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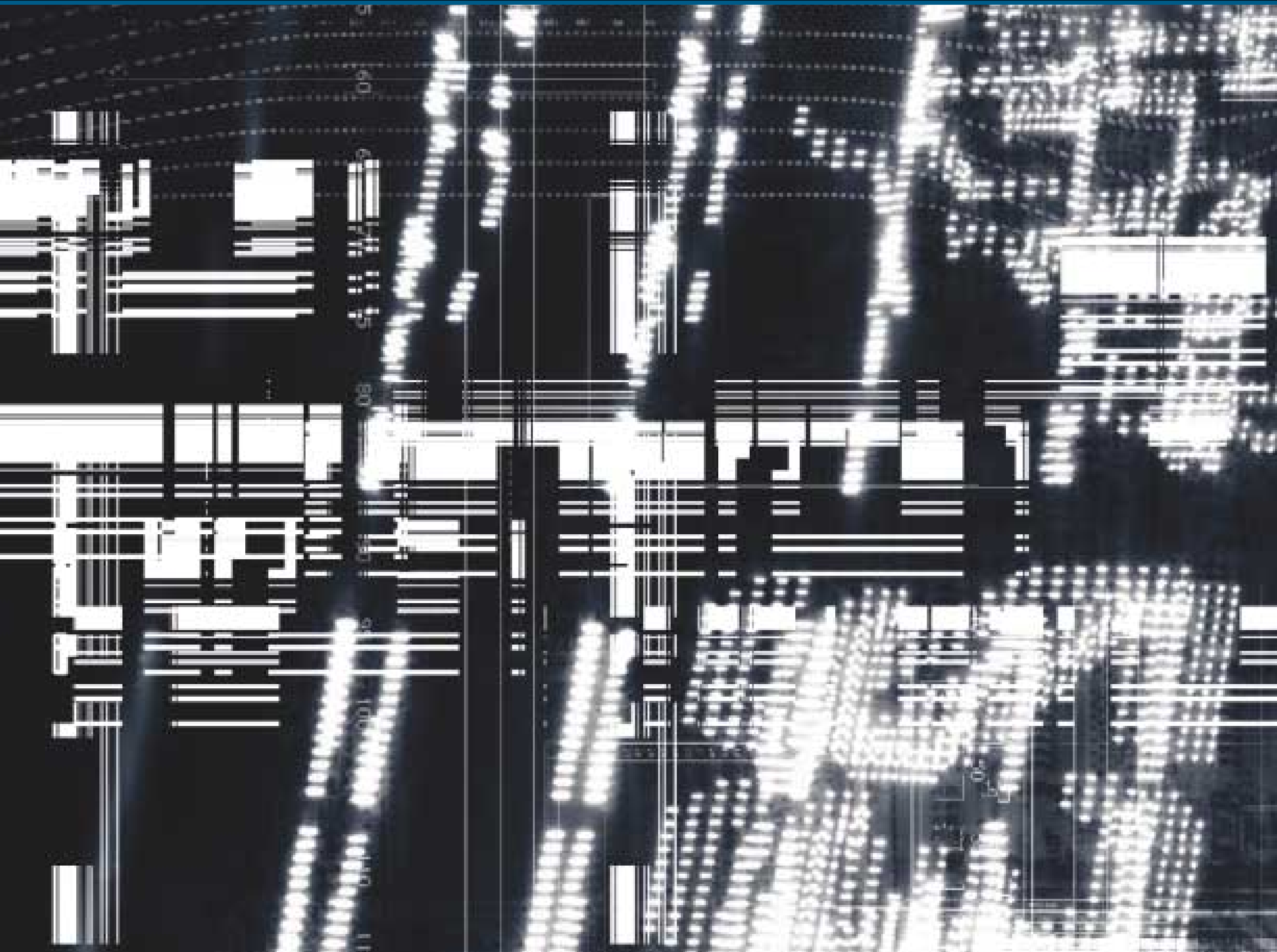
Representing Chief Information Officers of the States

May 2005

VoIP and IP TELEPHONY:
Planning for Convergence in State Government



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acknowledgements

NASCIO would like to express its utmost gratitude to the Infrastructure Sub-committee's Chair, John P. Gillispie, COO, Information Technology Enterprise Office, State of Iowa, and to Tom Shepherd, Information Technology Enterprise Office, State of Iowa, for their tireless dedication that made this publication possible. NASCIO would also like to thank the following members of the Infrastructure Sub-committee for lending their time and expertise to this publication:

- Dave Blackwell, Office of Information Technology, State of New Jersey
- Steve Dawson, CTO, State of New Jersey
- Marty Dunning, SUN Microsystems
- Gary Falis, Microsoft
- Christian Fuellgraf, IBM
- Michele Grisham, CISCO Systems, Inc.
- Stan Herrera, CTO, State of Alaska
- Lindsay Hiebert, CISCO Systems, Inc.
- Larry Johnson, CIO, State of South Carolina
- Denny Nincehelser, Telecommunications Division Director, State of South Dakota
- Gail Ulan, CIO, Office of the Governor, State of Arizona
- Victoria Wallis, Engineering Manager, Iowa Communications Network
- Kathy Williams, Iowa Communications Network

Finally, NASCIO would like to thank, Drew Leatherby, NASCIO Issues Coordinator, for his work on this project, and Doug Robinson, NASCIO Executive Director, Jack Gallt, NASCIO Assistant Director, Nancy Howard, AMR Marketing and Creative Services Director, and Chris Walls, AMR Senior Publications and Website Coordinator, for their guidance, editorial revisions and other assistance regarding this publication.

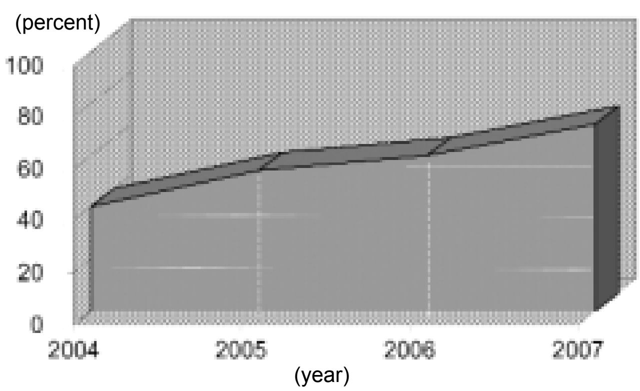
Please direct any questions or comments about *VoIP and IP Telephony: Planning for Convergence in State Government* to Drew Leatherby at dleatherby @amrinc.net or (859) 514-9178.

executive summary

Changing Telecom Business Models

The keen interest in Voice over Internet Protocol (VoIP) and Internet Protocol (IP) Telephony and its' accelerated adoption rate has led many to the realization that the long-established Public Switched Telephone Network (PSTN) voice business model will be facing its end of life in the next 10 to 15 years. Currently, VoIP business models are generally not well understood, are highly dynamic, and are affected by the organizations and vendors involved and dynamic regulatory issues. Standardization within the product lines and the promise of enhanced applications and reduced operational costs have put VoIP and IP Telephony on the fast track in many organizations. (See Figure 1.)

Figure 1. *VoIP/IP Telephony as a Percentage of New Phone Lines Installed*



Source: Gartner

This publication is intended to provide an understanding of the fundamental issues facing organizations as they assess VoIP and IP Telephony. It contains a description of the operation and functionality of the legacy Public Switched Telephone Network (PSTN), an explanation of voice transport over the Internet and IP Telephony, and a discussion of the issues and drivers associated with the migration from the PSTN to converged voice and data networks.

Using VoIP for Government Transformation

Today's economic and social climate is causing government to rethink how it operates. Efficient use of shrinking budget resources, optimizing revenue collection, or deploying services to meet the demands of a connected and "on demand" constituency, are some of the requirements placed on government agencies. When increased security requirements are added to this agenda, it is easy to see why government institutions must use transformational approaches and new technologies that enable transformation to change the way they operate to meet these new and ever increasing requirements.

As a result, government chief information officers (CIOs) are being asked to do more with less: increase quality while cutting costs, launch new projects with a smaller staff, standardize IT systems but reduce capital. At the same time, they have to ensure that communication and information flows smoothly between agencies and protects all mission critical applications from potential security threats.

Given these new demands on government services, and rapid changes in society and technology, business as usual is no longer an option. Working harder or longer has ceased to deliver the expected benefits and the use of isolated technologies that deliver point solutions are increasingly costly to maintain, integrate, and manage. Instead, a different operational approach and model is required. A transformation model is needed that takes advantage of organizational best practices and embraces the efficiencies and increased productivity enabled by using the right technologies; an approach that provides flexibility for future applications to be smoothly integrated, extended, and supported with common practices and infrastructure. VoIP and converged network technologies, along with proper planning processes, can make significant contributions to ensure transformation delivers effective results.

Transformation of government institutions means better leveraging of resources—both human and technology. One example is setting up inter-agency collaboration or optimizing citizen-services and shared government services. Transformation includes not only changing the business of government, but also the governance process itself. This represents a major change in the way organizations conduct their day-to-day operations, as well as how they think about themselves (e.g. taking a customer-centric approach) and promises dramatic benefits for those who reach their destination.

Transformation is a major undertaking. However, many organizations undertake the transformation journey without a clear roadmap and the experience of proven practices. As a result, these efforts never reach their ultimate destination. So while VoIP Technology and convergence technologies may enable transformation to occur, they do not necessarily guarantee that transformation will take place, because transformation is linked to the way people, processes, and the organization itself serves its constituents. For these changes to take place, initiatives of a VoIP project must be linked to, and driven by, the transformational initiatives as the key drivers. The key to the success of a longer term transformational initiative is that a proper framework is established in a cross-functional leadership team, that effective planning and processes are put in place to guide the technology (all technology) rollouts, and that partners and partnerships must be established and maintained for those who will help guide and support transformational processes.

Considerations for Transformation

Although government decision makers may face tremendous budget pressures in trying to enact their transformation efforts, they need to keep their focus on the ultimate goal of transformation—e.g. productivity increases, constituency satisfaction, and revenue generating opportunities—and not be content to accept cost reduction as the only objective.

Today, business decisions and processes are driving technology implementations. Equally important, technology serves to enable organizations to more efficiently address business needs. Therefore, it is critical that transformational VoIP projects link technology decisions to government business objectives and take into account the future goals of the organization.

Before making fundamental changes to the existing processes, procedures, applications, and infrastructure, CIOs should clearly understand and be able to articulate the value that they expect from business transformation. What are the interim and terminal objectives? What benefits are sought? What forces work in favor and in opposition to the proposed transformation? How will changes in one area affect processes, applications, and personnel in other areas? How can we minimize disruptions in services delivery and employee productivity?

