Voice over Internet Protocol (VoIP)
And, its applicability at
Rutgers, The State University of New Jersey

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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 Rutgers, The State University of New Jersey (the 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.

This document is intended to be brief, yet inclusive of key discussion points for consideration with VoIP deployment at the University. Reference is made to several technical documents that can be readily found on the Telecommunications Division (TD) website, http://www.td.rutgers.edu/documentation/ or from the bibliography.

Executive Summary

There has been much discussion about VoIP phone systems this last year within commercial and public environments. *"Although progressing rapidly, Internet telephony still has some problems with reliability and sound quality, due primarily to limitations both in Internet bandwidth and current compression technology"*.[1] 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. There are also large contingents that have opted to wait out their decision in hope of further maturation of the market, products and serviceability.

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.

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.

[1] See Voice over Internet Protocol, No. 2 Intranet Telephony Paves the Way for Internet Telephony

VoIP technology is certainly applicable within the University. Our environment is distributed and decentralized which subsequently fostered the growth of various systems. In truth, a centralized environment would be troublesome to financially justify its wholesale exchange to a VoIP platform, unless the installed telephone system / PBX was obsolete, dysfunctional or overly expensive to maintain and operate. Overall, an implementation plan becomes simpler when accomplished in smaller projects; therefore a distributed environment might prove to be more conducive to this approach. The University is however, striving to embark on unified voice architecture while leveraging existing assets accordingly. If done properly, we can capture the benefits of technology without undue financial burden. Some caveats to VoIP applicability at the University are as follows:

- There are a variety of telephone systems throughout the campus with differing degrees of newness. Many assets can and should be reused. A hybrid approach, (both existing assets and VoIP) is the most likely outcome.

- VoIP Call Managers, voice mail servers and gateways are localized yet integral to the network. Additionally, Adds, Moves and Changes involve several areas of responsibility. A shared model of system administration is probable due to these factors.

- VoIP systems require additional power to operate handsets and also depend upon the network and power utility for robustness. Concessions would have to be made regarding loss of power and the subsequent inability to make/receive phone calls, or the added expense of alternate power arrangements.

- The existing voice clearing account uses surcharges on lines and long distance calling to maintain operations and return a surplus to Central Administration. A new mechanism of funding the voice clearing account may have to be devised in order to recover maintenance, line and support costs associated with the telephone infrastructure.

- The addition of voice onto the Rutgers University Network (RUNet) will require additional resources in terms of technical support/expertise, hardware, software and monitoring tools. There will also be additional bandwidth consumption. Although this would appear to be minimal on the core of the network, Local Area Networks (LANs) within departments and buildings would be most affected. In some cases, VoIP may not be practical due to these local limitations.

- Relative to Disaster Recovery planning, all voice and data would traverse the same network infrastructure. Traditional phone systems use alternate facilities thereby allowing voice communication during network outages.

- A VoIP implementation would have shared components that would require services from the local Unit Computing Specialist (UCS), Network Operations Center (NOC) and the Voice Services group. A new repair process would need to be devised since trouble-shooting procedures would need to be clearly defined to avoid confusion across domains.

The Current Environment

The University has numerous types of telephone systems. This is primarily due to a decentralized approach for both funding and planning of voice services across departments and campuses. The following is a list of various telephone systems deployed across the New Brunswick area campuses in varying quantities.

Some are nothing more than a telephone instrument connected to a Centrex telephone line all the way up to a PBX and VoIP telephone systems utilizing Primary Rate Interface (PRI) Integrated Services Digital Network (ISDN) services and Direct Inward Dial (DID) numbering. The University has three primary telephone system vendors. They are Avaya (formerly AT&T and Lucent), Nortel Networks and Cisco. There are other vendors but they have a very limited presence i.e. 3Com, Executone, and Vodavi. The following is a breakdown of the types of equipment that we have on all our campuses:

Avaya:

- Single Line through 4 Line Telephone Sets
- 1A2 Key Sets (Six, 10, 20 & 30 button Sets)
- Merlin Classic Systems (We have some of every model ever produced)
- AT&T ISDN Sets
- Partner Systems (We have some of every model ever produced to include beta systems.)
- Merlin Legend (We have some of every model ever produced)
- Merlin Magix (We have some of every model ever produced)
- G3 Series PBX Systems
- Avaya Internet Protocol (IP) Office Systems
- Merlin Mail
- Partner Voice Mail
- Intuity Voice Mail
- Merlin Magix Mail
- IP Office Mail

