Designing VOIP in Campus Network

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1. Introduction

1.1 Telephone System

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 connections. This connection is the dedicated service that can guarantee the reliable, accessible, and quality. For example, you could talk as long as you want to talk if no care about the bill or you hear exactly whatever your friend said. [2]

1.2 Telephone Standard

International Telecommunications Union (ITU) is a nonprofit international organization that takes care and maintains the telephone standard. Many kinds of standard come up with the assigned letter of the alphabet standard such as G.729 or H.323.

G-Transmission systems and media, and digital systems and networks
H-Audiovisual and multimedia systems
P-Telephone transmission quality, telephone installations, and local line networks

1.3 How PSTN and VOIP work? [1]

PSTN :

- Whenever making a call, pickup the phone and listen a dial tone from the local office (telephone carrier) if there is no dial tone, just hang up and try it again.
- Dial the number
- This signal will be routed (in-band) to the switch to find the path going through the destination; as a result, the destination phone rang.
- Whenever somebody answers that phone, the connection circuit is completely established. (this circuit will be as long as you don't hang up)
- The circuit will be terminated and free up the resource whenever you hang it up.

VOIP :

- Soft phone or IP phone could be represented as telephone itself. Like PSTN system, dial tone will come from the VOIP service provider.
- Dial the number
- In this case VOIP service provider have the switch to map the IP address (source and destination) and Local number and the inter-switch will be used to reroute the packet to final destination
- Whenever it reaches the destination, the phone rang and somebody answer.
- There is no permanent circuit between them. Some packet will be sent back and forth during the conversation. (Usually, the packet will be sent by the same path to maintain jitter)

- If no one talks, there is no packet sent and this network can be used from other people.

1.4 PSTN and VOIP Components

Generally, there are mainly four components that define the telephone system requirements. (Table 1) In each layer, the different standard components between PSTN and VOIP will be represented; for example, PSTN Switching is used for PSTN but VOIP Router and Gateways for VOIP.

Circuit Switching (PSTN)	Packet Switching (VOIP)
Voice Encoding (PCM)	Voice Encoding (G Series)
PSTN Switching (LEC,IXC-> Class 1,2,3,4, and 5 Switch) : Multiplexing	VOIP Router and Gateways
Private Branch Exchange (PBX) :	IP PBX & IP telephone Server
Signaling (SS7) : Out of band	Signaling : (STP-IP Router, SCP-DNS&DHCP) VOIP Protocol
Telephones (Analog & Digital)	IP Phones and Soft phone

Table 1: PSTN and VOIP Components

- **Voice Encoding**

Digital Voice Signal (PCM) Analog->Digital is used to make the digital voice signal. Normally, human voice has a frequency range at 4000Hz (Analog Signal). According to the Nyquist Sampling, to guarantee the quality of voice, the signal has to be sampling twice as many as the original signal. Also, the quantization should be at 256 levels that are 2^8, so 8 bit will be used for this process.

Total Capacity per one user call = 4000Hz*2*8bit = **64 kbps**

As a result of this theory, in PSTN, the total capacity will be the factor of n (n users) multiply 64 kbps (100 users making call simultaneously, 6.4 Mbps will be used) However, to save the capacity consuming (VOIP), there are other techniques to code and compress human voice. (Table 2) [2] Although, it seems some standard can compress data up to 12 times, the computation overhead is paid as well; for example, powerful CPU and Memory requirement.

Codec	Data Rate	Packetization Delay	Jitter Buffer Delay	Datagram Spacing	MOS	Total Capacity Required
G.711u	64.0 kbps	1.0 ms	40 ms (2)	20 ms	4.41	87.2 kbps
G.711a	64.0 kbps	1.0 ms	40 ms (2)	20 ms	4.41	87.2 kbps
G.726-32	32.0 kbps	1.0 ms	40 ms (2)	20 ms	4.22	55.2 kbps
G.729	8.0 kbps	25.0 ms	40 ms (2)	20 ms	4.07	31.2 kbps
G723.1 MPMLQ	6.3 kbps	67.5 ms	60 ms (2)	20 ms	3.87	21.9 kbps
G723.1 ACELP	5.3 kbps	67.5 ms	60 ms (2)	20 ms	3.69	20.8 kbps

Table 2: Standard Codec Compression

*Datagram Spacing = each packet will be sent in every 20ms
*Packetization Delay = the coding and compression delay in Codec itself
*MOS (mean opinion score) is the standard metric for user perception of call quality

For example (G.729):

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

Figure 1: G.729 capacity requirement example

- **VOIP Gateway & Router : Transfer RTP voice through IP network**
  - VOIP Gateway use SS7 protocol to interconnect with traditional PSTN network, and also use “transcoding” technique between different codec for example, G.729<->G711.
  - Router and Switch are originally used for the packet switch network to move the data through destination.