Most importantly, who (what partners and vendors) can bring the exact set of complimentary skills, practices, and technologies that will lead organizations to their desired destination and ensure that the goals are achieved, not just marginal short-term benefits?

legacy telephony technology

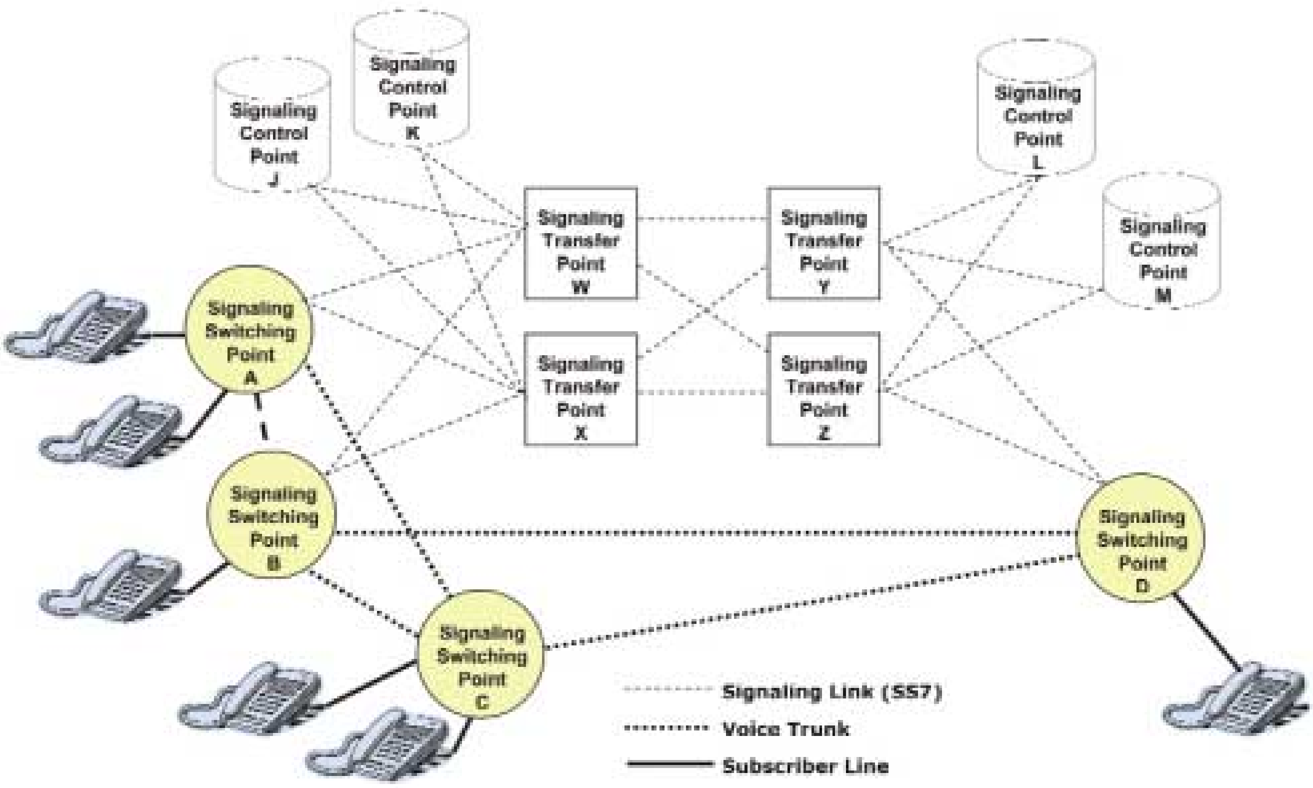
Public Switched Telephone Network (PSTN)

In order to understand the issues in replacing legacy PSTN systems with VoIP, it is important to understand the operational characteristics, and limitations, of each. (See Figure 2.)

The PSTN can be characterized by thinking of it as dumb devices driven by an intelligent network. In the PSTN, signals are sent between telephony switches to set up and terminate calls and indicate the status of terminals involved in the calls. These signals are carried over a separate data network known as Common Channel Signaling (CCS—represented by the black lines in the diagram below). The protocol used by CCS is Signaling System 7 (SS7). The entire system is called the Intelligent Network (IN).

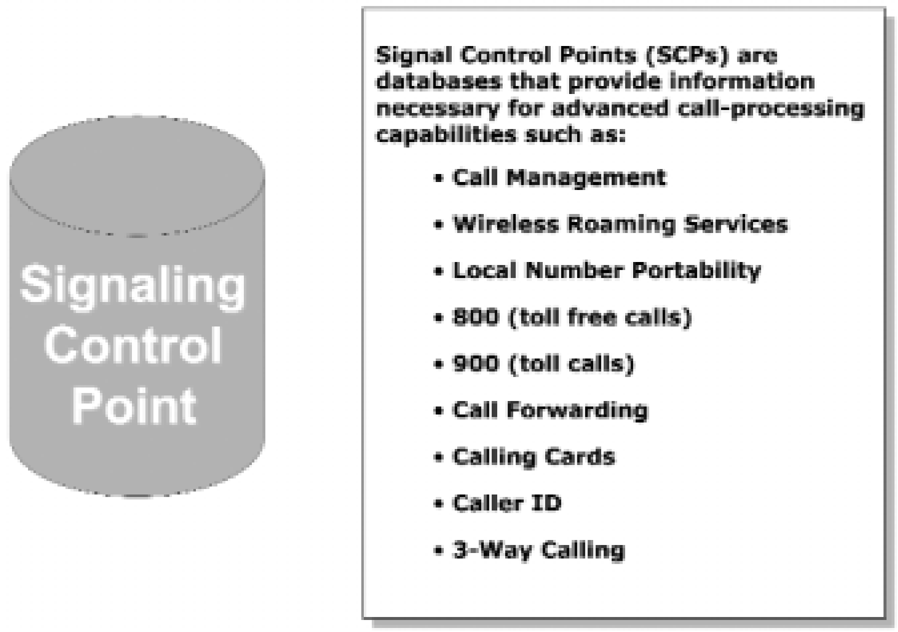
Signaling Control Points (SCPs) are databases that provide information necessary for advanced call-processing capabilities. For example, an SCP would be queried to determine the routing number associated with a dialed toll free 800/888 number or to validate the personal identification number (PIN) of a calling card user. (See Figure 3.)

Figure 2. *Public Switched Telephone Network*



Source: Cisco Systems

Figure 3. *Signaling Control Point (SCP) Functionality*



Source: Cisco Systems

Within the next few years, the existing PSTN will slowly be replaced by public packet networks. During this transition, the PSTN will continue to be heavily used because of the millions of users and non-IP devices still connected to it. In addition, the PSTN supports a variety of voice services through the signaling control points as illustrated above. An Internet telephony device needing to connect with one of these services must use PSTN signaling for the foreseeable future.

Centrex

Centrex is a service provider-based telephony solution that provides business class telephony features to enterprises. Centrex is built on a carrier's central office switch. The customer can have either analog phones, digital phones or both on their premises. The phones are then connected via individual copper pairs back to the central office switch. With larger Centrex implementations there may be access nodes on the customer site's fiber connected to the central office to reduce the dependence on copper bundles to the central office.

The main components that comprise the Centrex service are:

- Centrex lines (consisting of copper pairs access nodes or directly to the central office switch and its line card ports).
- Analog or digital handset.
- Centrex features (e.g. Caller Line ID or Call Center features).
- PSTN connectivity.
- Voice mail (optional).

The Centrex line charge is a monthly cost that can range from as low as $20 per month for large Centrex customers with multiyear contracts up to $40-$50 per month for small Centrex customers. This cost varies from carrier to carrier and is often reduced by one to five-year contracts. Contracts are usually written with minimum line commitment thresholds. Often Centrex involves contract cancellation penalties which can diminish the return on investment for planned migrations that are outside of the end of lease contract dates.

The analog or digital hand set is the responsibility of the customer in most cases. While in the past the phone companies rented sets, most have moved out of that business. This means that enterprise customers must purchase or lease their phones. Costs for hand sets vary, but even in large volumes they can be as much as $250.

Centrex features provide enhancements like Caller line ID, Call Center Features, and Simplified Message Desk Interface (SMDI) that are chargeable. (Note: SMDI defines a way a phone system provides voice-mail systems with the information needed to intelligently process incoming calls.) Customers have the choice of using a hosted voicemail service from the carrier or using their own voice-mail system.

When purchased from the carrier, voicemail is a shared system based on services located at the central office. Customers subscribe to the service on a monthly basis per account. If a customer chooses to source their own voicemail system, it is typically located at the customer premises. Connectivity is achieved through special analog Centrex lines and an SMDI link. These analog lines are typically more expensive than regular Centrex lines and the SMDI link is a significant cost.

The cost of Centrex moves/adds/changes (MACs) is also significant. With many customers averaging 1-2 changes a year, MACs add considerably to the three year total cost of ownership.

IP technology solutions overview

VoIP

The term Voice over Internet Protocol, or VoIP, has been used as a catch-all phrase in the industry to refer collectively to a large group of technologies designed to provide Internet-based communications services. More accurately, VoIP refers only to the underlying transport protocol that encapsulates voice traffic or voice media streams and allows them to be carried over data networks, using IP network technologies or internet protocols. VoIP, however, is not IP Telephony, nor is it the more widely used industry terminology called IP Communications that refers to an even broader definition of communications networking applications and technologies.

VoIP can be understood as simply a transport protocol for carrying voice over any packet network, usually between sites. The term *convergence*, also sometimes referred as a multi-service network, refers to the integration of data, voice, and video solutions onto a converged network infrastructure.

IP Telephony vs. VoIP

IP Telephony refers to call processing and signaling technologies that are based on the open Internet protocol family of standards that provide end-to-end voice, data, and video communications services. These standards have been defined by the Internet Engineering Task Force (IETF) and the International Telecommunications Union (ITU) to provide interoperable networking and communications services for public carrier networks and for the Internet.

The key point is that IP Telephony is more than simply VoIP (transport of voice over an IP network) because it also involves a larger family of communications standards needed to deliver voice and video services in the enterprise using open packet telephony. IP Telephony generally refers to the use of the H.323 signaling protocol used to setup, control and manage voice and video sessions. Because these services can be easily deployed within a converged network, using standard layer 2 and layer 3 network technologies, they provide significant benefits over traditional voice circuit switch network technologies. A summary of these benefits from a technology viewpoint include the following:

- IP Telephony allows the communication call processing services to be located anywhere on the network and to use packet networks, rather than the traditional Time Division Multiplex (TDM) networks for communications services.
- IP Telephony, unlike Hybrid IP-PBX technologies, allows for services to be delivered over a completely converged network, so that dual wiring and cabling and network equipment for, and connections to, PBX or Hybrid IP-PBX equipment is not required.
- IP Telephony allows support of communications services for voice media in addition to a variety of media types and modalities—web, e-mail, instant messaging, video, and conferencing services.
- A "true" or "pure" IP Telephony system provides the capability to carry traffic across different geographies, across and between many vendors, spanning many countries, because they can interoperate with a variety of Internet technologies and existing telephony technologies more flexibly, with greater benefit and reduced costs.

These benefits and capabilities are in contrast to limitations with existing voice legacy technologies that are built with TDM technologies. With legacy voice PBX and Hybrid IP-PBX systems, there is much greater expense and difficulty to: 1) accommodate differences in disparate technologies and equipment; 2) traverse geographic boundaries; 3) manage many sites centrally; 4) change the way resources are used on the network; 5) traverse regulatory boundaries; 6) deliver new communications services using different media types; and 7) provide the level of integration, ease of use, ease of access, and ease of management found in an IP Telephony system.

IP Communications (Beyond IP Telephony)

The foundation of a converged network are the capabilities and tools that allow an enterprise to flexibly, securely, and cost effectively carry any combination of data, voice, and video packets across the same links, using the same switching, routing, and gateway platforms. The properties of scalability, resiliency, fault tolerance, security, flexibility, and manageability are inherent in the converged network. To gain the advantages of the converged network, applications use the underlying services of network intelligence to ensure quality of service, availability, reliability, and security.

Applications also extend and amplify the capabilities of a converged intelligent network when they are built to use the underlying IP networking protocols and are based upon a server or network appliance model.

Beyond IP Telephony that primarily provides "dial tone" and "multi-media video conferencing services" using IP protocols, the term IP Communications refers to the additional robust suite of communications applications and technologies that take advantage of a converged IP communications network infrastructure.