Nortel Networks:

- Norstar (Various sizes)
- Business Communications Manager (Various Sizes to include VoIP implementations
- Nortel PBXs (Option 11, 61, 81)
- Nortel PBX Option 11 VoIP System
- Nortel Voice Mail (various versions)
- Call Pilot Voice Mail

Cisco:

- Cisco Call Manager VoIP System
- Unity Messaging System

Number of users

The University has a combination of Plain Old Telephone Service (POTS) lines on single line telephones and telephone systems sharing POTS telephone lines on numerous telephones. The number of telephone lines in use is approximately three quarters of the total number of telephone sets. The University has approximately 15,000 telephone lines with 20,000 plus telephone instruments.

How they are managed today

Today the University telephone systems are managed by individual departmental staff, Voice Services staff, Avaya technicians, Blackbox technicians (formerly NextiraOne), and Dimension Data technicians. Since the University uses a distributed model for telephone systems it is up the department to either support their systems with their own staff or pay for technical support from one of our outside vendors.

How call accounting is handled

All Centrex POTS lines are billed to the responsible department through the University Billing System. When a department shares multiple lines they account for the usage of the lines using the University Billing System. However, if they are trying to account down to the station it is up the department to purchase a call accounting system that they put on one their own personal computers to allocate costs by users or cost centers. It is a one-time set-up that is turned over to the departments to manage.

Typical cost of system

A basic telephone (single telephone number) will cost less than a hundred dollars. However, the cost of a telephone system with a Voice Mail platform can range anywhere from $600 to $1800 per station. The lower the cost the less functionality the system has both in features and potential size. The typical cost for the installation of a telephone line is $120 plus an ongoing cost of $17.20 per month. In the case of a system sharing telephone lines, the recurring cost is $17.20 per line but the ratio is less than 1 line for each person i.e.100 users may share 30 telephone lines.

Typical features

Because telephones have been deployed based on a distributed model, features vary from system to system. The following is list of typical features that are common across the University but they are not necessarily present in all systems:

- Call Transfer
- Conference Calling
- Call Forwarding
- Call Forwarding Busy

- Call Forward No Answer
- Automatic Callback
- Call Waiting
- Multiple Line Appearance
- Call Pick-up
- Speed Calling – Individual
- Speed Calling – Group
- Call Hold
- Speaker Phone
- Hands Free Answer Intercom
- Mute
- Voice Mail
- Auto-Attendant Services
- Broadcast Messaging

How repairs are handled today

Repairs are handled in a number of ways. For the majority of repairs, calls are made into the Voice Services Repair Desk where an assessment is made of the problem. If the problem cannot be solved at the Voice Services Repair Desk or with the assistance of a staff member, then a trouble ticket is opened with the appropriate vendor for repair. If the system is under warranty the work is subsequently covered. If it is not under warranty, the appropriate vendor is paid for time and material repairs from the repair/maintenance account. In some cases a departmental administrator can make repairs such as changing a name or resetting a password on their own. Additionally, some administrators can do extensive troubleshooting and can even make moves, adds and changes. If they are unable to make the repair, they contact the Voice Services Repair Desk.

Distributed environment

The University is currently utilizing a distributed model for telephone services. Each department or school within the University is given money from the Central Administration for telephone services. In some cases the department or school uses 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 has resulted in over 350 different types of telephone systems distributed across the University. This model is becoming almost impossible to manage given the many different types of systems, voice mail, system software, etc.

Operational Considerations

In addition to technical considerations when deploying a system, there are general and operational aspects that must be considered when deploying a system. For example, will the deployment be localized to a department and building or will it be significant in scope such as a campus wide implementation? This choice will dictate the options available for overall administration; call accounting, local expertise requirements, etc. In general, some base assumptions must be made with respect to the University. One of these is that regardless of the type of installation, there will be a requirement to connect to the core of the University infrastructure and exist within the current support model. Admittedly, some processes must change to reflect the addition of a different technology and its' unique or special requirements, however as long as they are pertinent to an overall architecture and applicable University wide, this is an acceptable criteria. Some important criteria and background are as follows:

A VoIP system is largely considered an application running over a network. For comparison purposes a traditional voice system contains some type of PBX, PABX (Private Automatic Branch eXchange), key system or similar, and conventional transmission facilities. Facilities can be analog or digital and either trunks, tie lines or access lines. For the most part, the PBX is a switch that simply establishes and breaks down calls, and routes calls based upon a numbering convention. If the system is on premises and customer owned, links to the Public Switched Telephone Network (PSTN) enable calls to be established with the public. If a service such as Centrex is employed, the telephone company (Verizon) maintains all of the equipment and simply supplies lines and extensions or some localized equipment. Call accounting is extracted from a centralized system linked to one or many PBXs. With a service model, most aspects of administration are handled directly by the phone company with a few relegated to the customer.

With VoIP systems, the transmission facility is the data network. A Call Manager/Server and Distributor replace the PBX, and links are still required to the PSTN for enabling external reach-ability. Similar to multiple PBXs, there would be several Call Managers and Distributors to service multiple geographic areas as practical. Additionally, there would be routers and gateways as well as handsets that are able to communicate over the network back to the Call Managers. Call accounting would be accomplished similarly in a centralized fashion. Voice mail functions would be handled separately through a Voice Mail server.

For clarification, a "Call Manager" is Cisco's specific implementation of a Call Server (CS). The CS provides basic facilities to enable a network of Internet Protocol (IP) devices to emulate the functions of a PBX: station-to-station calling, digit dialing, least-cost call routing, PSTN interface, call transfer, call forwarding, etc. CS may optionally provide additional services such as voice mail, conferencing, PSTN gateway, XML (eXtensible Markup Language) services, auto-attendant, etc. There are several open-source, GPL (General Public License) implementations, including Asterix and Bayonne that can be explored and tested for viability. Of course, there are also commercial products readily available as well.

Up to now at the University, there haven't been any architectural recommendations for designing, implementing or operating a unified phone system except for the "Voice Infrastructure Development Proposal" put forth by TD. The University has largely depended upon Verizon for support through traditional Centrex type services. At present however, it is feasible to architect a University wide phone system as long as the approach is consistent with technically sound design and implementation guidelines and a reasonable support model is defined. Some considerations and noteworthy items are as follows:

System management

Overall, there has been a consistent lack of architectural recommendations (undirected, incremental growth) that lead to an archipelago, which will need to be integrated for successful operation and management. This is consistent with our decentralized operation. Central management of IP and PBX boundaries will be necessary for a successful model.

The Voice Services group will need to use centralized tools to monitor IP-capable systems in addition to traditional PBXs. Where possible, the NOC can be used to monitor IP-capable, centrally managed IP-PBX and IP voice systems. *"Network Management Systems are the heart of a network operations center (NOC). NOCs are often typified by the graphical network displays projected on large screens on their walls"*[2] Monitoring however, can partially be integrated into some existing tools such as HPOV OVIS (Hewlitt Packard Open View and Open View Internet Services) for call managers, distributors and gateways. *"In addition, dozens of customer relationship management applications, billing packages, service-creation tools, and other applications now exist that can be acquired and integrated into a full solution."*[3]

It will be necessary to cross-train staff and integrate design and management processes. This is largely due to cross-domain integration and convergence.

Call accounting

Both commercial and open source tools exist to produce Call Detail Records (CDRs) equal to or exceeding the quality of existing CDRs from Centrex. This will require some research and testing before settling on a desired toolset. It is envisioned that call accounting would fall into a shared domain similar to call manager administration.
TD will need to further explore capabilities and applicability of soft-PBX and call server products to ascertain the appropriate choice for the University.

Scalability

Scalability is determined by the collective components of the network. VoIP systems include a DHCP (Dynamic Host Configuration Protocol) server; call server/DNS (Domain Name System), access/media gateway, trunk gateway, the IP phone or soft phone, and the IP LAN/WAN (Wide Area Network). Scalability is determined by analyzing current network and application usage/capacity, anticipated/proposed network upgrades, and size of VoIP service offering.

It will be crucial to benchmark various levels of activity on a smaller scale or perhaps in a lab environment so that the impact of VoIP along with other applications can be measured accordingly.

Further research and prototyping will be required to ascertain the boundaries of operability. This largely depends upon the scope of design and planned deployment. It is premature to discuss the scalability factor of a unified architecture, especially since a University deployment will contain distributed elements.