- **IP PBX**
  - IP PBX usually provides the similarity with the same PBX-PSTN.
  - GateKeeper : (H323 protocol to provide Call admission Control and other management functions such as address lookup)

- **VOIP Protocol**
  - Call setup protocol (H.323, SIP, SCCP, MGCP, and Megaco/H.248)
    SIP is currently supported by most VOIP device companies such as Cisco, and Nortel and also, Microsoft bundle SIP client interface with Windows XP OS.
  - Voice streaming protocol (RTP)

Ethernet (18B) Header	IP (20B) Header	UDP (8B) Header	RTP (12B) Header	RTP (Varies B) Payload

Ver (4b)	Pad (1b)	Ext (1b)	CSRC Cont (1b)	Marker (1b)	Payload Type (7b)	Sequence (2B)	Time Stamp (4B)	Source Identifier (4B)

Figure 2: VOIP frame requirement (B=bytes, b=bits) a. Ethernet datagram
b. RTP datagram

- **IP Phone and Soft phone**

To make users simply use the VOIP system, IP phone (hardware) will be designed to substitute the original phone but the different is the connection media. RJ45 is used instead of plugging to RJ11. Also, for advanced users, there is no need for physical phone if you could access online with microphone and speaker. Soft phone is designed for this purpose.

1.5 VOIP Benefits and Obstacles [3]

Although it seems VOIP is the new coming system that provides a lot of advantages, it has some drawbacks as well. Designing and pre evaluation of VOIP will be needed before implementing this network. (Mostly it is because the nature of data and IP network itself)

Advantages:

- Cost Saving
  - Centralize Management and Single Network (Data and Voice)
  - Long-Distance Service Calling Saving
- New Feature Upgrade
  - Unified Messaging
  - Advanced Call Routing
- Facilitate to add, move, and change the telephone (with DHCP enabled)

Disadvantages:

- Reliability- Can not guarantee QOS (Latency, jitter, lost packet)
- First time investment
  - ROI (Return of Investment): This metric is used to measure to be worthwhile for the long run.

VOIP ROI Estimate	**Year1 (2^{nd} Q)**	**Year1 (4^{th} Q)**	**Year2 (2^{nd} Q)**	**Year2 (4^{th} Q)**
Total costs per year				
Total returns per year				
VOIP Planning & Evaluation				
Personal Costs (IT&Training)				
Capital Costs (Hardware/Software/Others)				
Outsourcing (Consultants/SA)				
VOIP Returns & Estimate				
IT capital saving				
IT recurring costs				
IT Staff productivity improvements				
Telephony capital savings				
Telephony recurring costs				

Table 3: VOIP ROI estimate [2]

1.6 Call Quality

The main problem for implementing VOIP is all about the voice quality because it is not the permanent circuit like PSTN. Everything is all share network so to make better quality, some parameters have to be restricted; for example, End to End Delay: <150ms, Jitter: <40ms, and Lost Packet <=0.5%. Table 4 shows two satisfactions metric; MOS is used for PSTN network and R for VOIP Network.

Satisfy	Delay (ms)	Condition
Acceptable	0-150	Good quality and no echo
Acceptable under conditions	151-400	Acceptable under certain conditions and echo canceling is needed
Unacceptable	401+	Unacceptable for real-time voice traffic and planning and testing purposes only

Table 4: Delay Guideline ITU-T G.114

User Satisfaction	MOS	R
Very Satisfied	4.3	90
Satisfied	4.0	80
Some Users Dissatisfied	3.6	70
Nearly All Users Dissatisfied	3.1	60
Not Recommended	2.6	50

Table 5: MOS and R (Call Quality Metric)

*In perfect condition, R=R0 and in any condition, R=R0-Is-Id-Ie+A
Is=Simultaneous impairments to the signal
Id=Delays introduced from end to end
Ie=Impairment introduced by the equipment, including packet loss
A=Advantage factor

2. Planning and Designing VOIP

Before implementing the VOIP network system, the planning and designing process is necessary. It is not because to measure if it is worth to invest or how much the money has to be spent but also to plan in advanced to use this technology.

2.1 Capacity Design

The first factor that has to be investigated is the capacity of telephone usages; for example, Number of Calls (Peak) => 4,500 Calls (External and Internal) (Figure 3 and 4), and also the number of network usages. Also, according to Ethernet recommendation (IP over Fast Ethernet), the maximum capacity which could be used efficiently is around 60%. Table 5 shows the number of maximum line usages in different coding methods (1Gbps Faster Ethernet Reference)

Codec	Data Rate (kbps)	Total Capacity (Mbps)	Line (1Gbps) Capacity (60%) left
G.711u	87.2	392.4	2,380.7
G.711a	87.2	392.4	2,380.7
G.726-32	55.2	248.4	6,369.6
G.729	31.2	140.4	14,730.8

Table 6: Codec Capacity Requirement (after 4,500 lines deduction)

Furthermore, to foresee the calling and network usages when the company has grown up, the VOIP designer should collect the information from the users in order to both easily upgrade in the future and choose the VOIP component to fit the requirement with minimum cost. Figure 5 represents network usage by Day, Week, Month, and Year regularly basic. In this example, mostly the network traffic is around 80 Mbps so if the maximum capacity is up to 320 Mbps, 240 Mbps is still left. G.729

is implemented for this network, presently another 460 telephone could be supported by not implementing new network infrastructure.