IP Communications applications include applications such as Rich Media Conferencing, Unified Messaging and IP Contact Centers. These applications, like IP Telephony, eliminate the barriers of time zones and geographic distances between physical sites and organizations. By using IP Communications applications, governments can better connect constituents with services, while enhancing the capability and value of those interactions by allowing for greater customization, support, and personalization of services. Clients can obtain faster, easier access to services. Services can be shared across organizations and departments. Improved communications, collaboration, and operational processes can be enabled throughout governmental entities. In short, IP Communications improves the ability of the organization to leverage its resources to serve constituents flexibly and cost effectively.

IP Communications, then, is the broader term that refers to the entire suite of communication applications and the intelligent converged network capabilities that are built to work together with IP networking protocols.

Benefits of IP Communications over a Converged Intelligent Network

The benefits of IP Communications applications over a converged intelligent network are derived from a series of fundamental capabilities within IP networks that provide for the advantages of economy, flexibility, resilience, and productivity.

Economy

As opposed to connecting elements and applications of a communications system using expensive legacy voice technologies such as DS1 and DS0 line cards, trunk cards and digital signaling technologies, IP Communications networks allow customers to build network communication services based on IP networking technologies using Ethernet economics, often called silicon economics, or the application of Moore's law for the historical delivery of rapid advances in information technology or computing performance.

Looking at the economies of IP Communications versus traditional voice and PBX technologies, the cost to connect a typical enterprise PBX system with connections to the PSTN can be found in the cost of ports, cards, and circuits. One Ethernet port can replace 50 or more legacy voice circuits, line cards, and chassis equipment needed to provide equivalent service. The key point here is that the costs are significantly less to provide connections to other sites and to other applications. IP Communications is more flexible because it allows the use of broadband and voice technologies to support communications and systems needs.

Flexibility

As opposed to connecting elements and applications of a communications system using legacy technologies that are proprietary, monolithic, and restrictive in nature, IP networking allows connections to be made with virtual reach—resources to be distributed anywhere as needed; economies to be gained by centralization of gateway resources, circuit, and server resources; and the use of many types of media and applications to be brought together to facilitate communications within an organization. IP Communications systems are also more capable in supporting mobility requirements, telecommuting, moves/adds/changes, centralized management, outsourcing operations, extension mobility, desktop integration, front office, back office integration and applications, enterprise directories, and taking advantage of emerging web innovations and services such as instant messaging, presence, and mobility. Beyond achieving a higher degree of security with the application of data networking technologies for secure voice, video, and data, IP Communications are vastly superior over legacy voice technologies in deploying and integrating wireless LAN applications, IP video surveillance, IP video on demand, streaming video, video conferencing, and rich media conferencing applications.

Resilience

With business continuity and disaster recovery high on the agendas of many organizations, the resiliency of connectivity and abilities provided by IP Communications to keep the organization connected make it an ideal candidate for survivable services. Redundancy is built into intelligent layer 2 and layer 3 networking technologies and applications. Clustering and hot standby technologies, fault tolerant storage technologies such as RAID, dual power supplies, and UPS systems are now common in the industry. Internet protocols offer superior failover, redundant and self-healing capabilities that are easy to deploy, open standards based, and can support not only voice, but all of an organization's communications services. The fact that the most resilient military and enterprise communications systems can now use IP Communications and Internet protocols to achieve five nines of reliability and availability provides a superior alternative to rigid voice technologies. These legacy technologies are far more expensive, and are unable to provide the overall system resiliency needed for as broad a range of services and applications as can IP Communications.

Productivity

With the shift of focus moving from savings to enabling end users to become more productive with applications that help them accomplish higher quality communications more quickly and easily, it is easy to see why IP Communications is superior to legacy voice technologies. Custom IP phone applications can be provided that use any existing web or enterprise database on an IP network. End users can take advantage of open enterprise directories; e-mail systems for sending, receiving voice mail, fax, and e-mail messages; and use general tools for programming communication rules. Voice recognition technologies and softphone support at the desktop can be added to an IP Communications environment. While all of these possibilities also exist for legacy technologies, they are more expensive, less scalable, and more difficult to deploy.

Building Blocks of Converged IP Communications Networks

Typically, the following components are required to build a converged IP Communications network:

Network Infrastructure

This includes the intelligent switches, routers, and specialized components (such as gateways, services, and software) that form the physical infrastructure and deliver intelligent network services such as security, quality of service, and resiliency to the IP Communication infrastructure and applications.

Applications

The real power behind convergence is found in the new capabilities provided by integrated data-voice-video applications such as Rich Media Conferencing, Unified Messaging, and IP Contact Center applications. These applications work better together in a secure IP Communications network due to the trunk-less and port-less nature of IP Communications, as well as having the ability to use the innovations of web-enabled technologies and applications that continue to drive communications innovations and productivity.

End Points (Client Devices)
The access points for users to take advantage of applications. These can be IP phones, personal digital assistants (PDAs), mobile phones, and applications for personal computers such as softphones.

Call Processing
The heart of any IP telephony system is call processing software that can run on network appliance servers or third-party servers.

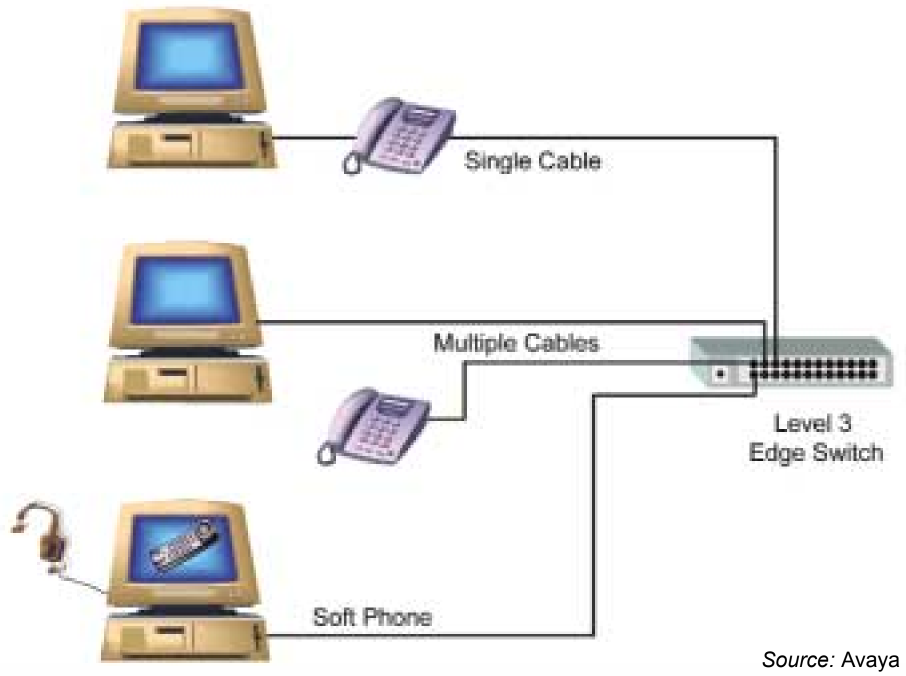
Major IP Communications Solutions

IP Telephony
IP Telephony provides the benefits of advanced dial tone and enterprise class telephony features and capabilities, improved support for mobility and call processing services over a converged packet network.

IP Telephony thereby eliminates the need, cost, and expense of running separate voice and data networks. Because IP Telephony is more flexible, open, and adaptive than legacy voice environments, it also reduces expense, provides for better and more productive applications, and improves an organization's ability to extend capabilities on its' network. IP Telephony also supports voice in addition to data applications that can be delivered to the IP phone or softphone from web services, backend / front end databases, or office systems. IP phone service applications include enterprise directories, emergency alerting, Amber Alerts, support for N11 services, kiosk applications, and custom applications.

IP Telephony allows organizations to reuse their existing network infrastructure. With IP Telephony, employees can use IP hand sets or softphones anywhere on the corporate network. Generally only a single Ethernet port is required to provide both desktop and IP Telephony (voice) services. Power over Ethernet using the 802.3af standard or pre-standards based power can be used with great advantage throughout an IP Telephony environment to power other network appliances such as 802.11a/b/g wireless access points, IP video surveillance cameras and IP phones. IP Telephony systems also provide the advantage of lower cost moves/adds/changes administration allowing employees and departments to more easily move from location to location without the burdens of administration and service interruption. (See Figure 4.)

Figure 4. *Three Different IP Phone Configurations*



Source: Avaya

Because IP Telephony works with packet networks, the model of support used for IP Telephony can be the same as the one used for the desktop, thereby reducing and simplifying staffing requirements. Since the IP phone or softphone is an intelligent endpoint on the network, it has the intelligence to seek out communications services from one or many IP Telephony call processing services. In addition, clustering and high availability networking technologies provide for a much more business resilient environment at a far lower overall cost.

As mentioned above, IP Telephony can be used to: 1) supplement; 2) extend; or 3) replace existing voice services that are provided by legacy voice technologies.

Applications of IP Telephony include connecting it to an existing PBX environment and using IP Telephony and its applications to support mobile workers and telecommuting applications with a combination of softphone and VPN services. Because workers can be more flexible and IP Telephony can support extension mobility (hotdesk) applications, enterprises and organizations can save real estate costs while increasing the productivity of their workforce. Typically, productivity can be increased from 10 percent to 20 percent while increased flexibility and reduced costs from IP Telephony can account for an additional 15 percent to 20 percent of cost savings. In some cases, the cost savings are very high. This is true in Greenfield sites (new sites), where the single IP telephony cabling infrastructure, single network infrastructure, and one set of common operations are required to support data, voice, and video services. Most organizations plan their migration to IP Telephony based on key events or needs to evolve their infrastructure to IP Communication over time.

Typical requirements for an IP Telephony solution are the converged network components consisting of switches and routers that provide for quality of service (QoS), security, in-line power, and virtual local area networks to provide control and separation of voice, video and data traffic, as well as wireless traffic. In addition, management tools and protocols that support the ability to ensure multi-service networking traffic such as real time and streaming video can also be supported and accommodated with appropriate QoS tools. The ability to support a robust set of gateway technologies to enable connections to any broadband technology, or PSTN technology, (voice interface cards and WAN interface cards) and support for protocols that handle multicast, and/or network content distribution, caching and filtering (for video on demand, etc.), as well as secure wireless technologies, are other considerations for evaluation of the converged network infrastructure.

Considerations for Deploying IP Telephony

When deploying IP Telephony, there are several prerequisites that should be considered within a network readiness assessment.

- Does the organization have category 5 wiring that supports full duplex? (note: category 3 wiring is also possible, although not recommended)
- Can the access switches support in-line power?
- Are the switches capable of supporting priority queuing?
- Is the organization currently using 10/100 switched networking technologies within the infrastructure, or still using something less than this?
- How many sites will ultimately be tied together and supported with one IP Telephony server (or server cluster)? One? Three? Five? One Hundred?
- How are these sites networked together now?
- What is the bandwidth between these sites?
- Is the bandwidth QoS enabled in terms of latency, delay and jitter?
- Is there sufficient bandwidth to support the voice between the sites?
- What type of PBX technology will the IP Telephony system be connected to?
- Is voice mail or unified messaging services also needed?

Extended applications of IP Telephony also include support for Enhanced 911 services so that IP phones and IP devices are automatically located on the network, regardless of their physical location. IP

Telephony also provides support for extension mobility, also called hoteling or hot-desking, providing the ability of any employee to have their extension appear on designated IP phones. Softphones provide the ability to use virtual private network (VPN) technologies and PC desktops or laptops to gain access to corporate communications services from anywhere in the world without incurring toll charges.

