral Switch Controller (PSC)

The Peripheral Switch Controller (PSC) card is installed in all peripheral nodes. The PSC card provides control for all Peripheral Interface cards, and a fiber optic cable connects the FIM to the main control.

Peripheral Interface Cards

Peripheral Interface cards join telephone trunks and peripheral devices (such as SUPERSET telephones) to the system. Peripheral interface cards include line cards and trunk cards.

Line Cards

Line cards connect to single line sets, SUPERSETs, attendant consoles, and DATASETs. They include

- **Digital Network Interface (DNI) Line Card --** supports music-on-hold and paging and interfaces with MITEL digital network devices. (including SUPERSET telephones, attendant consoles, and DATASETs). The DNI line card provides 16 voice and data lines and has 16 circuits.
- **On-Premise (ONS) Line Card --** has 16 circuits that connect up to 16 standard telephones with line loop resistance usually not exceeding 400 ohms. It also supports modems and fax machines.

Trunk Cards

The system can connect to the public switched network or to private networks over both digital and analog trunks. Trunk cards provide an interface from the system to the public switched network and leased lines. Trunk card which is used in this system is:

- **Loop Start/Ground Start (LS/GS) trunk card --** interfaces to the analog LS/GS Central Office (CO) trunks, and is used to terminate eight CO trunks (non-dial-in trunks).

4.1.3 Fiber Interface Module

The FIM connects the control node to a peripheral unit. At the transmitting end, the FIM converts electrical signals into pulses of light to be transmitted over the cable. At the receiving end, the FIM converts the pulses of light back into electrical signals usable by the node.

Links

- single mode and for short distance multimode fiber is used between nodes
- 6/12 core fiber optic cable is use
- max. loss – 6 dB using 62.5/125 μm cable with N.A.-0.275 inch

4.1.4 Power System

Two redundant power modules in the redundant control node, one power distribution unit (PDU) in each peripheral cabinet (AC or DC) and one power converter in each peripheral cabinet (AC or DC) are used .In a DC powered peripheral ,the -48 V power is used directly. 21/19 plate battery backup is given in parallel, to keep the network active when the power system of any node fails.

4.1.5 Users handling capability of this network

From the brief description of Dhaka University telephone network the total output lines which can operate different types of peripherals, can be easily obtained as follow:

No. of peripheral nodes: 11

No. of slots in each peripheral node: 12

No. of circuits in each ONS line card: 16

So, the total voice and data lines for 11 peripheral nodes: 11*12*16 = 2112

Through extender these 11 peripheral nodes can be extended up to 22. That way, this network has the capability of handling 22*12*16 = 4224 voice and data lines. But this network is using 2100 voice and

data lines and the no. of trunk lines= 180 i.e., the no. of users of these network is- 2280.

4.2 Convergence of Existing PABX System & IP PABX System

At Mitel Networks convergence of two networks is possible by using Mitel networks 3300 Integrated Communication platform. The 3300 ICP is a resilient network appliance that adds feature rich IP telephony and advanced user applications to the corporate LAN/WAN. The 3300 ICP offers the ability to migrate from an existing SX-2000 LIGHT PBX system to an IP PBX system. The database is converted and restored onto the 3300 ICP by using the Mitel Networks 3300 Configuration Tool. Mitel Networks' architecture uses the IP network for connecting IP telephony devices and provides a supplementary TDM (Time Division Multiplexing) subsystem for switching calls between traditional telephone devices. The 3300 ICP has the advantage of being able to optimally switch all types of traffic, IP or TDM.

4.3 VoIP Perspective

VoIP is the routing of voice conversation over the internet. As packet switched networks were designed to carry data, while carrying voice, calls might experience delays and distortion. While making a regular call, subscribers are charged against distance and duration of the call, in this case these factors do not matter, rather a low fixed rate price is charged for the internet bandwidth. A simple model of VoIP is given in figure 3.

4.4 Upgrading of SX 2000 LIGHT

To upgrade the SX 2000 LIGHT, two 3300 MXe Expander is used to replace the control cabinet. The suggestive system topology of Dhaka University campus is shown in figure-4.2

4.5 Planning and Designing VoIP

The issues, related to designing an IP telephony or voice over IP (VoIP) network for transporting voice and data over a common LAN or WAN infrastructure are covered in this section. Understanding the underlying technology used to transport voice traffic is important in designing an IP telephony network. Design principles used to deploy a successful LAN-based VoIP network will not necessarily work when you apply them to a WAN configuration. This document discusses the major hurdles that need to be addressed when designing either a LAN or WAN based VoIP network.