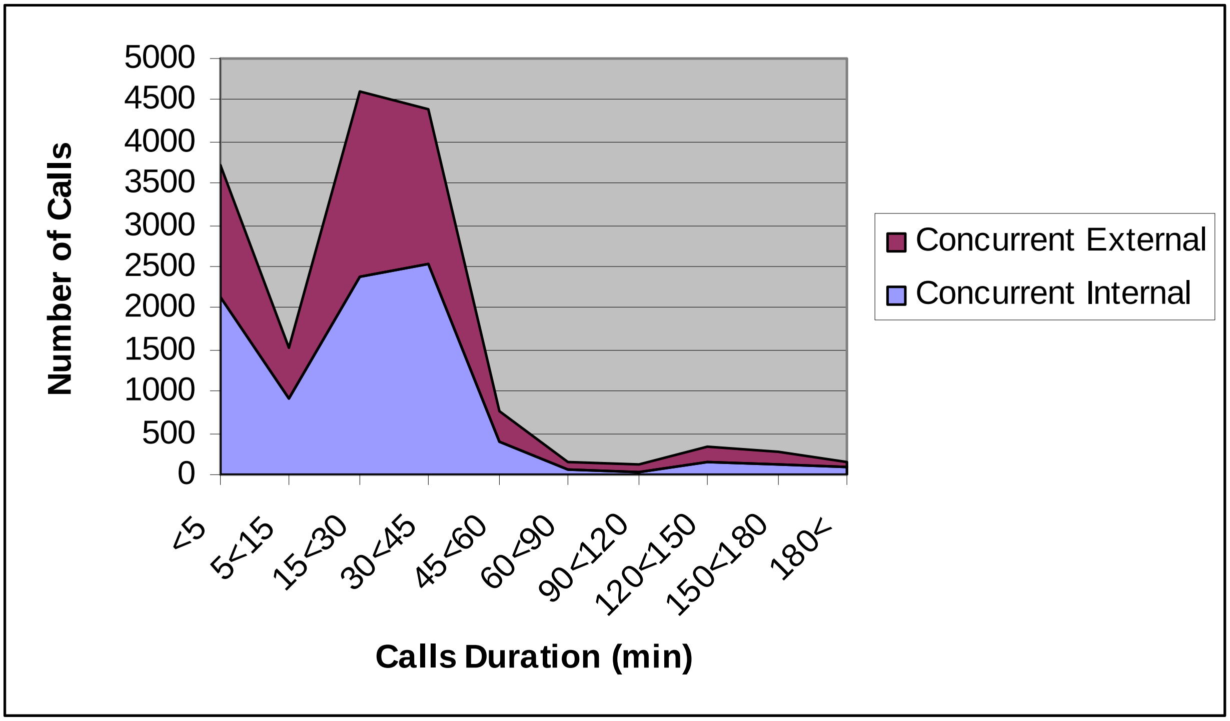


Figure 3: Statistic Example of usages per day (Calls Duration)

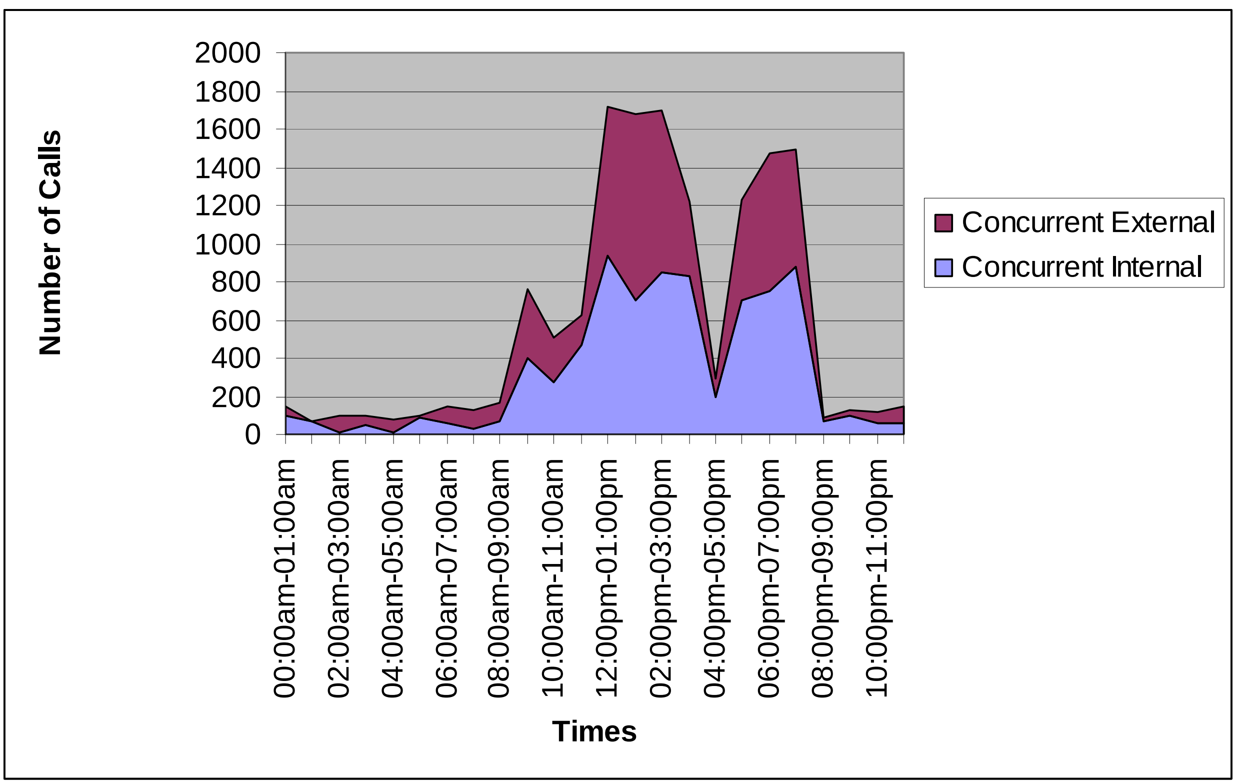
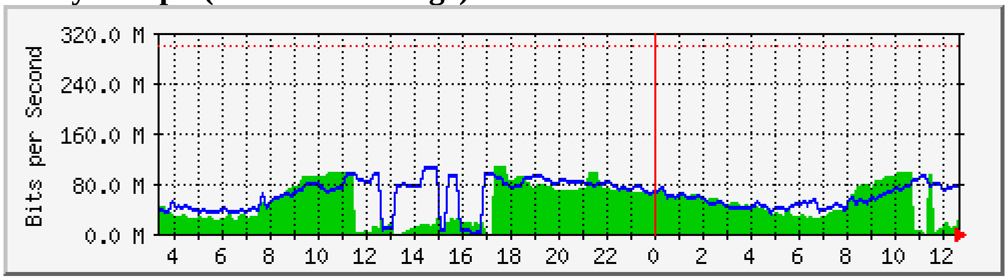