In summary, IP Telephony generally provides an organization with a higher level of productivity and flexibility, and reduced capital and operating cost. In addition, IP Telephony, as an application, is deployed to:

- Better handle growth, change, and complexity.
- Support more economical communications with Toll By Pass, VPNs and broadband technologies.
- Providing ongoing operating savings with network simplification—infrastructure convergence.
- Provide for ongoing savings, efficiencies with staff convergence and coordination.
- Improve the overall quality of all communications services in terms of availability, security and flexibility.
- Support the convergence of management functions (centralized management) on the network.
- Improved workforce mobility and productivity.
- Increase business resilience, continuity and provide for improved disaster recovery.

IP Conferencing

IP Conferencing, or IP Audio Conferencing, provides organizations with the benefits of using secure on-net IP Conferencing resources to save significant expenses over using alternative legacy voice conferencing equipment connected with TDM equipment and circuits or conference services that are supported by a service provider. The advantage of IP Conferencing is found in the fact that the conferencing resource can be located anywhere on the network and can use existing IP converged network bandwidth, as opposed to expensive stand-alone voice circuits to support the conferencing service, including the ports and cards to connect, support and terminate conference calls. This is generally true, whether the legacy voice conferencing service or IP conferencing service is outsourced or not.

IP Conferencing can be supported either with software or hardware resources and is typically provided in terms of the number of simultaneous ports (users) of conferencing that the system can support. With IP Conferencing, resources can be more easily deployed, pooled, aggregated from anywhere on a converged network, and more easily accessed and used by any user from any location. IP Conferencing generally rides on IP WAN networks that already exist within the enterprise, meaning that essentially IP Conferencing rides for free, or for relatively little incremental cost, over what is already provided.

Typically, IP Telephony systems and IP endpoints have a certain basic amount of IP Conferencing support built in. This is one of the additional advantages of an engineered IP Communications system vs. a Hybrid IP-PBX. In general, the basic IP Telephony features that support IP Conferencing provide anywhere from three to six sessions of IP Conferencing in ad-hoc IP conferencing scenarios. This means that any IP phone user can add up to five additional parties, using IP conferencing by simply hitting the IP conferencing button over an IP network. This capability is provided within the IP Telephony system using IP phones or softphones.

With IP Conferencing, speakerphone capabilities built into endpoints also enhance the overall productivity of workers using conferencing services. For reservation-less and scheduled audio conferences, an IP Conferencing bridge or resource is needed. This resource can be provided with software or hardware resources.

As a general rule of thumb, most organizations would require anywhere from 5 percent to 10 percent of their employees to be IP conference ports. In larger organizations, nearly 50 percent of all voice traffic minutes are terminated on conferences, so this number may be higher.

The most significant benefit of IP Conferencing is increased workforce productivity. When employees are provided with conferencing resources that are available, accessible, and easy to use, productivity goes up. From an organizational perspective, IP Conferencing provides the benefit of predictable and near zero cost of operations, after the initial investment for hardware or software IP Conferencing resources, since the service is supported on the converged IP network. IP Conferencing services are also more easily grown, changed, and supported.

IP Contact Centers

IP Contact Centers provide organizations with the benefits of using server based resources that integrate the functions of Interactive Voice Response, Automatic Call Distribution, and Call Flow Scripting into general purpose service based appliances that can be added to an IP Network to support sophisticated customer service applications. Within the public sector environment, IP Contact Centers allow local, state and federal government agencies to deploy integrated shared services, 211/311 services, and web portals to serve their constituencies.

Because IP Contact Center services use web based technologies and use IP to connect components of the customer contact center together, they are not subject to the distance limitations or the prohibitively high costs of maintenance and support as found in traditional voice legacy contact center technologies. This allows IP-based Interactive Voice Response/Automated Call Dispatching (IVR/ACD) and Customer Contact Center agents to be located anywhere on the network and to use IP Networking technologies in addition to PSTN technologies to support contact center operations.

IP Contact Centers allow for media blending, global queues, and global reporting. Media blending is the ability to support voice, web, and e-mail with a common process and customer contact in-box. This allows one set of agents the ability to respond more quickly to all forms of communications with one virtual blended case queue.

IP Contact Centers provide the ability to support skill-based routing, time of day/day of week routing, and centralized reporting over an IP network. This allows sites and various contact center operations to be consolidated or distributed more easily and contact center operations as well as agents to work from anywhere in the IP converged network. Call Flow scripts can be updated, and with web based administration, operations can be centralized. New services can be deployed consistently, quickly and easily to sites anywhere in the world. IP contact centers also provide the ability to support continuity of operations more easily and cost effectively with resiliency built in, and the ability to outsource contact center operations to other countries or geographies. Lastly, IP Contact Centers provide the ability to achieve much higher levels of staffing efficiency, and workforce productivity with superior integrated desktop tools, better access to databases, and the ability to consolidate operations over a converged IP Communications network.

Unified Messaging

Unified Messaging (Unified Communications) is an IP Communications application that streamlines workers' message management burden and provides for increased productivity and responsiveness. Having to check messages in a variety of formats all day long—e-mail and voice-mail on multiple phones—can be a full-time job in itself.

Collapsing all of these messages into a single mailbox and allowing employees to retrieve those messages in the format handy to them at a given time ("listening" to e-mail, "reading" voice-mail, for example) is a huge time-saver. The Radicati Group, a research firm in Palo Alto, California, estimates that unified messaging generates 25 to 40 minutes of additional productivity per employee per day.

Like IP Contact Centers and IP Conferencing, and other applications of IP Communications, Unified Messaging also allows organizations to use the portless and trunk-less model when provisioning communications services in an IP Communications environment. That is, rather than being required to use expensive line cards and digital voice circuits, when riding over a converged network, supporting voice, fax, and e-mail messaging requirements are much more economical. Ethernet is one to two orders of magnitude less expensive than purchasing bandwidth using DS0 or DS1 technologies, and IP Networking is much more flexible.

Unified Messaging also allows an organization to take advantage and reuse assets that they already have in place to support voice messaging requirements. For example, if an organization is using Microsoft Exchange or Lotus Notes e-mail servers, they can also use these message stores to support storage for their voice and unified messaging needs. This allows an organization to centralize their voice messaging resources to save additional capital and operational expenses from equipment, operations, and software licenses. At the same time, Unified Messaging also "voice enables" the e-mail infrastructure, so that employees can now receive, listen to, and respond to important e-mail messages, from their cell phones, any IP phone, any web portal, or any public phone.

Not only are common server and software message stores much less expensive for storage of voice messages than their proprietary voice counter-parts, they also can be managed, centralized, distributed and support a wider variety of media and different types of messages with a common in-box. Additionally, Unified Messaging can be further extended with voice recognition technologies that allow access to corporate directories, conferencing services, or personal address books that are voice directed and voice activated, in addition to single number reach capabilities and alerts that can be provided to any cell phone or pager for message management. Unified Messaging comes with built-in time of day/day of week automated attendance and scripting rules that can further automate messages, group distribution lists, and broadcasts to departments, workgroups and outside callers.

Like IP Communications and softphones, Unified Messaging can be accessed from anywhere over an IP Network using VPN services. It provides much stronger support for mobility workers and telecommuters while saving on long distance costs. Unified communications generally integrates with, and reuses, familiar desktop e-mail clients and directory applications such as the Microsoft Outlook Client, the Lotus Notes Clients, Active Directory, and IBM Lotus Address Books.

Rich Media Communications—Integrated Audio, Video, and Web Conferencing

Rich Media Conferencing is an extension of IP Audio Conferencing. However, it also provides the integration of rich media applications—voice, web, video conferencing, instant messaging and presence services, as well as document sharing and desktop sharing to support collaboration, meetings, lectures, virtual teams, webinars, and conferences.

Like it's counterpart IP Audio Conferencing, the benefits of using secure on-net Rich Media Conferencing resources saves significant capital and operating expenses over using alternative legacy voice conferencing equipment and separate services of web and audio and video that is connected with TDM equipment and legacy voice circuits.

Rich Media Conferencing services can be located anywhere on the network, and can use existing IP converged network bandwidth, as opposed to expensive stand-alone voice circuits to support the conferencing service, including the ports and cards to connect, support and terminate Rich Media Conference calls.

IP based Rich Media Conferencing services can support ad-hoc reservation-less or scheduled conferences and easily integrate with TDM voice PBX equipment to provide these services to existing PBX or service provider environments where needed. Services of Rich Media Conferencing can be accessed from IP phones, desktops, or from legacy voice terminals and desktop web browsers. It is not required that web-audio and video services be used on any given conferencing call. It is also not required that services for desktop, application, or document sharing are used on any given conference call. In Rich Media Conferences, participants see each other, see who is attending, see who talking, can participate in text chat with each other; can easily schedule these conferences from existing e-mail, calendaring, or Instant Messaging client tools; can participate and collaborate in shared workspaces; can have breakout sessions; and can control the interaction of the virtual meeting, including the ability to automatically record and play back meetings.

Rich Media Conferencing provides significant productivity benefits to the organization to the degree that resources are made easily available and accessible to employees. With IP, the costs of these services after initially deployed are relatively small when reusing the converged IP Communications network.

IP Videoconferencing

Three additional IP Communications services that further extend the capabilities of an IP Communications environment include IP Videoconferencing, Video Telephony, and streaming communications services such as video-on-demand and streaming audio-on-demand.

IP Videoconferencing provides an organization with the ability to use real-time, two-way, or multi-party, multi-site video conferencing services using H.323 protocols over a packet network. The advantages of using IP Videoconferencing are flexibility, higher quality, greater availability and reliability, and much lower equipment and operating cost over traditional ISDN videoconferencing network technologies and services. Because IP Videoconferences can take advantage of existing data networks and lower cost H.323 endpoints and equipment, IP videoconferencing can support a greater number of users and more flexibly with desktop or dedicated videoconferencing terminals. IP Videoconferencing solutions also integrate with and interoperate with existing H.320 (ISDN) videoconferencing systems and endpoints, using H.323 to H.320 gateways where needed. This allows anyone using any type of technology to easily participate in IP Videoconferencing calls.

Because IP Videoconferencing systems are much less expense to deploy and operate, they provide a very flexible and economical way for an enterprise to video-enable its infrastructure. Videoconferencing systems must be designed carefully, however, due to the higher bandwidth requirements that video services typically place on a network, and also take into account whether conference calls are using lecture model / broadcast style, or many-to-many style communications. The issues of how voice and video traffic are handled over the network with intelligent networking protocols such as multicast, RSVP, call admission controls, and dial plans using gatekeepers, switches and routers in the network are also important. The services of IP Videoconferencing are flexible and can be extended easily to accommodate new locations as needed. IP Videoconferencing systems are generally comprised of gateways, videoconferencing bridges, gatekeepers and endpoints, in addition to the converged network.

IP Video Telephony

IP Video Telephony extends the integration capabilities of an IP Communication environment by providing integrated video with voice technologies that automatically use IP Videoconferencing services for converged desktop video endpoints, integrated video endpoints, or H.323 video endpoints, as needed. For example, with video telephony, video is just a phone call. Video is available with every phone call, and ad-hoc videoconferencing at the touch of a button is available for every phone call. In other words, the level of integration is much higher with video telephony that automates the usage and user experience with video telephony features being automatically provided by the underlying IP Videoconferencing resources.

Extension Mobility

Extension mobility allows users to log into any IP Phone and receive their own phone numbers and privileges at that location. Extension mobility is often called hoteling or hot-desking. When used together with softphones, employees also have the ability to use VPN technologies and PC desktops or laptops to gain access to corporate communications.