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[2] See Operations Support Systems, No. 1 Network and Trouble Management
[3] See Internet Model for Control of Converged Networks, No. 1 Introduction

IP addressing requirements

Current estimates fall between 50,000-70,000 of unique stations that will require addresses in a full-scale implementation. Additionally, call servers; voicemail servers, and PSTN gateways will require addresses. An address space like 10.0.0.0/8 will need to be held in anticipation of a large-scale IP voice/video implementation. Network Architecture within TD will devise an addressing plan based on anticipated IP Voice/Video penetration across the University and allocate super-net address blocks to geographic regions to aid route summarization and management.

Required skill sets

Call servers, voice mail servers and some gateway software is based on commodity hardware, hence requires fundamental Windows and/or UNIX system administration skills. It is better to run applications on professionally managed servers for standardization and access to a common knowledge base. Service applications (actual call server software, voice mail server software, etc) are generally user-friendly, but will require large-scale vision to integrate into successively larger models. A higher skill set will be required for successful integration.

Traditional telephony skills are required to support voice application architectural needs such as numbering plans, codec selection, PSTN support, signaling, etc. Also, traditional data communication skills are required to support packet network architectural needs such as IP routing, L2 switching, QoS (Quality of Service), PSTN gateways, WAN support, etc. A combination of both skills would be ideal to bridge the gap between the traditional voice and data paradigm. Cross training will be necessary to develop specialized skills for monitoring and troubleshooting unique packet voice problems.

Power over Ethernet requirements

Power over Ethernet (PoE) is not a formal requirement for providing VoIP services. PoE is used in combination with uninterruptible power in the IDF (Intermediate Distribution Frame), and is intended to ensure voice service during power loss episodes in the facility. For this to be effective, all IP-PBX elements must be proofed against power loss, including PoE switches, call servers, PSTN gateways and any supporting telecommunications infrastructure. PoE devices in a closet require additional HVAC (Heating Venting Air Conditioning) facilities since these devices use power rectifiers to convert AC (Alternating Current) to DC (Direct Current) power, and generate more heat than non-PoE data switching equipment. Closets may require two to four times the present tonnage of cooling. Additionally, PoE devices have greater power requirements as they draw more wattage (KVA), than non-PoE devices, depending on the overall number of available PoE ports. For this reason, closets may require up to eight times more power than current power requirements. PoE devices are most often used to provide power to down-line devices during power outages. Given their stringent power and HVAC requirements, room-size UPSs (Uninterruptible Power Supply) may be required, perhaps exceeding 10k VA (Volt Amps), in each PoE-equipped closet.

Also, it should be noted that there are two standards for PoE at present, 802.3af or 802.3at (PoE Plus). The 802.3at standard has not been ratified, however IEEE (Institute of Electrical and Electronics Engineers) is projecting completion sometime in 2007. 802.3af allows some devices to be powered through standard RJ-45 category 5 cabling with maximum power deliverable to 15.4W. 802.3at is targeting to deliver at least 30W to devices. A cabling standard is under development and tests are being conducted to determine what degradation the proposed increase in power would have on various cable types (cat3/5/5e/6).

Network assessment

A complete network assessment will be required before implementing a VoIP system. This is necessary to achieve a baseline before the introduction of a new application onto the network. Some of the tasks that should be performed are as follows:

- Measure network utilization – Core, Edge and LANs where appropriate
- Perform tests to determine delay (delay variance), jitter, and packet bursts/loss
- Simulate signaling and call traffic to calculate available bandwidth on various network segments.

Monitoring tools whether centralized or distributed will need to be introduced to verify service integrity.

Numbering scheme

Currently the New Brunswick area campuses for administrative purposes are utilizing two telephone exchanges, 732-932 –xxxx (for the campuses on the south side of the Raritan River) and 732-445-xxxx (for the campuses on the north side of the Raritan River). Almost all numbers in both exchanges are exhausted. With the implementation of either PBX or VoIP solutions in New Brunswick, we would have to move away from a shared number model and have dedicated numbers for each telephone instrument. This would require additional exchanges and a new numbering plan. A potential scheme could be as follows:

732-932-xxxx	- College Ave Campus
732-New-xxxx	- Cook/Douglass Campus
732-445-xxxx	- Busch Campus
732-New-xxxx	- Livingston Campus

Five-digit dialing could be maintained for on-campus dialing with the Cook/Douglass and Livingston Campuses just having a new lead number in the dialing plan.