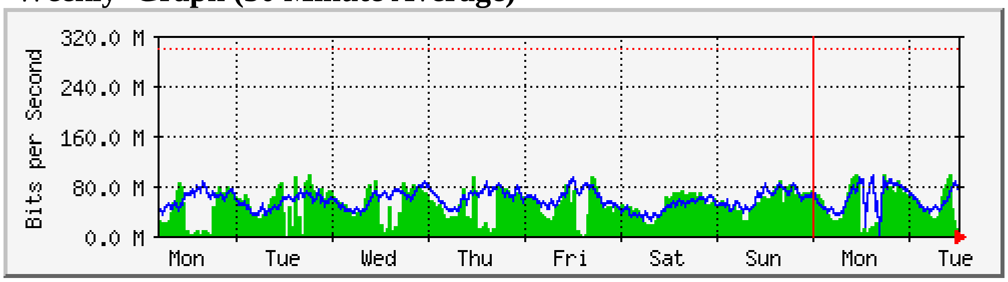
Figure 4: Statistic Example of usages per day (Number of Calls)

`Daily' Graph (5 Minute Average)



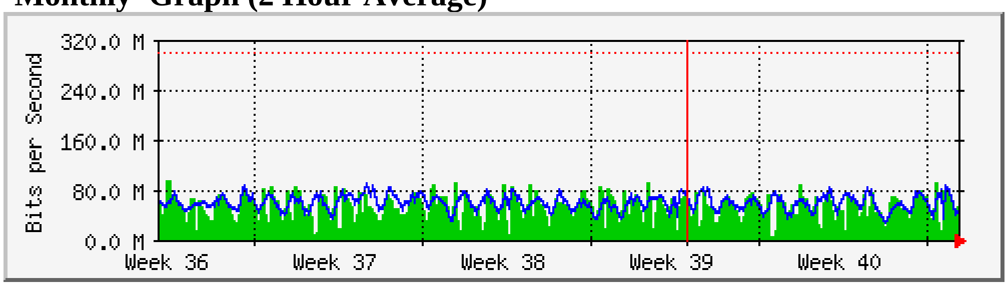
Max In: 111.2 Mb/s (37.1%) Average In: 50.7 Mb/s (16.9%) Current In: 24.6 Mb/s (8.2%)
Max Out:107.9 Mb/s (36.0%) Average Out:63.2 Mb/s (21.1%) Current Out:79.6 Mb/s (26.5%)

`Weekly' Graph (30 Minute Average)



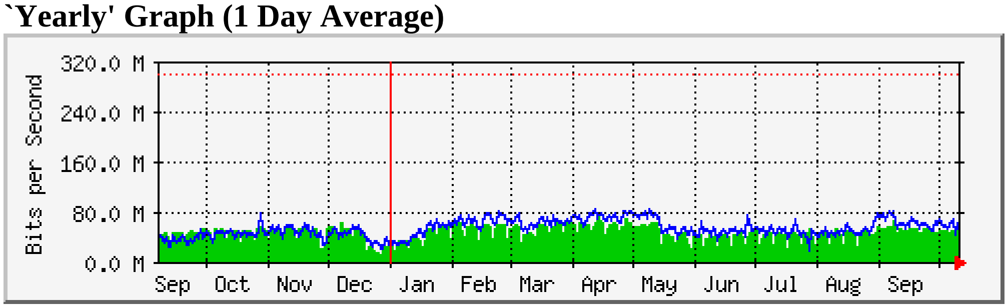
Max In: 101.1 Mb/s (33.7%) Average In: 50.0 Mb/s (16.7%) Current In: 14.6 Mb/s (4.9%)
Max Out:98.1 Mb/s (32.7%) Average Out:60.3 Mb/s (20.1%) Current Out:76.0 Mb/s (25.3%)

`Monthly' Graph (2 Hour Average)