Because extension mobility allows one IP phone to be shared among many workers when in the office, it can consolidate and reduce or eliminate the need for facilities workspace, saving real estate and facilities costs.

Unlike traditional PBX systems, IP phones support plug and play moves/adds/changes that allows users to take their IP Phones to new locations, plug them into the Ethernet jack, and have all user privileges and settings re-established when the phone is registered on the network. This reduces administrative costs and service interruptions that are generally experienced during moves. One additional advantage of converged IP Phones is that certain models can also support the IP Phone device and the desktop using a single Ethernet drop (one jack) for both IP phone and the PC desktop while still providing the treatment of switched (segmented) data traffic needs of 10/100 to the desktop.

IP Telephony Applications

Ease of deploying new web-based communication services that can be integrated with existing systems, databases, or third party tools easily and quickly is an additional advantage of an IP Communications system. By using standardized development application programming interfaces (APIs) and standardized web development technologies such as extensible markup language (XML), hypertext transport protocol (HTTP), JAVA, Java Telephony Application Programming Interface (JTAPI), and Telephony Application Programming Interface (TAPI), organizations can more quickly, easily, and cost effectively deliver Computer Telephony Integration (CTI) applications and desktop integration. Applications can be customized to support the process and workflow needs of a specific workgroup or department and be deployed as horizontal enterprise-wide applications. Examples include applications such as inventory management, medical transcription, executive information, time card applications, bulletins, HR functions, notifications, broadcasts, and alerting.

Applications can be provided so that an IP phone doubles as a kiosk for customer service operations in retail environments, such as reaching customer services, accessing local events, or purchasing tickets on-line more easily and quickly.

The key point is that with IP Communications, the IP phone supports more than just voice. It can also support a wide variety of open desktop and IP phone applications available from third parties, or those that can be developed by the customer, and managed, delivered and accessed from anywhere in the IP Communications environment. Such applications would include:

- Videoconferencing (wired and wireless)
- Instant messaging (wired and wireless)
- Whiteboarding—An area on a display screen that multiple users can write or draw on, enabling visual as well as audio communication.
- Collaborative browsing (also known as *co-browsing*)—A software-enabled technique that allows an employee to interact with a customer or another employee by using their Web browser (controlling it remotely) to show them something.
- Call forwarding and intermittent call transfer—Allows for the transfer of calls to the teleworker's cell and home phones.
- Enhanced Web Surfing—Integrating phone numbers embedded in Web pages with a software-based phone so you can click on and dial any phone number you see while surfing the Web.
- Interactive Voice Response (IVR) applications for constituents.
- Audible E-Mail—Text-to-speech capability for retrieving e-mail by phone.
- Call routing—Involves the routing of calls using rules-based call handling, e.g., you can have calls go to your voice mail during a meeting and have them sent to your cell phone in the afternoon.

Mobility Applications

Softphones/Soft-agents

Softphones/soft-agents (Customer Agent Desktops) can be used from any laptop or PC workstation connected to the IP Communications environment. Combined with VPN technologies or soft VPN clients and soft-tokens, softphones provide a very cost effective solution for support of mobile teleworkers and for telecommuting applications. Voice, video, and data services can be accessed from remote offices, home offices, hotel rooms, and international or domestic locations to avoid international and toll costs. Softphones also support corporate directories, or personal address books that can be stored on network servers, or on the local desktop machine. The advantages of deploying softphone and soft-clients further extends the power and productivity benefits of an IP Communications environment by allowing workers to expand their zone of productivity and ease of access to communication services.

In the case of soft-agents, with IP Contact Center applications, it provides for the ability to locate agents anywhere on a converged network, out-source operations, and save on real estate costs, in addition to the benefits of achieving continuous innovation in the communication application because it is software rather than hardware based. Lastly it allows organizations to reuse and leverage their existing resources and assets in desktop and converged networks.

802.11a/b/g Wireless LANs and Wireless or Soft IP Phones

On average, according to a study conducted by *NOP World Technology*, wireless LANs enable users to be connected to network resources 1.75 additional hours per day, which translates into the average user being as much as 22 percent more productive. Given a reported average salary of $64,000 for a professional worker, this indicates that the annual productivity improvement per user is worth, on average, $7,000.

For a modest cost ($300 to $500 per person, or $1 to $2 / per person / per day, including equipment, installation, training, and annual support), organizations can extend their existing wired networks to locally mobile professionals in a campus setting using wireless LANs. The wireless network extensions have comparable speeds to the very high speeds of the wired network, making the wireless LAN basically an equivalent, portable version of the enterprise network. By extending the worker's productivity zone, wireless networks protect organizational investments in existing wired networks because those networks become accessible to greater numbers of users for more of the time.

These benefits are further amplified by the fact that 40 percent of enterprise class laptop computers now ship with wireless LAN capabilities already included. Although there were approximately 2,000 wireless LAN hot spots in the United States in early 2005, this number is expected to grow to at least 6,000 by the end of 2006, according to *Gartner*.

In short, wireless LANs will further extend the benefits of IP Communications by increasing the zone of productivity in terms of ability to access corporate data and communications resources and support the mobile professional with:

- Increased productivity;
- Increased responsiveness;
- Improved business resilience; and,
- Increased utilization of IP Communications and technology assets.

Now that wireless LAN standards have matured and important security enhancements have been added to them, a wireless LAN user's flexibility to stay on top of communications throughout the workday rather than waiting till 5 p.m. to deal with a day's worth of messages could supply a colleague or a constituent with vital information in time to complete a transaction or avoid some unnecessary delay. Multiply this by hundreds or thousands of workers across an organization, and the payoff adds up quickly.

Teleworker / Support

Organizations are constantly striving to reduce costs, improve employee productivity, and keep employees within the organization. These goals can be furthered by providing employees the ability to work from home with similar quality, function, performance, convenience, and security as are available in the office. Employees who are occasional or full-time teleworkers require less office space. By providing a work environment in the residence, employees can optimally manage their work schedules, allowing for higher productivity (less affected by office distractions) and greater job satisfaction (flexibility in schedule). This transparent extension of the enterprise to employee homes is meeting the objective of a robust teleworker solution. The capabilities further extend IP communications network capabilities by providing IP mobile access for vehicles and people in the field (e.g., state patrol, case workers, etc).

To summarize the benefits of the teleworker voice and data solution, this solution extends the advantages of VPNs (e.g. cost savings, data application support, extended availability, security, and privacy) to provide secure enterprise voice services to full-time and part-time teleworkers.

Extended IP Communications Applications

Emergency Alerting Applications

Many state and local governments are gaining significant value from applications such as Berbee's Informacast emergency alerting application, and similar third party XML broadcast applications that can provide immediate broadcast alerting functions to all government offices using open IP alerting functions that have been integrated with the IP Communications system. These applications can support scheduled broadcasts, alerts, and bulletins or ad-hoc emergency alerts throughout all emergency responder and state, local, and city government offices using IP networking technologies. Unlike TDM technologies, alerts can be customized, including visual and verbal system-wide or group paging functions to ensure fast and effective homeland security and emergency alerting services are delivered throughout a community or jurisdiction.

Business Continuity / Disaster Recovery

Land and Mobile Radio Convergence

Land mobile radio convergence provides for the consolidation, integrated management and communications across heretofore separate, disparate and isolated radio and legacy voice technologies that provide for emergency fire, police, and marine radio services. IP Communications and IP Multicast networking technologies allow for the cost effective convergence of these legacy radio technologies, and for streamlined process convergence to improve coordination between agencies. By using IP applications and networking services to converge and combine these radio networks onto one more logical IP Conferencing bridge, advance coordination, monitoring, receive only, and/or transmit services can be provided for ad-hoc, meet-me, and scheduled IP conferencing services.

These services use IP media streaming services and H.323 standards to support interaction between existing analog, digital voice technologies, and disparate radio systems that can be integrated within an IP Telephony communications environment using IP phones and desktop applications. The ability of IP protocols to span geographies and connect communication independent of physical limitation on specific hardware enables improved flexibility and scalability to support convergence of communication services using disparate legacy technologies.

N11 Services—211, 311, 511, 711 Services to Relieve Overburdened 911 Systems

The unique benefits of IP Contact Center technologies and applications to combine voice, data, web, and e-mail traffic of client requests from various sources can facilitate and streamline the delivery and implementation of shared e-government service portals, thereby streamlining the number of governmental employees required to support N11 services, while providing higher levels of flexibility and automation for these services.

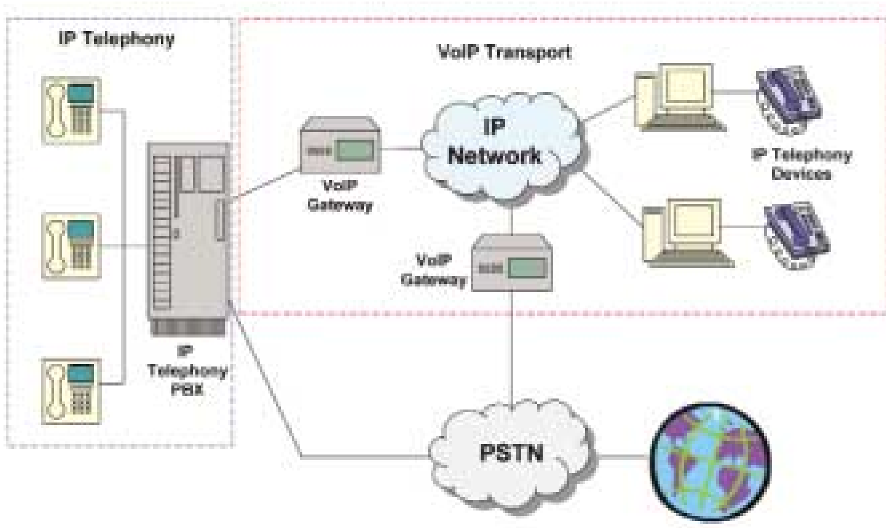
Video/Audio-On-Demand, E-Learning

Video-on-Demand and Audio-on-Demand are used to provide training to employees using e-learning and streaming video-on-demand network services deployed on the IP Communications network infrastructure. The value of these services is that they can be deployed anywhere on the IP converged network to support e-learning, training, staff development activities, and knowledge management for the organization. Streaming IP services can be delivered to IP endpoints and desktops that support standard streaming and Windows Media Player protocols.

IP Telephony Implementation Considerations

If you implement IP Telephony, you will still need to connect to the PSTN for certain services and to have access to phones that are not part of your IP telephony deployment. As mentioned earlier, the PSTN supports a variety of voice services through the signaling control points. An Internet telephony device wanting to connect with one of these services must use PSTN signaling for the foreseeable future. This, and the need to connect to PSTN customers, leads to a hybrid PSTN/VoIP configuration. (See Figure 5.)