4.5.1 Bandwidth Management

Where bandwidth is at a premium voice compression is a requirement and now widely available voice coding algorithms/compressors are G series codec. Each voice codec has a benchmark score (MOS) based on- speed of the conversion, speech quality, and data loss characteristics. To get the maximum MOS value and to achieve the best quality for voice traffic G.729 is chosen. Table 2 shows standard Codec Compression for G.729.

The capacity calculation using G.729 series is given below:

Header = Ethernet Header(18)+IP Header(20)+UDP(8)+RTP(12)

= 58 bytes

Datagram Payload Size :

$$\frac{\text{Codec Speed(bits/sec)*Datagram Delay(ms)}}{\text{(8 bits/byte)*1000(ms/sec)}}$$

= 8000*20/(8*1000) = 20 bytes

Packet Size = 78 bytes

Overall Capacity =(1/20ms)*78*8 = 31.2kbps

Using G.729 encoding series on a 100 Mbps Ethernet network, each voice call takes up to 31,2 kbps

in each direction supporting up to 3205 calls on full duplex link. On a Gigabit backbone, up to 32050 simultaneous calls can be handled. The 2 MXe expander ICP 3300 controller can support upto 2800 IP phones which will need the capacity of 31.2 * 2800 = 87.36 Mbps.

4.5.2 QoS Design

The major factors to tune QoS – Congestion control, Reliability , Throughput, Delay , Jitter

To ensure all the factors -

- Traffic conditioning is mandatory.
- Differentiated Services (DiffServ) is to be provided.
- Delay guideline is to be followed

Table 3 shows the mechanisms for traffic conditioning and the respective network effects.

4.5.3 Delay Guide Line

To make better quality following conditions are must be fulfilled:

- End to End Delay: < 150 ms
- Jitter: < 40 ms
- Lost Packet :< = 0.5 %
- reduce the -TCP/IP window size and MTU size
- Finally - the settings of the QoS on the router is done by configuring it accordingly

4.5.4 Queuing

In case of that there are so many packet waiting the Router queue, WFQ (Weighted fair queuing), CBWFQ (Class-based weighted fair queuing), LLQ (Low-latency queuing), and WRED (Weighted random early detection) could be configured to the latency of voice packet. Figure 4 shows an example of CBWFQ router configuration.

4.5.5 Traffic Shapers

This method could be applied for the inter-network connection usually because the network link is limited by ISP (external link) (There are too many users required these connection in the same time) For example, in TCP connection, whenever the packet gets lost, retransmission process will be applied again and again. As a result of this, it may need to shape the connection in order not to waste the capacity for retransmission. Figure 5 shows the traffic shaping configuration example.

4.5.6 TCP/IP Tuning

- TCP window size: It is better to reduce the TCP/IP window size in busy network.
- MTU size and low latency: due to the smaller VOIP packet, it is better idea to set MTU path along the path to fit in the packet in slower-speed link Also, Routers have no need to pay for fragment delay.

Figure 7 shows MTU (maximum transfer unit) configuration example.

4.6 Topology Design

Topology is designed as it has a 100 Mbps Ethernet backbone with equipments- 7204 VXR router, 1900 series switches and links - Cat 5 cables, DSL, ADSL, optical fibers. The IP network topology is given in figure 8.

4.7 Convergence Strategy

The 3300 MXe Expander located at the Register Building and a 1900 switch located at the IIT has been connected. Since the distance between the two location is nearly 1 km, the interconnectivity by UTP cable directly not possible. So media converters are used which supports network extensions upto kilometer range. LH1706A-ST-US media converter uses fiber optic cable and can extend upto 2 km.

Manager and the application server are connected using their Ethernet ports via local switches using UTP cables. This topology consists of three parts : the telephone network, the IP network and their interconnection. The positions of the nodes and the distribution boxes are given below:

1. Control node- 3300 ICP Controllers

(MXe Expander) : 2 – Administrative Building

2. Peripheral nodes : Total – 15

a. Business Studies –3 nodes

2 distribution boxes

b. ISWR –1 node

1 distribution box

c. Curzon Hall – 2 nodes

1 distribution box

d. Khondokar Mokarram Hussain Building – 2 nodes

1 distribution box

e. Arts Building – 2 nodes

1 distribution box

f. Register Building – 4 nodes

3 distribution boxes

g. Science Annex – 1 node

1 distribution box

Figure 9 shows the IP PABX system topology i.e. the topology of the converged network.