E911 for VoIP

Enhanced 911 or E911 service is a North American Telephone network feature of the 911 emergency-calling system that automatically associates a physical address with the calling party's telephone number. "*In order for emergency services to be dispatched to a 9-1-1 caller, the caller must be located, the call must be routed to the nearest of over 6,500 Public Safety Answering Points (PSAPs), and the callers location must be automatically* *delivered to that PSAP."*[4] This is generally done by a form of reverse telephone directory that is supplied by the telephone company as a computerized file. Computer software makes an association between the caller's line and an address. This provides emergency responders with the location of the emergency without the caller providing it. This is especially useful in times of fires, break-ins, or other events where communication is difficult or impossible.

While VoIP systems do have the E911 software available for purchase with them, in our Centrex telephone services environment this software must be fully integrated with the telephone company's databases and University Police's E911 PSAP to make them functional.

The nature of IP makes it difficult to geographically locate network users without full software integration. Emergency calls today are routed to the public network through ISDN PRIs and the associated Gateways within our VoIP systems. In some cases a 911 call will only display the number of the ISDN PRI circuit and nothing more. If the caller is not in the building where the VoIP system and PRIs are physically located then he or she would have to provide the E911 Center with an address and location of where they are located. However, if the caller is unable to give an address, emergency services may be unable to locate them in any other way. When the system administrators use DID numbering for each telephone it can be registered with the telephone company databases, but VoIP systems allow users to pick-up and move their telephones from building to building. In this case the VoIP System E911 Software would need to allow for this and subsequently update the appropriate telephone company database. This will be crucial as changes occur otherwise the caller and University could be at risk in emergencies that the E911 PSAP at the Police Station won't be able to locate.

Disaster Recovery

A disaster recovery plan for the University community is a huge undertaking spanning many departments in addition to requiring significant resources from OIT (Office of Information Technology). A starting point when developing a plan is to define high-risk areas or those that are susceptible to utility outages first. Then of course, vulnerabilities in equipment, facilities, services, or site specific characteristics or anomalies should be noted. Also, requirements for essential services need to be clearly defined so that the recovery plan can be of proper scope and quick to enact.

As stated earlier, VoIP systems use the network infrastructure, require additional power and have several more components than traditional phone systems. Decisions about the level of service will dictate the resource requirement from a facilities and equipment perspective.

In brief:

- Will phone service be considered primary or secondary?

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[4] See Is VoIP Without E9-1-1 Worth the Risk? : Challenges, Approaches, and Recommendations for VoIP Service Providers, No. 2 The Current 9-1-1 Landscape

- What is the potential impact of a network outage on phone service?
- What means of communication are available during a disaster scenario?
- What level of capital is required to maintain service during power failures?

All of the above questions are real considerations relative to disaster recovery planning and specific to VoIP system deployment.

Service Level Agreements

RUNet operates on best effort presently. Although it would be easiest to remain at best effort going forward, it would seem plausible that having all University services (voice, data) on the network would eventually necessitate a basic Service Level Agreement (SLA). The foundation and integral components of SLAs are as follows:

- *"correlating multiple events from one or more managed resources across one or more management domains*
- *running diagnostic tests*
- *calling on the knowledge experts*
- *isolating the problem and its root cause(s)*
- *eliminating symptomatic errors*
- *determining the impact on users and services*
- *notifying appropriate management and customers*
- *applying corrective actions to resolve each problem*
- *taking the failed component(s) out of productions for repair*
- *submitting trouble tickets"*[5]

Additionally, there are several items that need to be clarified prior to suggesting one. Some of these are:

- Power Availability- UPS with auto restart, generator backup, and UPS monitoring
- Network Reliability- Design, Redundancy, Hardware support and response time
- Software Availability- Software stability
- Diagnostic and Repair Process
- Demarcation of responsibilities within a shared environment (UCS, TD, Other)
- Handset and other equipment standardization

Industry Best Practices generally use two years of data to calculate an effective SLA. The above also assumes that proper monitoring and test equipment have been incorporated to the NOC.