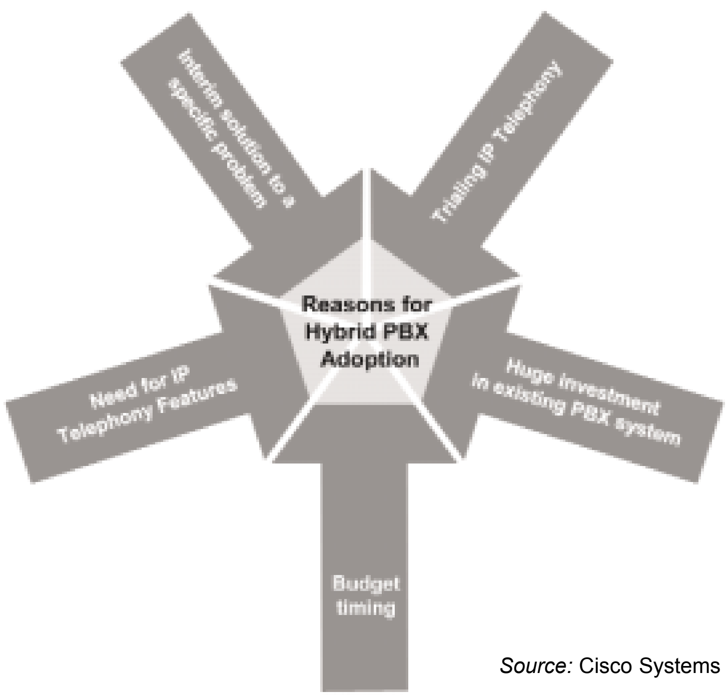
Figure 5. *Hybrid PSTN/VoIP Configuration*



Source: www.voip-info.org

In considering the adoption of IP Telephony, you may choose an incremental implementation approach for a variety of reasons. The implementation can be phased to coincide with the termination of equipment leases or service agreements. There may also be drivers for the IP implementation that would prioritize IP telephony for certain work groups or locations (e.g. call centers or remote offices). (See Figure 6.)

Figure 6. *Reasons for Hybrid PBX Adoption*



Source: Cisco Systems

A "Typical" VoIP Configuration

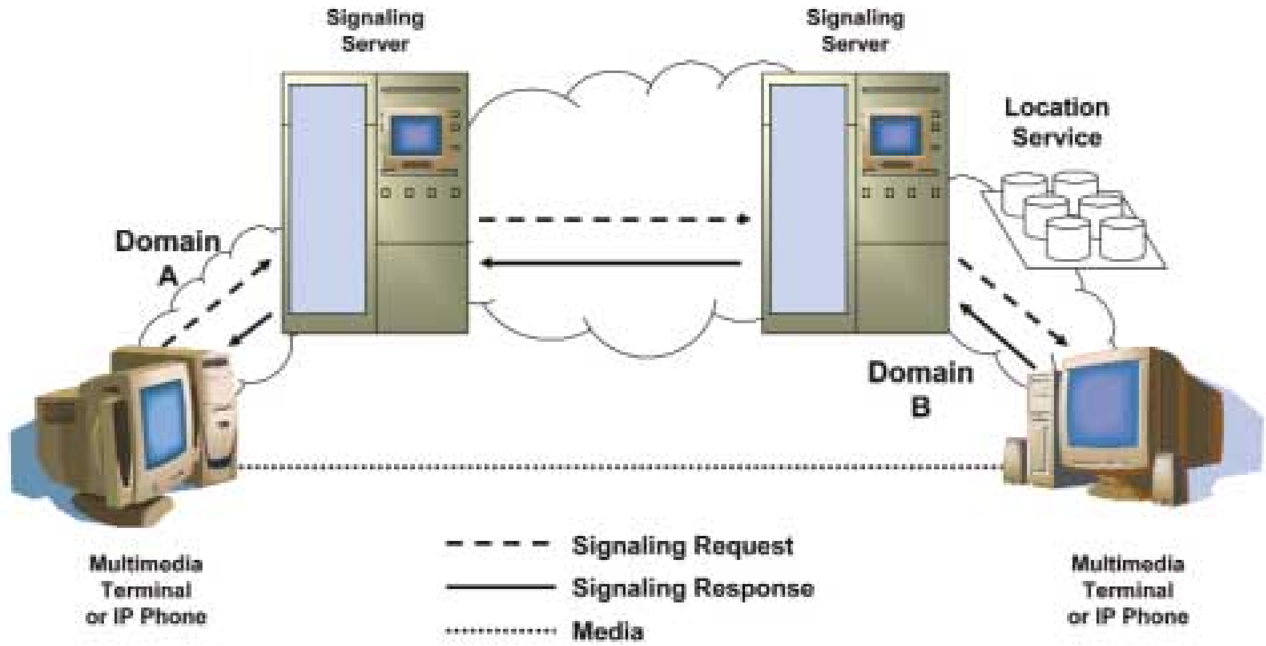
A typical VoIP system consists of a number of different components:

- Gateway/Media Gateway
- Gatekeeper
- Call agent
- Media Gateway Controller
- Signaling Gateway and a Call manager

The gateway converts media (e.g. voice, video) provided from one type of network to the format required for another type of network. For example, a gateway could terminate bearer channels from a switched circuit network (e.g. DS0s) and media streams from a packet network (e.g. RTP streams in an IP network). This gateway may be capable of processing audio, video and T.120[1] alone or in any combination, and is capable of full duplex media translations. The gateway may also play audio/video messages and performs other Interactive Voice Response (IVR) functions, or may perform media conferencing. The call control "intelligence" is outside the media gateways and is handled by media gateway controllers (also called call agents). These elements synchronize with one another to send coherent commands to the media gateways under their control. A control protocol is used to control VoIP gateways from the external call agents. (See Figure 7.)

A media control protocol is implemented as a set of transactions composed of commands. The call agent may send commands to gateways that create, modify, and delete connections, or that create notification requests and auditing commands. Gateways may send notification and restart commands to the call agent. All commands include text-based header information followed by (optionally) text-based session description information.

Figure 7. *Simple VoIP Configuration*



Source: www.voip-info.org

[1] T.120 supports multiple users in conferencing sessions over different types of networks and connections.

the economics of VoIP

Selecting the Right Products for Investment Protection and On-Going Management

In migrating to IP telephony, one of the larger issues in selecting a vendor would have to be investment protection and upgradeability. It is important to know where products purchased are in their lifecycle and the future path for upgrades and growth. Past performance is generally indicative of future activity, unless the future product plans are explicitly stated. Be aware of the vendor's on-going support for existing products. How long will products of this scale be supported by each vendor being considered? Have they traditionally been upgradeable with minimal (or with any) hardware and/or software, or is the upgrade path a complete replacement? Related to this issue is the frequency of maintenance upgrades. This can be a double-edged sword, as new features and functions are nice, but doing large scale upgrades frequently can be troublesome for the staff as well as the end users. How much testing is done before maintenance releases are issued? Understanding any potential changes in how the network will react is imperative to adequately providing information to the users affected.

Another consideration is the maintenance contract itself. How long are maintenance releases, and maintenance calls in general, and are they included at no cost after the initial purchase? If it is determined to be the best path, can the maintenance function be brought in-house? What implications will this have on the contract for maintenance releases and releases involving added features or functionality? Fully understanding future direction following deployment may aid in determining the right equipment provider.

The individuals managing the network will need access to reports ranging from the standard voice reports covering traffic patterns, trunk usage, etc., to the IP reports covering dropped cells and trunk utilization. The reports necessary to manage the VoIP network will be similar to the reports in use today for monitoring the voice network, though there will be additional reporting criteria in addition to the traditional voice reports that are required to adequately manage the network. All of the reports will have to be created, as the sources for the information will be entirely different. It is imperative these reporting capabilities are available when the system is deployed to aid in trouble avoidance and resolution.

The Business Case for IP Communications—Return on Investment (ROI) and Total Cost of Ownership (TCO)

Because a VoIP migration project may come under considerable internal and public scrutiny and throughout its three to five year evolution as new services are added, it is important to put in place some basic metrics that will be used to communicate and measure the success of convergence initiatives over the migration timeframe, and beyond. Ideally, these will be outlined and described at appropriate levels for executive, management and constituents, to describe the value of a converged communication system in terms of what it provides for the organization.

Because converged IP Communications networks provide capabilities that can be used by many communications applications including data, voice, video, wireless and application systems, it creates many synergies to achieve measurable benefits. For example, with common network infrastructure, end-to-end security management needs, mobility needs, and new productivity applications can be more easily added and extended to the workforce. Knowing services are deployed from a baseline, in terms of what capabilities exist, their ongoing costs, how difficult to expand, change or grow, and their impacts on staffing, labor, workflow and process can set the stage for useful metrics and a plan that clearly aligns with key priorities and transformational objectives.

Some categories of metrics and ROI areas to look at when considering the use of new technologies, using IP communications technologies include the following:

- Cost reductions;
- Centralization of services;
- Reduction of infrastructure;
- Reduction in cost of PSTN services;
- Administration of Moves/Adds/Changes (MACs);
- Remote Administration;
- Interoperability;
- Productivity improvements;
- Investment Protection with reuse, extension, evolution of intelligent converged network and applications; and,
- Optimized staff (consolidated telecommunications staff, reduction in training required over the long term).

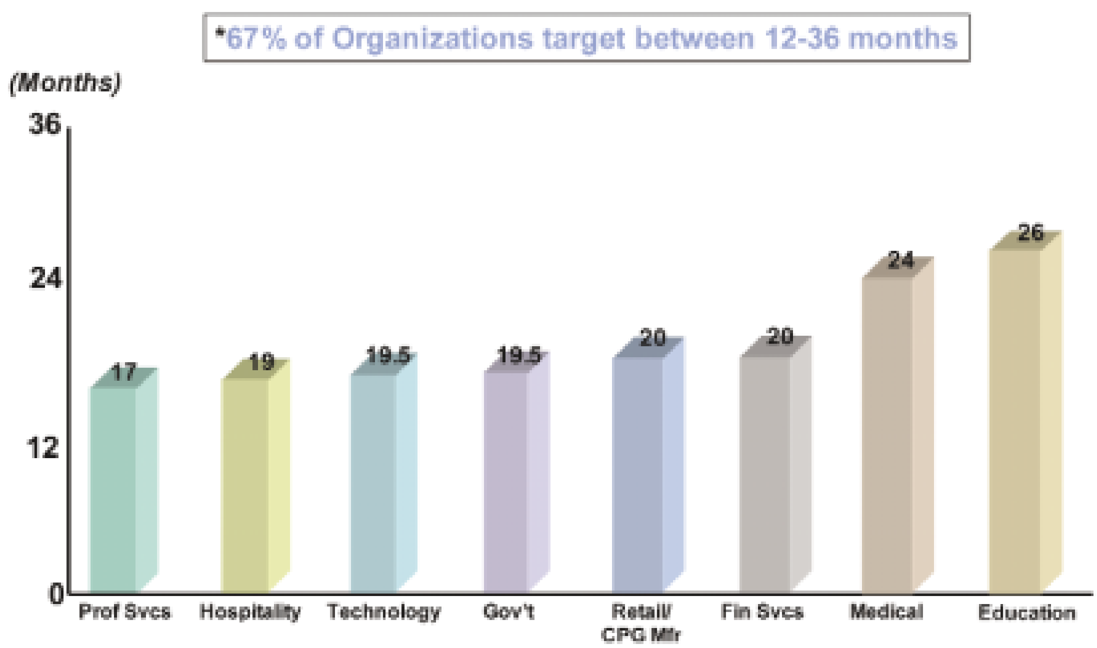
A single converged infrastructure incorporates common resources for security, resiliency, multiple converged applications, and converged communications. This can create multiple logical networks on the single network. IP communications also provides the ability to use carrier based transport for state and local governments to create their own transport network using technologies such as metro-Ethernet or using available dark fiber that is typically state owned. The addition of optical network capacity and services to use these assets can translate to significant savings and benefits in terms of a state's ability to deliver a broad range of services and new applications over time.

There are also anticipated savings from:

- Using a common unified security / management infrastructure.
- The ability to more easily gain new technology features, which are more simple and less expensive (than PBX technologies) for software upgrades and the type, range, richness, and economy of applications deployed.
- Productivity improvements that can be measured in terms of FTE hours, labor and staffing costs, and workforce productivity benchmarks.