Max In: 98.7 Mb/s (32.9%) Average In: 52.0 Mb/s (17.3%) Current In: 55.0 Mb/s (18.3%)
Max Out:91.4 Mb/s (30.5%) Average Out:61.3 Mb/s (20.4%) Current Out:49.6 Mb/s (16.5%)

`Yearly' Graph (1 Day Average)



Max In: 74.8 Mb/s (24.9%) Average In: 47.3 Mb/s (15.8%) Current In: 51.4 Mb/s (17.1%)
Max Out:83.8 Mb/s (27.9%) Average Out:55.1 Mb/s (18.4%) Current Out:61.6 Mb/s (20.5%)

Figure 5: Network Statistic (MRTG)
Green : Incoming Traffic in Bits per Second
Blue : Outgoing Traffic in Bits per Second

2.2 Topology Design

The second parameter for VOIP Design is the network topology and equipment. Network along the path that telephone system will be implemented has to be investigated. Network Switch and Router have been recommended to use instead of Hub. It will be better to implement fiber optic instead of copper wire in backbone network. Also, after collecting the network statistic above, some network infrastructure will have to be upgraded. Figure 6 shows the network infrastructure of Washington University in St.Louis. In the backbone network, Gigabit Ethernet has been implemented and mostly Fast Ethernet for other networks. From Table 5 and Figure 5 on one link, since 80 Mbps maximum capacity is used at that time, another 12,166 lines could be used in Campus with G.729.

IP Addressing Scheme is another important scheme whenever designing the VOIP network. Because the nature of Voice itself is delay-sensitive, it would be better to group the Voice Network to another VLAN (Campus Network). Not only it is used to provide the Voice efficiency (QOS), but also it will be easily for VOIP Management.

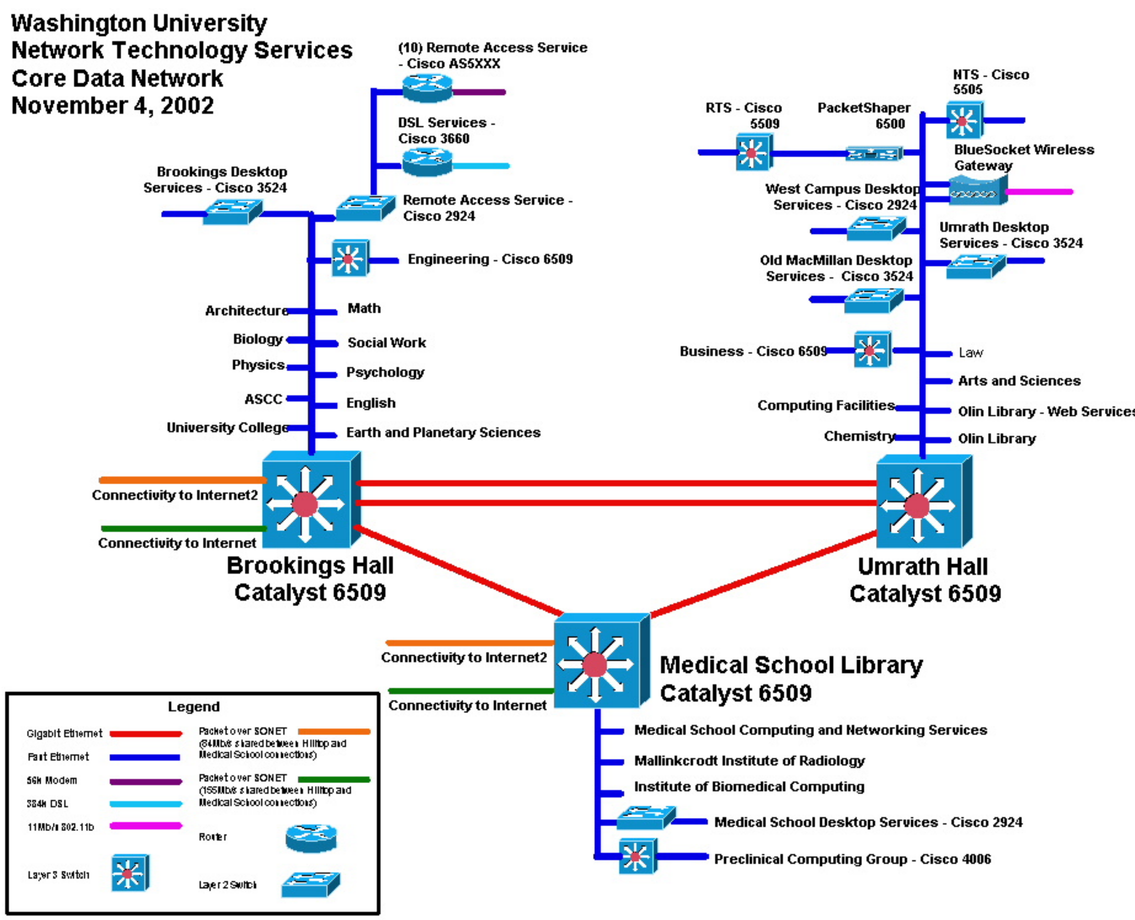


Figure 6: Network Topology (WUSTL) [3]

2.3 QOS Design

The third main issue about VOIP design is about quality of service of VOICE. Maybe it is the main obstacle why VOIP network will not be implemented world widely. There are many factors that QOS could be tuned to at least fit with users' satisfactions.