5. Findings

➢**Cost effectiveness-**Subscribers are charged a low fixed rate price for the bandwidth, so long distance calls are very cheap.

➢**Capacity enhancements-**By connecting more ICP 3300 controllers, IP phones and computers, the capacity can be increased without further network congestion.

➢**Redundancy-**In fail over mode the users can use the analog line by pressing a programmed key, when the IP connection has failed.

➢**Plug and Work solution-**This solution enables access to the voice network of DU from any location with a broadband internet connection.

➢**Improved media integration-**IP phones can be enabled to add media to an ongoing call as required, e.g., viewing a picture or drawing on a whiteboard. Using workstations themselves as IP phones can facilitate providing this function, whereas the standards are not yet there for coupling traditional phones and workstations.

➢**New services-**As IP Telephony evolves, it can be used to provide new services (like user-defined call processing) or to integrate existing concepts, e.g., Presence, Location Awareness or Instant Messaging. Because of the open standards available for these services, they need not to be limited to vendor-specific solutions. In other words, it can be much easier to deal with issues such as CTI (Computer Telephony Integration) and so pave the way to a completely new way of understanding telephony.

➢**Research-**The protocols and standards used for IP Telephony are open and publicly available. This allows research institutions to work on their own services and solutions. It is important to point out that before introducing IP Telephony into the network of an organisation; several issues unknown to the old telephone system have to be taken into account. A rough, non-exhaustive list may include addressing (special subnet/VLAN for phones), Quality of Service (QoS), security, positioning of gateways, interfacing of firewalls and, last but not least, maintenance of the system (backups, spares, etc., - something not very common in the legacy PBX world).

6. Recommendations

At present Dhaka University telephone network can support upto 4224 telephone lines and the designed IP PABX system has a capacity of 2800 lines. And the value to an organization in unifying communications is to improve workflow and efficiency, ultimately leading to better service delivery and an improved bottom line. So, the ultimate aim of unifying communication can be the complete migration of the legacy telephony to IP telephony in the coming days.

In this network very high capacity fibers have been used whereas 2 or 3 core fibers are adequate for the system. The rest of these unused fibers (dark fibers) can be utilized by using them as links for other purposes. Such as a very high speed internet connection at low cost can be provided by connecting these fibers to the 2 layer switches of our network and a SEA-ME-WE cable.

Again, if wave division multiplexing is used in fiber optic communication, the transmission rate can be increased to 1.1 trillion bits per second (1.1 Tbs) over 150 km and 2.6 Tbs over 120 km using 132 different wavelengths in the interval 1529.03-1563.86 nm (European Conference on Optical Communication). Thus, in this case independent signals are carried by a single fiber multiplying the capacity of an individual fiber making it capable of carrying enormous rates of information.

7. Conclusion

This report is meant to serve as a catalyst for discussion relative to VoIP applicability and deployment at the University. It is not exhaustive in content but certainly inclusive of major considerations when contemplating an implementation. As with every new technology, there are caveats, special considerations and different skills required to manage them. Some of these, such as shared administration, power considerations, disaster recovery, repair and troubleshooting, localized bandwidth consumption and monitoring capabilities are quite significant in scope.

The difference in a straightforward or “vanilla” installation and one that might occur in a mixed assets environment is dramatic. Clearly the interoperability of all environments, the re-use of existing technology and the economics of deployment are the decision variables. VoIP is not free, or cost effective in all cases. It is most efficient in plain vanilla cases when traditional systems are displaced for a variety of reasons, including obsolescence or high maintenance cost. At the University, we are in the throes of trying to establish unified voice architecture, while leveraging viable assets so that we could achieve a technologically sound solution, which is also economically feasible. Hopefully this report will help spur useful discussion in our quest to deliver high quality, maintainable phone service throughout the University.