For the average payback of VoIP by vertical industry, (See Figure 8.)

Figure 8. *Average Payback of VoIP by Vertical Industry*



Source: Cisco Network Investment Calculator (CNIC), March 05

The delivery of IP communications productivity applications and support for collaboration, rich-media, teamwork, unified messaging, mobility, wireless, and teleworkers reduces costs and supports a more distributed, effective workforce. While this is well documented in the industry, specific metrics for any given project or initiative must be established. The metrics must be reasonable, valid, and accepted by all key stakeholders for the results being sought.

The benefit of IP communications can be realized when any of the following organizational scenarios exist:

- Maintaining multiple equipment, personnel, and budget for multiple networks (e.g. voice and data).
- Existence of many branch offices, especially international sites, where traditional PBX installations and toll fees can be expensive.
- Undergoing change through growth or a planned move to new physical office space, where PBX installations are costly and hook-up time is critical.
- Reliability and durability of communication is required in case of emergency situations. IP was designed with this in mind, whereas when a destroyed or damaged PBX will sever communication.
- Heavy call center usage that may benefit from IP Contact Centers to streamline operations, increase customer satisfaction, and tie together web and back-end systems.
- Need to quickly update employees with critical information in a more efficient manner.
- Deploying an IP communications application such as unified messaging, rich media conferencing (audio, web, video), wireless, support for mobility, teleworkers, telecommuting.

There are several ROI metrics that can be captured effectively once a baseline is established. This will provide measurements prior to, during, and periodically throughout a migration to IP communications. They are:

<u>Savings from Reduced Network Infrastructure Costs</u>

- Expense for new sites.
- Remote site network equipment.
- Equipment upgrade and replacement cost.
- Ethernet wiring drops: an IP Phone and PC share a single jack.
- International and domestic interoffice call charges.
- Number of external communication lines, especially to the public switched telephone network (PSTN).

<u>Savings from Improved Administrative and Operational Costs</u>

- Improved productivity of network support staff. For example, unified messaging increases employee productivity by allowing users to access voice mail from their PC and e-mail messages from their IP phone. Independent studies demonstrate that employees save 20 to 40 minutes per day processing messages versus managing two separate systems.
- Ten to forty percent productivity improvement after convergence.
- Shift staff focus from administration to value-added projects that leverage expertise.
- Consolidated help desk.
- The cost of moves/adds/changes is significantly reduced from $75 to $125 each.
- Onsite support and maintenance contracts.
- Centralized call processing reduces branch office administration expense.
- Ongoing network design, project management, and implementation.
- 80 percent of large enterprises view the deployment of IP-enabled, integrated contact centers as a key converged application.

Maturing VoIP standards and technology are making the concept of deploying converged infrastructures much more viable. When making the decisions to implement VoIP transport and/or IP Telephony, it is helpful to understand the drivers associated with each and the experiences of similar organizations.

The typical VoIP Savings include:

- Elimination or reduction of PSTN toll charges.
- Elimination or reduction of service and support contracts on existing PBX hardware.

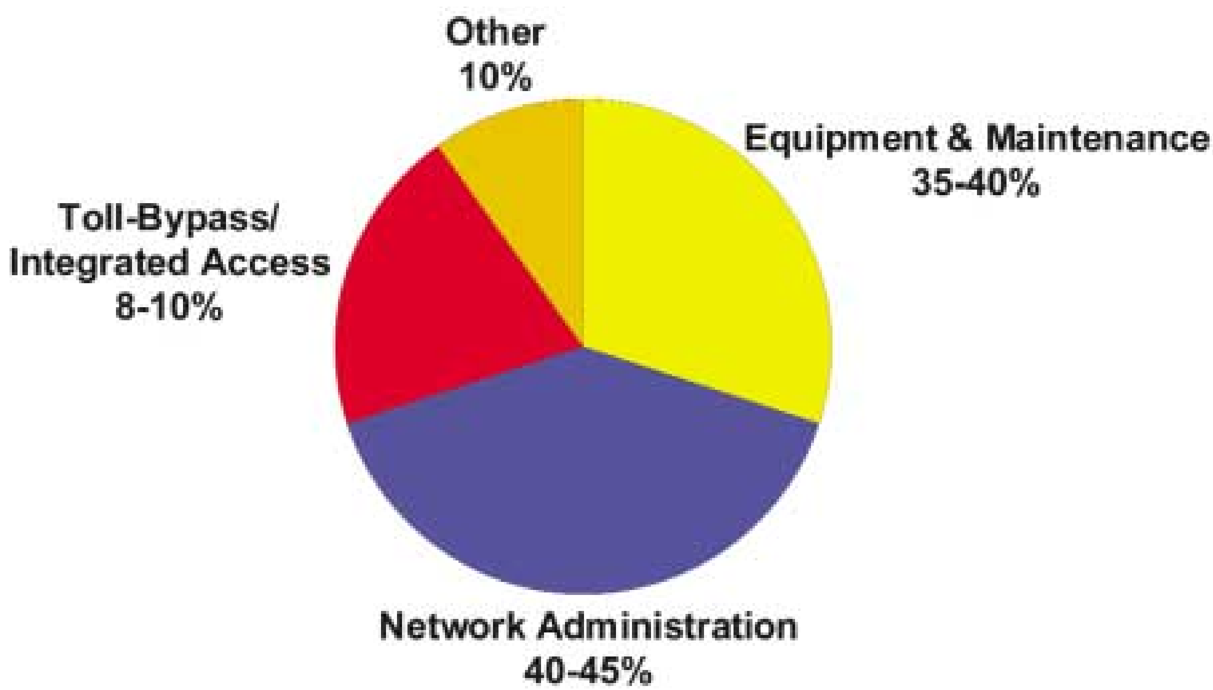
- Elimination of the requirements for Centrex services and charges.
- Collapsing voice and data infrastructures support, and management resources into one converged network to reduce the need for staff, maintenance, and upgrades and realize a reduction in the costs associated with moves/adds/changes.
- Reduction in the on-going costs for the support and maintenance of separate voice messaging systems.
- Improved productivity for remote and mobile workers by extending the enterprise to them (e.g. providing them with the same integrated capabilities as office workers) from any place on the Internet.

The experience of previous deployments indicates four major sources of savings from VoIP and IP Telephony implementations. (See Figure 9.)

Wholesale replacement of PBXs is extremely rare. It is much more common to see federated decision models in organizations with political subdivisions, segmented funding streams, and dispersed geographies. Finding the balance is the goal here.

Many enterprises with voice and data convergence projects underway are consolidating voice and data staff, usually under a CIO. This trend has proven to be the most difficult culturally, but also is the most rewarding from a business standpoint. CIOs must be creative as they assimilate teams. Research shows a converged staff creates its own culture and will consider a variety of resolutions to a business problem, regardless of the incumbent technology vendor, as opposed to being convinced that the next release of "product X" will solve it. As a result, converged staffs tend to be more business-solutions-oriented and regard technology as a tool to accomplish this.

Figure 9. *Sources of Savings from VoIP and IP Telephony Implementations*



Source: Cisco Network Investment Calculator (CNIC)

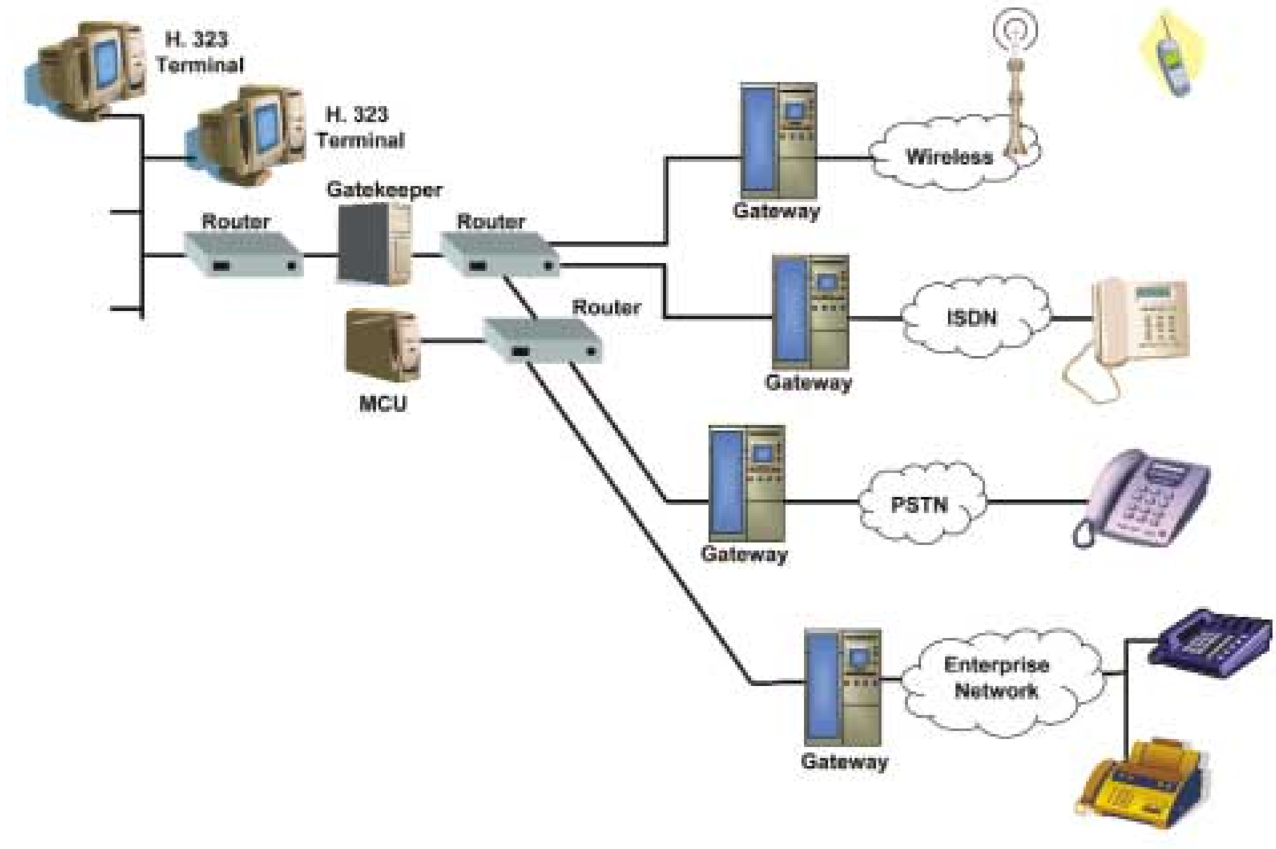
VoIP implementation and planning

The amount and type of IP Telephony products and services will be unique to an organization's current telephony configuration and future needs. Additionally, there are multiple configurations for connecting IP phones to the network. (See Figure 10.)

Converged networks extend IP networks to leverage a common infrastructure for voice, video, data and all other converged communications. Major telephony equipment vendors have stable IP-based solutions to complement their traditional TDM voice switching gear. Network infrastructure vendors have optimized their products specifically for converged applications and are working closely with application vendors to fully exploit the underlying intelligence in the network. Organizations have a choice of proven solutions and are not locked into a single vendor for a highly functional end-to-end solution. The move toward a converged communication infrastructure should be incremental and protect investments in existing legacy infrastructure to ensure a smooth, low risk transition.