2.3.1 QOS Traffic

QOS could classify to many factors in different level (At the edges of network, In the middle of network, and at the end of network) and different kinds of solution have been proposed; for example,

- At the edges of a network: Usually, it is in the external link-WAN long distance so traffic shaper or access router may be used for this purpose.-> *Traffic Shapers and Link-layer QOS*
- In the middle of the network: Normally, in campus network, there is plenty of capacity such as Gigabit or FDDI but the Router or even Switch itself has to be concern about Queuing, Buffer, and Networking Delay.-> *IP QOS and Queuing*
- At the end network node: Although VOIP network recommend users to run on 100 Mbps network but sometimes there are so many consuming capacity application running on the same network; for instance, database replication and backup application. The congestion will be on some period that may affect voice packet. Also, the sharing and broadcasting protocol, NETBIOS for example has direct impact to other computers in the same network. -> *Tuning Policy*

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 thresholds
Scheduling	PQ, CQ, WFQ, LLQ, WRR, MDRR	◆ Bandwidth management: Traffic Priority ◆ Set service sequence
Shaping	GTS, FRTS	◆ Conforms traffic to committed bandwidth ◆ Interwork with 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 7: Traffic Condition and Mechanism [7]

Voice over IP CiscoCTI Clarent CUSeeMe Dialpad H.323 I-Phone MCK Commun. Megaco Micom VIP MGCP Net2Phone RTP	**Client/Server** CORBA CVS Folding@Home FIX (Finance) Java Rmt Mthd MATIP (Airline) MeetingMaker NetIQ AppMngr OpenConnect JCP SunRPC (dyn port) **Content Delivery** AOL	**Directory Services** CRS DHCP DNS DPA Finger Ident Kerberos LDAP RADIUS SSDP TACACS WINS whois	**Healthcare** DICOM HL7 **Host Access** ATSTCP Attachmate SHARESUDP Persoft Persona SMTBF TN3270 TN5250 **Legacy LAN and Non-IP** AFP AppleTalk	**Music P2P** Aimster AudioGalaxy Rhapsody Mac Satellite Blubster DirectConnect EDonkey Emule Overnet FileRogue Furthurnet Gnutella Acquisition Ares BearShare Furi

RTCP SIP Skinny (SCCP) T.120 VDOPhone	Backweb Chaincast EntryPoint Kontiki Marimba PointCast NewsStand WebShots **ERP** Baan JavaClient JD Edwards Oracle (7,8,9i) SAP **Internet** ActiveX FTP, Passive FTP Gopher HTTP Tunnel IP, IPIP, UDP, TCP IPv6 IRC Mime type NNTP Socks2http SSHTCP SSL TFTP UUCP URL Web browser **Database** FileMaker Pro MS SQL Oracle 7/8i Progress	**E-mail and Collaboration** Biff cc:MAIL IMAP LotusNotes MSSQ Microsoft DCOM (MS Exchange) Novell GroupWise POP3 SMTP **File Server** AFS CVSup Lockd Microsoft-ds NetBIOS-IP NFS Novell NetWare5 rsync **Games** Asheron's Call Battle.net Diablo II Doom EverQuest Kali Half-Life LucasArts (Jedi*) MSN Zone Dark Age of Camelot Quake I, II, & III SonyOnline Tribes I,II Unreal Warcraft III Yahoo! Games	DECnet IPX FNA LAT NetBEUI MOP-DL/RC PPPoE SNA **Messaging** AOL Instant Messenger ICQ IRC MSN Messenger Yahoo! Messenger **Misc** AOL **MultiMedia** Multi-cast NetShow NetMeeting QuickTime RTP Real Audio Streamworks RTSP MPEG ST2 SHOUTcast WebEx WindowsMedia **Thin Client or Server Based** Citrix Published Apps, Nfuse, IMA RDP/Terminal Server	Gnotella Gnucleus gtk-gnutella LimeWire MyNapster Mactella Morpheus Mutella Nap Share Phex Qtraxmax Qtella Shareaza toadnode XoloX Groove Hotline iMesh KaZaA KaZaA Lite Napster Amster audioGnome File Navigator Gnapster Grokster gtk napster jnapster MacStar Maxter My Napster Napigator NapMX Napster Fast Search Napster/2 Napster, MacOSX OpenNap Rapster Snap Spotlight WebNap WinMX Scour Tripnosis

Table 7: Network Protocol

2.3.2 Network QOS

There are many kinds of QOS techniques to handle the packet priority and certainly it depends on the network infrastructure and nodes themselves. Usually, the QOS can be tuned in different layers (ISO) L2 is for VLAN priority, L3 is for IP precedence and TOS/RSVP, and upper layer; RTP and MPLS are the examples.

- **Data link Layer:**

Looking at the Ethernet frame, in IEEE802.1p/Q Trunking, priority bits could be tuned up for small size VOICE packet in case of having been waited for steaming media or long transfer packet.