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Table 1. Differences between security measures

Security Concerns	**Wired PSTN Measures**	**VoIP Measures**
Confidentiality	Physical security	Encryption techniques
Integrity	Physical security	Encryption techniques
Availability	Physical access control	Network/Service access control
Authentication	Physical connectivity, voice recognition, caller ID	Login, password
Authorization	Caller ID, access control	Access control role-based authorization
User Expectation	Assumed and static	Variable
Implementation and Design Concerns		
Software design	Large, monolithic, complex	Variable, distributed, complex
Interoperability	Centralized and tested	Distributed and potentially ad hoc
Software implementation	Centralized and tested	Distributed and potentially ad hoc

Table 2. Standard Codec Compression for G.729

Codec	Data rate	Packetization delay	Jitter Buffer Delay	Datagram spacing	MOS	Total capacity required
G.729	8 kbps	25 ms	40ms(2)	20 ms	4.07	31.2kbps

Table 3. Traffic conditioning and mechanism

Traffic conditioner	**Mechanism**	**Network effect**
Marking	IP Precedence, DSCP, CoS	• Sets IP precedence/ DSCP • By apps, protocol, address, etc
Policing	CAR, Class Based	• Enforce a maximum transmission rate • Conform or exceed threshold
Scheduling	PQ, CQ, WFQ, LLQ, WRR, MDRR	• Bandwidth management: traffic priority • Set service sequence
Shaping	GTS, FRTS	• Conforms traffic to commited bandwidth • Interwork L2 notification, BECN
Drop	RED, WRED, Flow RED	• Avoid congestion by notifying source • Prioritize which traffic is told to reduce
Compress	CRTP	• Reduce the volume of traffic sent
Fragment	LFI, FRF.12	•Reduce delay on slower-speed links • Split, recombine larger frames