"Ever Greening"

Ever greening or the replenishment/refresh of equipment is crucial to maintaining an adequate service level and useful state. In general, all hardware eventually becomes outdated, maintenance costs increase, newer features require updated hardware and software, etc. A regular plan is necessary so that obsolescence can be avoided. This is

[5] See Service-Level Management, No. 2 Background

especially true on RUNet since the infrastructure is significant and there are more than 2,000 devices on the network. Presently, there is no formal “ever greening” plan or capital budget for RUNet. This will eventually become problematic due to the quantity of devices that will coincidentally become unsupportable. Also it is much easier to methodically replace/refresh equipment in smaller, defined quantities. As we move toward convergence, the need for “ever greening” is especially crucial.

Interconnectivity of Traditional and VoIP Systems
There are several approaches to integration of traditional and VoIP telephone systems. There are also inherent advantages and disadvantages as well as tradeoffs that need to be made when considering a unified architecture.

Several brands of IP-based call servers support FXS/FXO (Foreign eXchange Subscriber and Foreign eXchange Office) hardware to allow direct interface with traditional, non-IP PBX systems. Some traditional PBX systems support IP modules to allow remote management, signaling and IP digital trunking. Where IP support is not present, it may still be possible to interface with IP systems if the IP-PBX supports traditional FXS/FXO facilities. For high level voice architecture and high level RUNet – VoIP migration, see Diagrams 1 and 2.