(Cisco Link Fragmentation and Interleaving (LFI) could be used to allow a small packet to break into the large packet)

Tagged Frame Type ‘8100’ 2B	802.1p Priority 3B	Canonical 1b	802.1Q VLAN IP 12b

Figure 8: Layer 2 QOS VLAN (B=bytes, b=bits)

- **Network Layer :**
  - RSVP: Usually the Routers along the path have to enable this function so it may be impossible to use this feature to improve QOS.
  - TOS/IP Precedence/DiffServ: (IP Header:TOS Byte DS Field (8b))

Apart from IP precedence and TOS, DiffServ is the latest implementation to provide QOS using the TOS byte. It uses the first 6 bits of the TOS know as differentiated services code point (DSCP) which represents 2^5 or 64 different classes of services.

TOS Byte : (IP precedence: 3 bits, TOS: 4 bits, CU: 1 bit)

P2	P1	P0	T3	T2	T1	T0	CU

DiffServ representations (DSCP: 6 bits, ECN: 2 bits)

DS5	DS4	DS3	DS2	DS1	DS0	ECN	ECN

Precedence (3b)	Type of Service (4b)
000 Routine 001 Priority 010 Immediate 011 Flash 100 Flash Override 101 Critical 110 Internet 111 Network	1000 Minimize Delay 0100 Maximize Throughout 0010 Maximum Reliability 0001 Minimize Monetary Cost 0000 Normal Service

Table 8: IP Precedence and Type of Service (B=bytes, bit=bits)

DiffServ Bits 0,1, and 2 (Precedence)	DiffServ Bits 3,4, and 5 (Delay, Throughput, and Reliability)
111 = Network Control = Precedence 7 110 = Internetwork Control = Precedence 6 101 = CRITIC/ECP = Precedence 5 100 = Flash Override = Precedence 4 011 = Flash = Precedence 3 010 = Immediate = Precedence 2 001 = Priority = Precedence 1 000 = Routine = Precedence 0	Bit 3 = Delay [D] (0 = Normal; 1 = Low) Bit 4 = Throughput [T] (0 = Normal; 1 = High) Bit 5 = Reliability [R] (0 = Normal; 1 = High)

Table 9: DiffServ Representation

- **Application Layer :**
    - RTP Header Compression: (disable UDP Checksum) : In some network that the link capacity is quite rare, instead of sending 78 bytes (G.729), 40-43 bytes could be sent by compressing RTP/UDP/IP header so capacity requirement will be only (1/20ms)*40*8= 16kbps. However, this compression consumes more CPU and adds delay.

2.3.3 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.

```
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 9: CBWFQ Configuration Example (Cisco Network) [5]

2.3.4 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.

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

Figure 10: Traffic Shaping Configuration Example (Cisco Network) [5]

2.3.5 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.