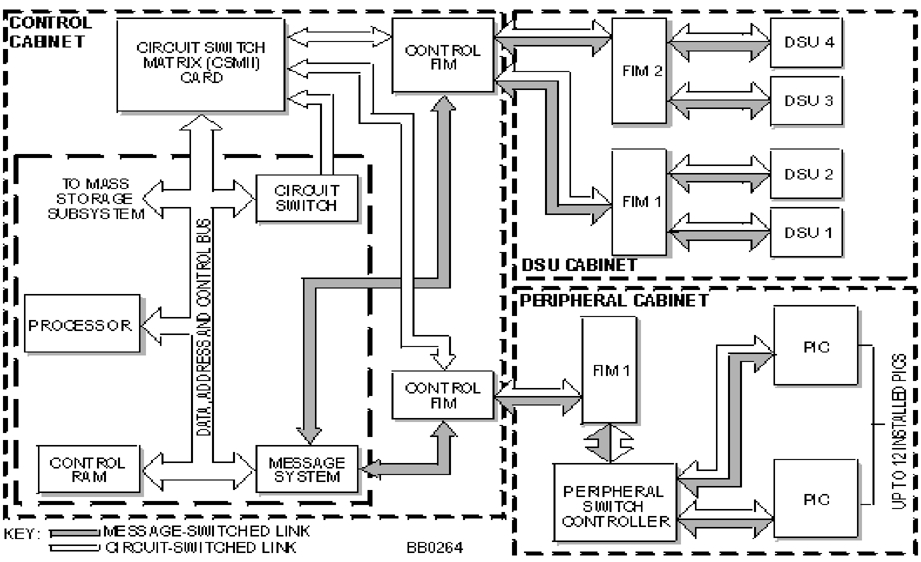

Figure 1. Basic System Architecture

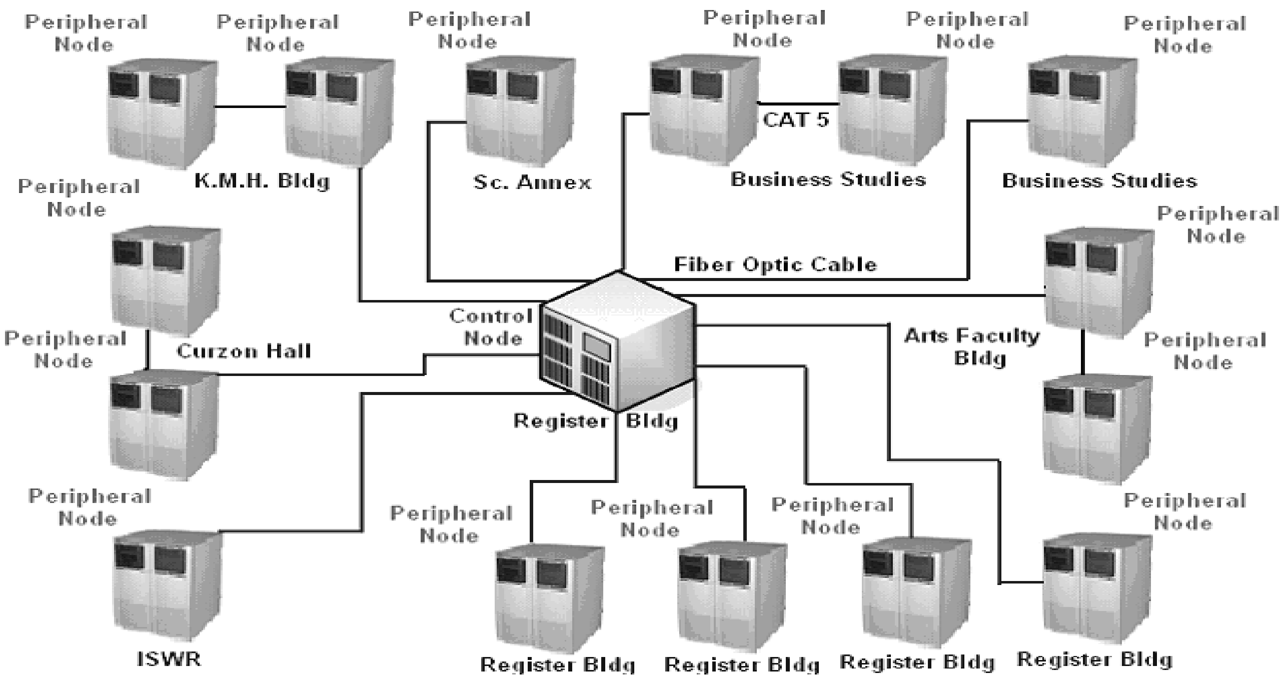

Figure 2. PABX system topology in Dhaka University campus

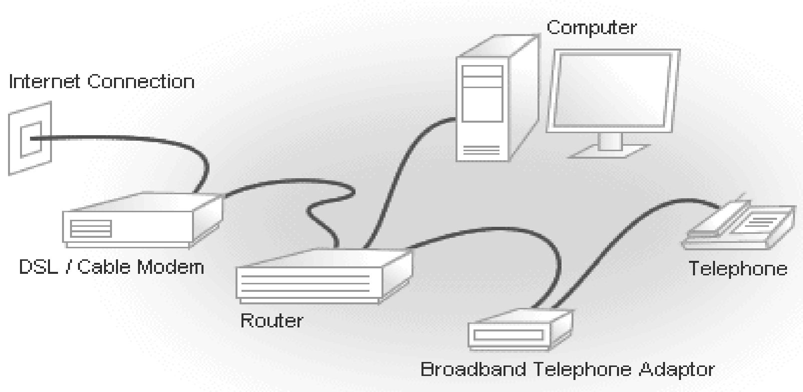

Figure 3. A simple VoIP model

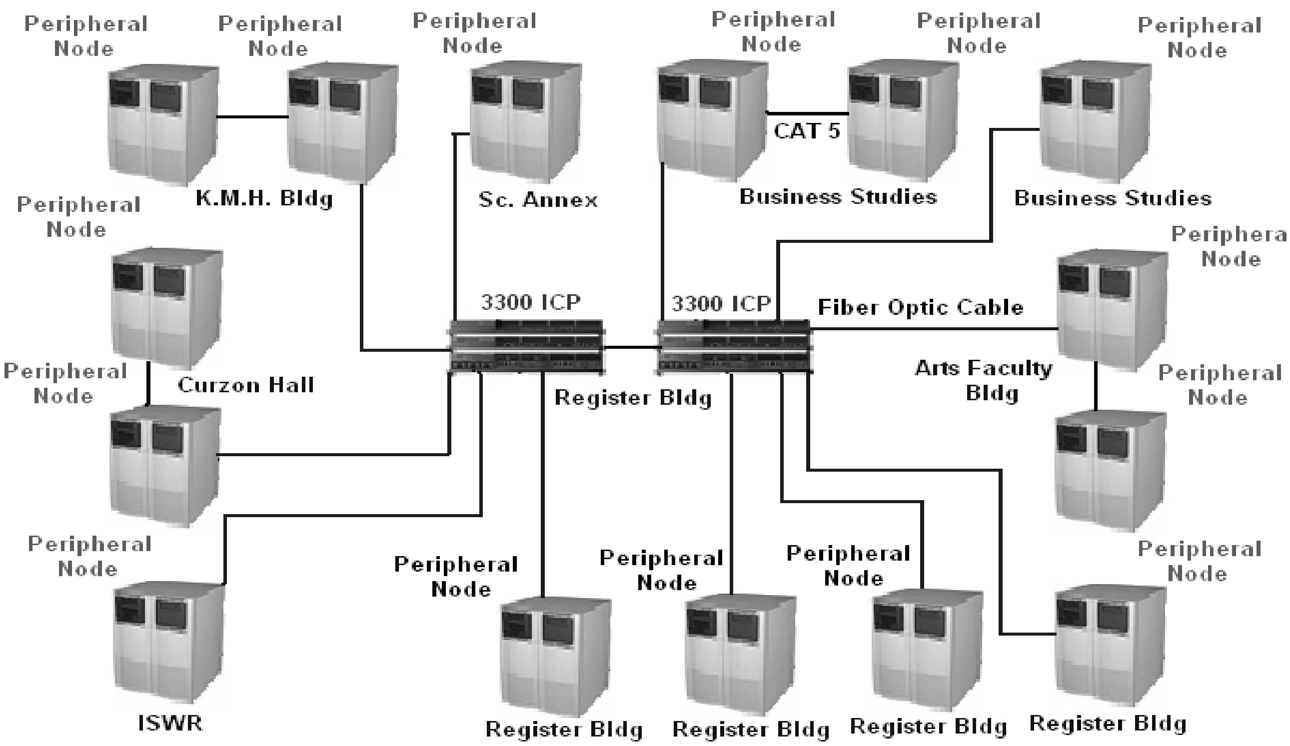

Figure 4. Suggestive system topology in the campus

```
Router(config)# policy-map shape-cbwfq
Router(config-pmap)# class cust1
Router(config-pmap-c)# shape average 384000
Router(config-pmap-c)# capacity 256
Router(config-pmap)# class cust2
Router(config-pmap-c)# shape peak 512000
Router(config-pmap-c)# capacity 384
Router(config-pmap-c)# configure terminal
Router(config)# interface Serial 3/3
Router(config-if)# service out shape-cbwfq
```

Figure 5. CBWFQ Configuration Example (Cisco Network)

```
interface <serial interface or sub-interface>
        traffic-shape rate 64000 8000
46320000
interface <LAN interface>
traffic-shape rate 64000 8000 46320000
```

Figure 6. Traffic Shaping Configuration Example (Cisco Network)

```
interface pos 0/0
mtu 1500
interface pos 0/0
ip mtu 1500
```