Diagram 1. High Level Voice Architecture

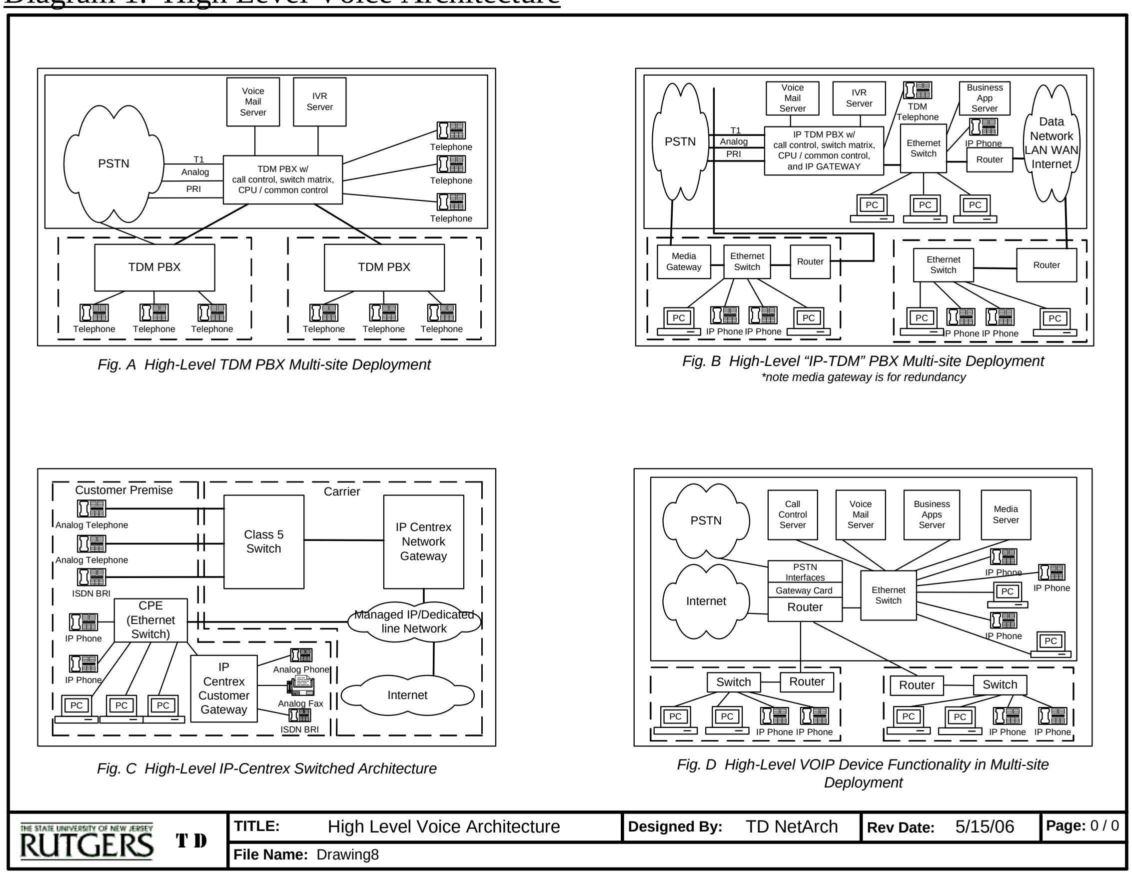


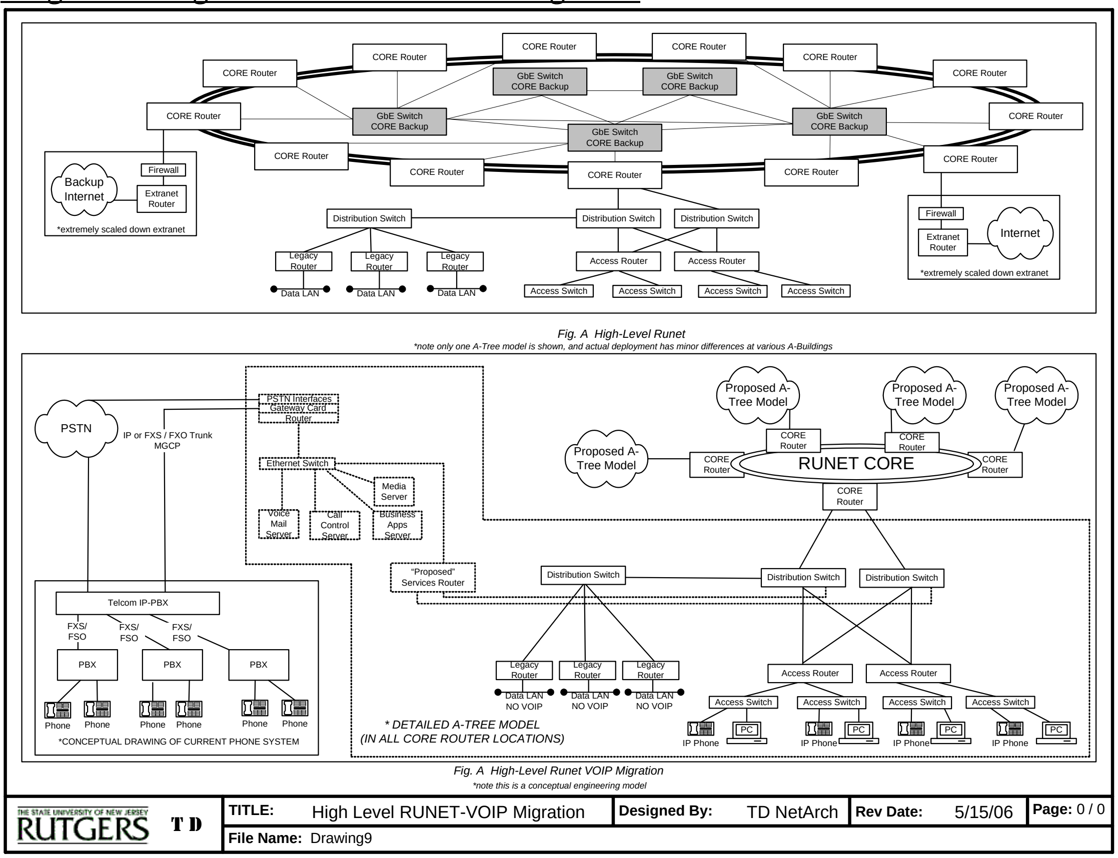
Fig. A High-Level TDM PBX Multi-site Deployment

Fig. B High-Level “IP-TDM” PBX Multi-site Deployment
**note media gateway is for redundancy*

Fig. C High-Level IP-Centrex Switched Architecture

Fig. D High-Level VOIP Device Functionality in Multi-site Deployment

Diagram 2. High Level RUNet – VoIP Migration



Some possible advantages of interconnection between traditional and VoIP systems are:

- We would be able to integrate existing assets of voice infrastructure into any new central IP voice implementation
- Potential for least-cost routing within the centrally managed IP voice plant
- Potential for a central or distributed PSTN gateway to reduce line costs
- Opportunity to define a migration path from non-IP to IP

Some disadvantages are:

- Hybrid system will have unique support challenges
- Increased complexity with few clear benefits to end-users
- May be difficult to support centrally and monitor accordingly

For comparisons and engineering considerations between VoIP protocols H.323 and SIP (Session Initiation Protocol), see Diagram 3.