```
interface pos 0/0
mtu 9000

interface pos 0/0
```

```
ip mtu 9000
```

Figure 11: MTU Configuration Example (Cisco Network) [5]

3. VOIP Management

3.1 Configuration Management

This management is all about the setup the VOIP component. Whenever making changes in this system, it has to be kept anytime. Knowing the Network Components (Switch, Router, VOIP Gateway, IP PBX, IP Telephone Server) and Network Inventory are needed to be done before making change. After setup the network to be run smooth and stable, maintaining these critical configuration files (Backup System) has to be needed to keep in other system to rollover whenever the systems crash.

Name/Model	Location	Address	Role
Catalyst 6509	Brooking Hall	10.1.1.100	Backbone
OS	**Memory/Disk**	**CPU**	**Installed modules**
12.01T	Flash 128M	1GHz	Gigabit*4/Fast*24

Table 10: Network Component Description

3.2 Event Management

To prevent VOIP downtime, event management is needed to keep eye on; for example, Windows log, Syslogs, Application-specific event, and logs. Figure 9 shows the event viewer program from Microsoft window to show all events that are processed at that time. Perhaps, the logging or event software will be bundled with the VOIP application so it is a must for system administrator to look after these events before making VOIP system crash.

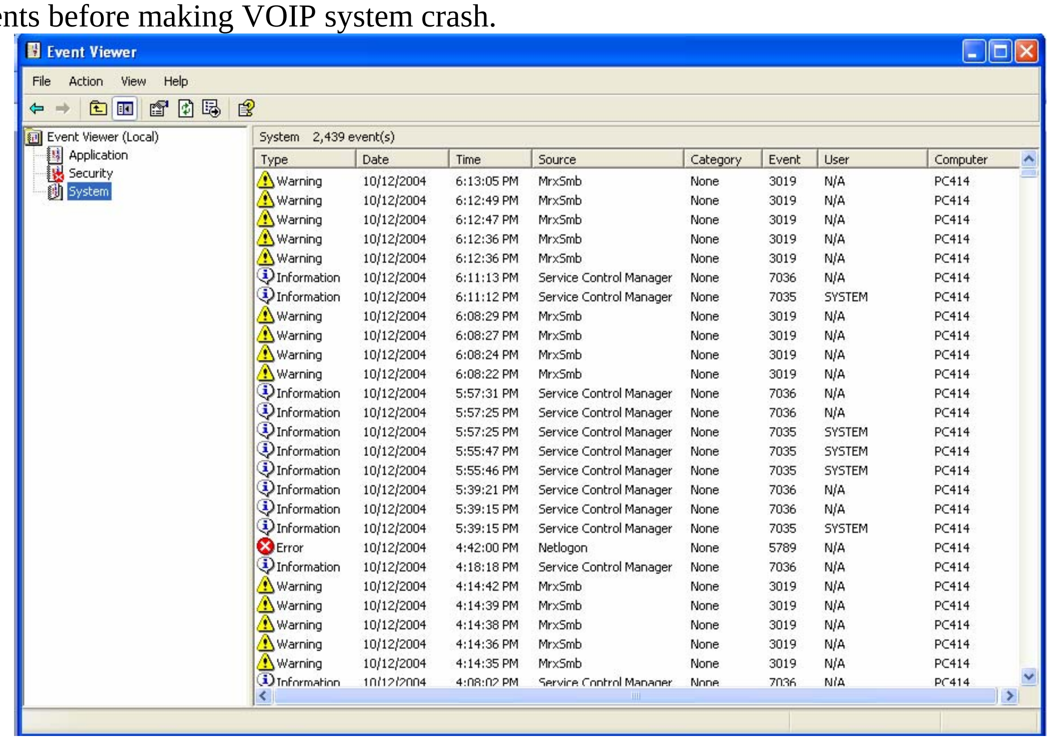


Figure 12: Event Viewer (windows)

3.3 Fault Management

Unfortunately, if VOIP system goes down, some actions have to be taken immediately. The longer the system downs, the more the company profit will be lost. “Log Files, Performance Counter, SNMP, OS API” are the important factors to look through.

4. VOIP Security

It may not need to implement high security policy for PSTN but it is a must whenever VOIP is deployed. A few minutes could be spent to capture IP packet with personal computer and LAN interface but it would take so long time to tap the telephone line with some specific equipment. There are three stages to manage VOIP security [1]

- **Prevention**

Since VOIP is based on IP network, standard IP security will be applied like Email and Web Server. Usually, Firewall and Antivirus Software have been used for this purpose and also the system administrator requires to patch and update software or even often fix the software hole.

In campus system, it may be difficult to apply this system efficiently because there are a lot of computers, so some centralized update will be used; for example, OS, “windows auto-update system with local server update” and Antivirus Software, “Norton or Mcafee with local server update”.

- **Detection**

Naturally, it is very difficult to prevent the breaking of security but it will more sense to know immediately whenever the security has been penetrated. There are two systems for intrusion and detection process.

- Host-Based intrusion detection system (HIDS): In this case, some agents have to be installed in each host computer. It could be updated individually or even centralize updated
- Network-Based intrusion detection system (NIDS): This agent will run on some server independently and it will monitor packets that flow in network. Some signature and characteristic will be matched and then usually, this system will relate to the Networking equipment to turn off or block the risky connection. This system can be run automatically with centralized management.

*Usually for campus network, Network-Based is implemented because it is very easy to maintain and management. Firewall and IDS is used at core network; however, since we know most commonly the invading traffic comes from the insiders, some techniques, VPN connection, may be used to implement in secure network; for example, account database and billing connection.

- **Reaction**

When we know who invades the system or even what is the invading traffic, some policies have to be applied; for example, blocking and discarding traffic. Also, after investigating the problems, some rules will be implemented to prevent the recurring events.

4.1 VOIP Security Recommendations

Three areas have to be concerned about VIP Security; the physical security, server security, and connection access security.

Physical Security

This is the fundamental step to secure any systems. Any equipment; Server, Gateway, IP PBX, Router, Switch should be kept in the security rooms with no personal contact. Usually some systems will have the backlog for maintenance purpose so anyone could be system administrator whenever he could access the Servers directly.

Not only Servers themselves but also the physical media has to be secured from both the invaders themselves and some accident events; someone broke the fiber optic because it lay across the way to go to restroom. The media should be laid not for anyone to easily tap the data and also fiber optic is the recommended media.

Server Security

Because Servers are made to work for several purposes such as E-mail, Web, and Ftp Servers could be run in the same computer, each application server has some different security characteristic so Hardening Server will be required for each application. (Some secure server require Host-Based intrusion system) Also, if the Servers are computer based, Booting Sequence and BIOS setup may have some restriction rules.

Connection Access Security (AAA)

Authentication, Authorization, and Accounting have to be implemented obviously. Who could access this system, What they can do, and What do they do after granting the access are all we need to know.

4.2 VOIP Security Guideline

- Separate the VOIP traffic both data and signaling traffic in different VLAN
- Replace HUP with Switch and plug this connection to IP phone or soft phone directly
- Apply the private IP instead of public IP (NAT may be used)
- For secure connection, VPN has to be implemented (IPSec)
- Deploy HIDS for VOIP server (hardening servers) and NIDS for core network with Firewall
- User Policy will be needed to apply in authentication, authorization, and accounting. (LDAP)

5. Conclusion

In this report, some VOIP characteristic has been mentioned in order for system administrator to go up one step to understand the VOIP idea before implementing. Whether we should change the telephone system to be VOIP depends on both pros and cons that impact to our organization. If it is the need to have high voice quality, PSTN is the perfect system for but if not, to reduce the telephone system cost with single data and voice management, VOIP may be good choice for that.

Moreover, whenever VOIP is chosen to implement, the good design and planning have to be concerned. Not only achieving the low cost with high quality but also making plan to upgrade whenever the company has grown up for years. Also, VOIP management and security are the important factor that system administrator need to look after. You may have nothing to do when network run smoothly but you will be getting trouble only if your CEO could not make a call to his daughter.

6. Reference

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