Figure 7. MTU configuration example

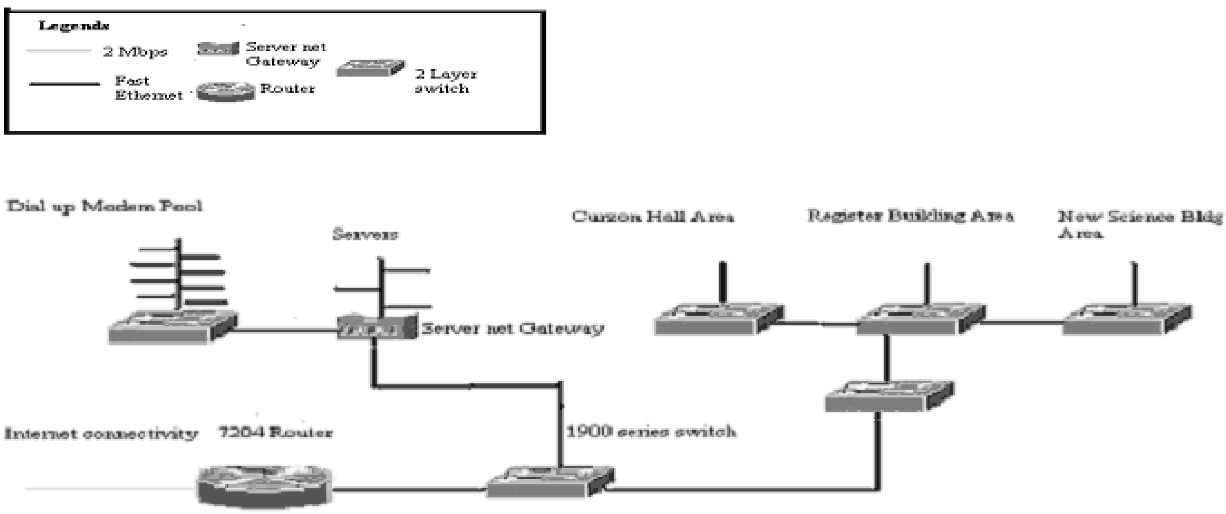

Figure 8. IP network topology

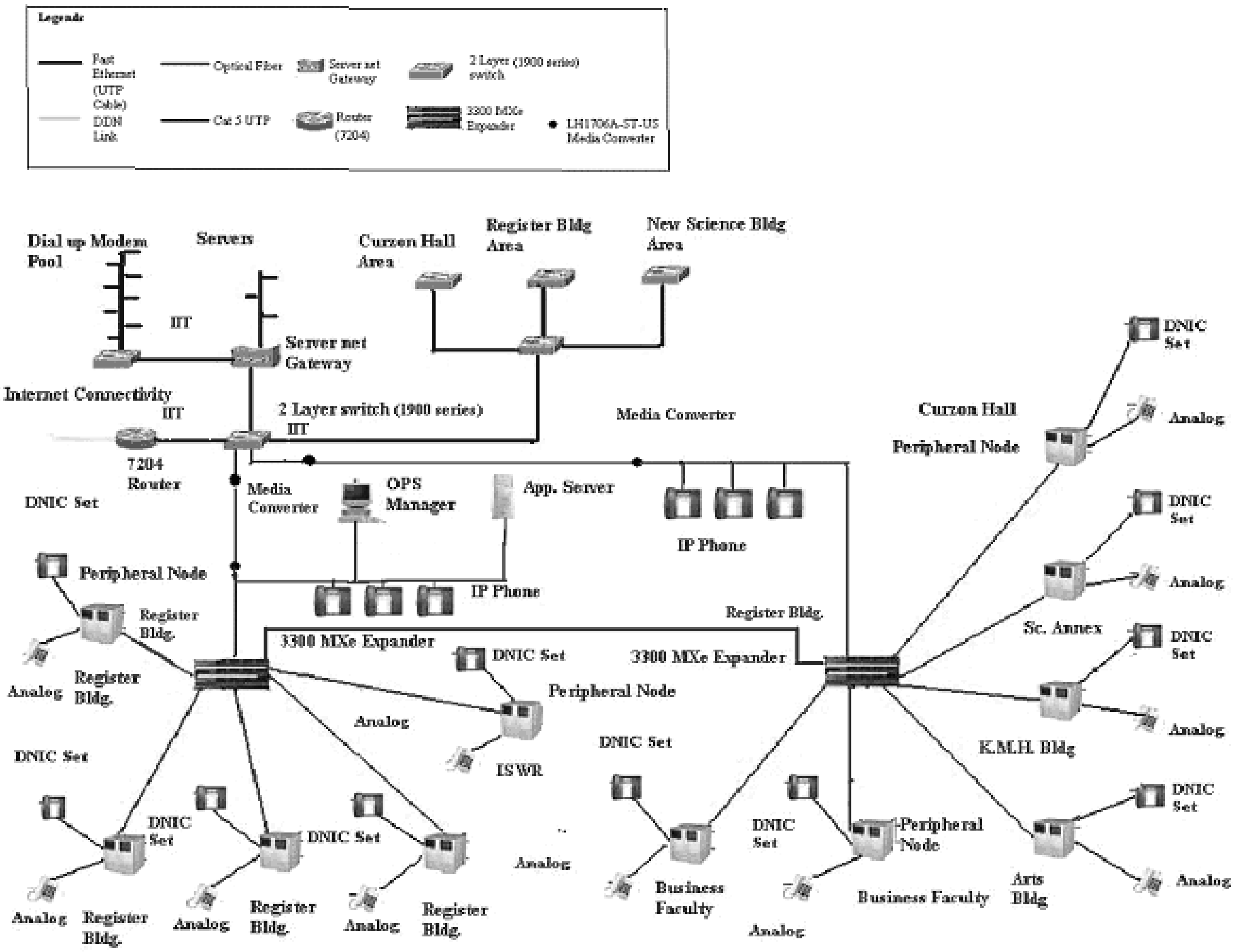

Figure 9. Converged network topology