Diagram 3. VoIP/ Network Protocol Overview

Factor	H.323	SIP
Design	Complex	Simplex
Number of Elements	100's	37
Messages	Based on ASN.1	HTTP and RTSP
Call Setup	Multiple Requests	Single Request
Extensibility	Non standard	Uses Session Description Protocol
Large-Number Domains	Designed for LAN	Designed for IP Network
Server Processing	"Hold" state for all calls	Pass through
Conferencing	Limited	Open to all sizes
Feedback	H.245 doesn't work in multicast	RTCP
Firewall Support	Difficult	Easier
Interoperability	Not Common	More Common

Fig. A Engineering considerations between VOIP protocols H.323 and SIP

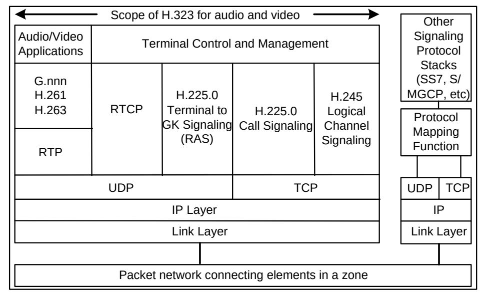


Fig. B VOIP Protocol H.323 Protocol Stack

OSI Model	Brief Description	Examples
7 Application	provides a means for the user to access information on the network through an application	HTTP, SMTP, NTP
6 Presentation	relieves the app. layer of concern in syntactical differences in a message's data representation within the end-user systems	ASCII, MPEG, MIDI
5 Session	provides the mechanism for managing the dialogue between end-user application processes	NetBIOS, SAP
4 Transport	Provides reliable and cost-effective transparent transfer of data between end users	TCP, UDP, RTP
3 Network	performs network routing, flow control, segmentation/ desegmentation, and error control functions	IP, ARP, RIP, BGP
2 Data Link	transfers data between network entities and to detect and possibly correct errors that may occur in the Physical layer	Ethernet, ATM, Frame Relay
1 Physical	defines all the electrical and physical specifications for devices	100BASE-T, POTS, T1

Fig. C OSI Model used in data networking

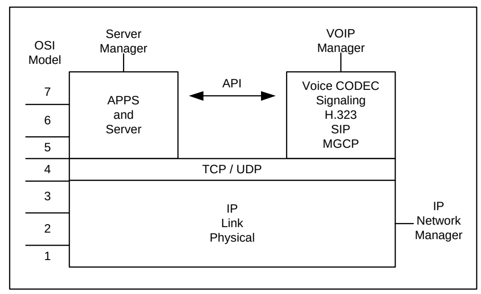


Fig. D Common VOIP protocols and where they reside in the OSI model

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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.

Glossary

AC	Alternating Current
CDR	Call Detail Record
CS	Call Server
DC	Direct Current
DHCP	Dynamic Host Configuration Protocol
DID	Direct Inward Dial
DNS	Domain Name System
FXO	Foreign eXchange Office
FXS	Foreign eXchange Subscriber
GPL	General Public License
HPOV	Hewlitt Packard Open View
HVAC	Heating Venting Air Conditioning
IDF	Intermediate Distribution Frame
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
ISDN	Integrated Services Digital Network
KVA	Kilo Voltage Amp
LAN	Local Area Network
NOC	Network Operations Center
OIT	Office of Information Technology
OVIS	Open View Internet Services
PABX	Private Automatic Branch eXchange
PBX	Private Branch eXchange
PoE	Power over Ethernet
POTS	Plain Old Telephone Service
PRI	Primary Rate Interface
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RUNet	Rutgers University Network
SIP	Session Initiation Protocol
SLA	Service Level Agreement
TD	Telecommunications Division
UCS	Unit Computing Specialist
UNIX	A type of multi-user/multitasking operating system
UPS	Uninterruptible Power Supply
VA	Voltage Amp
VoIP	Voice over Internet Protocol
WAN	Wide Area Network
XML	eXtensible Markup Language

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