Voice traffic is converted to IP telephony packets by different devices, depending on the architecture of the solution. In toll bypass applications, gateways

Figure 10. *Connectivity Options*



Source: www.voip-info.org

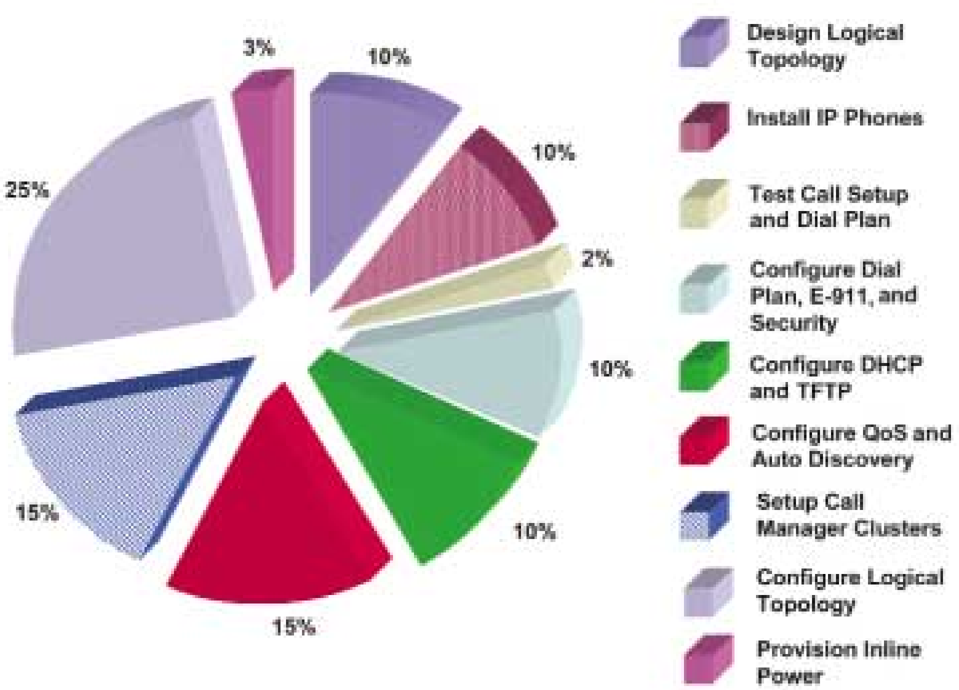
convert voice between the PBX and IP network. In most IP telephony deployments, packet conversion occurs at the IP phone as well. In the case of architectures involving analog phones and phone hubs, IP telephony conversion occurs at the phone hub. Most IP phones use industry standard protocols such as SIP or H.323 to communicate with other IP devices. However, similar to TDM-based PBX and certain proprietary handsets, not all IP phones will work with just any vendors' IP call manager due to proprietary client/server protocols.

For an organization that is considering voice and data convergence, putting IP telephony on the WAN is as important as IP telephony on the LAN. IP telephony on the WAN is where the advantages of toll bypass become evident. The reasons for this are primarily economical. Cost savings can be immediate when long distance phone calls are diverted from PSTN and sent over an existing IP-based WAN. Implementing an IP telephony infrastructure needs to be an evolutionary step with proper consideration for legacy TDM PBX systems and adequate planning. The total planning time is entirely dependent on the size and complexity of deployment. The individual planning considerations and average percentage of time spent on each aspect of planning are represented below. (See Figure 11.)

Centrex Replacement

There are numerous examples and tools available to help justify the cost savings of moving from Centrex. In general, most customers can achieve significant cost savings and additional ROI benefits from superior telephony services, higher productivity and greater flexibility and resiliency with IP Communications. Savings of 30 percent to 50 percent or higher, with payback periods averaging 48 months or less, are common. Examples of recent deployments and ROI include:

Figure 11. *IP Telephony Planning Estimates*



Source: Cisco Systems

- City of Mississauga
http://newsroom.cisco.com/dlls/prod_012203b.html
- Alabama Department of Public Health—Case Study
http://www.cisco.com/en/US/netsol/ns339/ns395/ns359/ns363/networking_solutions_customer_profile09186a0080134142.html
- City of Southfield—Case Study
http://newsroom.cisco.com/dlls/ts_120303.html

Common Centrex Migration Concerns

Once customers have decided to deploy IP Communications solutions, careful considerations must be given to how the migration will take place. A key consideration for most large implementations (e.g. greater than 200 phones) is that a flash cut is not practical—the organization must plan on migrating to IP Communications over time.

Migration planning will involve the need to support the Centrex service and any legacy voice equipment as well as the IP Communications solution during the migration period. The management and planning for migration and operations within a hybrid environment will involve additional steps and considerations. Planning should also take into account that there will be a mix of Centrex users that are unaware which other users have migrated to IP Telephony. To provide an overall user experience that is as seamless as possible, the issues of interoperability and feature transparency between the Centrex environment, the legacy PBX voice equipment and the IP Communications system needs to be carefully considered, with people, processes, and project management that ensures a successful transition.

In a mixed or hybrid telephony environment involving Centrex and IP Communications equipment, the following list of features can be used as a high level checklist for features that should be supported between IP phones, between IP phones and Centrex phones, and between IP phones and the PSTN.

- Basic telephony features, dial tone, dial-tone multi-frequency (DTMF), multiple lines, hold, call transfer, call conference (ad hoc).
- Retaining telephone numbers on the Centrex service and porting these to the new IP phone with Direct Inward Dial numbers.
- Replicating Centrex dial plans, if standardized.
- Ability to support 4 or 5 digit dial plans.
- Ability to support IP phone to Centrex phone using existing 4 or 5 digit dial plan.
- Ability to support Centrex phone to IP phone using existing 4 or 5 digit dial plan.
- Ability to support local calling: 9 plus 7 or 10 digit.
- Ability to support long distance calling using 9-1 plus 10 digits or 8-1 plus 10 digits or 1 plus 10 digits.
- Calling number display between IP phones and Centrex.
- Calling name display between IP phones and Centrex.
- Speed dials (system and personal).
- Intercom features.
- Ring again.
- Uniform Call Distribution (UCD) hunt groups.
- Automatic Call Distribution (ACD) queues.
- Voice mail integration.
- Existing voice mail box preferred.
- Ability to support voice mail networking.

Centrex customers can acquire an ISDN primary rate interface, a Channel Associated Signaling (CAS) T1 service or analog direct inward dial (DID) trunk along with new DID phone number extensions. (Note: CAS is the transmission of signaling information within the information band, or in-band signaling. This means that voice signals travel on the same circuits as line status, address, and alerting signals. As there are twenty-four channels on a full T1 line, CAS interleaves signaling packets within voice packets. Therefore, there are a full twenty-four channels to use for voice.) These new telephone numbers or DID numbers would preferably be from the same exchange as Centrex, but it is not mandatory. The new trunks connect to the appropriate gateway that provides PSTN connectivity for the IP Communications solution, as well as calling number and name display support between the PTSN and Centrex service and the IP telephony environment. Voice mail and unified messaging requirements can be supported with interoperability supported for

SMDI integration, and for using one of the following technologies to integrate with the customer's existing voicemail system or with a Centrex Central Office based voicemail:

- Audio Messaging Interchange Specification—Analog (AMIS-A): This is a standard analog voice message networking method that relies on telephone channels for connection.
- Voice Profile for Internet Mail (VPIM)—VPIM brings voice and fax into the realm of Internet messaging technology, allowing Internet mail facilities to include voice and fax message exchanges.
- Digital Bridge—A device that connects and passes packets between two network segments that use the same communications protocol.

There is typically a per transaction cost for these services and it requires certification testing for each install with the carrier. This is important since the carrier generally will not allow non-Centrex lines to have Centrex Central Office voicemail boxes. Should the customer wish to keep their existing voice mail account, an SMDI connection will be required between the IP Communications system and the existing voice mail system.

Considerations for Centrex

While Centrex in the past has been an attractive option for telecom needs, the typical customer has to deal with the following on-going challenges:

- **High monthly operational costs:** Centrex involves tariffed monthly charges per phone which may include Centrex line, PSTN, phone feature charges, voice mail, moves/adds/changes, 911, and possibly handset lease or rental costs. These rates vary depending on options, contract term, and size of install.
- **Increasing costs of Centrex:** While the cost of IP Communications ownership has decreased over the past few years, Centrex prices are going up in many areas.
- **Not universally available:** Centrex availability is very much tied to whether it is available on the Central Office switch serving the customer site. Very often a customer within a single city or region may find that only some of their offices have access to Centrex, meaning that other offices have to install key systems, PBXs or business lines. This means multiple types of systems to support and different feature sets for different offices.
- **Lack of a coordinated dial plan:** Customers that have a hybrid of Centrex and key systems often suffer from not having a coordinated city wide 4 or 5 digit dial plan. This means sites not served by Centrex often have to dial seven or ten digits to call non-Centrex sites and vice-versa. Some key systems are sophisticated enough to perform digit manipulation to hide this fact from the end-user.
- **Dissimilar unlinked voice mail systems:** A potential issue with a mix of Centrex and key systems is that the users would often have different voicemail systems with different user interfaces and no ability to transfer messages between sites.
- **Cost and lead-times of moves, adds, and changes (MACs):** The largest cost and loudest customer complaints often surround MACs. Lead-times can be five or more days and sample costs run anywhere from $50 to $150 per MAC, often with a minimum charge per visit. Many organizations average one to two MACs per employee annually, at considerable expense.
- **Lack of control and speed over new feature deployment:** Centrex customers have to deal with very few innovations and very slow deployment of new features based on the reality that a central office switch supporting tens of thousands of customers cannot be changed often. Customers also have no control over when switch upgrades would happen meaning that a switch upgrade could happen in their most critical time (e.g. year end, election time, start of legislative session).
- **Feature charges:** While a Centrex line charge provides dial-tone and basic features, carriers often charge extra money for features. Examples include shared line appearances, second number appearances on a set, name display, music on hold groups, and hunt groups. These often average out to several

dollars per month per user but aggregate to large amounts of money across an organization. These charges can be reduced or eliminated with an IP telephony implementation.

VoIP Support for Persons with Disabilities

VoIP-based applications can also be a promising technology for people with disabilities. VoIP integrates the phone, voice mail, audio conferencing, e-mail, instant messaging, and Web applications like Microsoft Outlook on one secure, seamless network. Workers can use their PC, laptop, or handheld as a VoIP phone from virtually anywhere, with the same phone number, which benefits telecommuters, including those whose mobility is impaired and must work from home.

The biggest advantage for supporting the physically challenged is that everything can be accessed through voice, audio, or a combination of both. Hearing impaired employees can place or receive TTY-compatible calls from their computer without the need for a legacy TTY device. TTY, which uses tones to transmit typed conversations over phone wires, is now the main form of phone communication for the hearing and speech-impaired. In fact, one of the implementation issues in IP telephony may be the ability to support TTY functionality for employees with such devices.

With VoIP-enabled applications, hearing impaired workers can read their voice mail from their e-mail program in a fraction of the time it takes with a TTY, which operates at a slow 45 baud per second. Vision impaired workers can use a Windows-based application called a softphone in conjunction with an IP phone to hear audible caller ID, a missed-call log, and line status without the need to memorize or mark buttons on the phone.

Managers should think of VoIP as empowering workers with disabilities to perform their jobs better. However, setting up VoIP to use assistive-technology software is not necessarily a snap. Just being a technically savvy employee might not be enough. It often takes a trained telecom or information-technology manager to assemble a VoIP system for workers with individualized assistive needs.

Fortunately, some VoIP providers understand this and are working with third-party vendors to create accessible solutions. VoIP is based on open standards, which make it possible for assistive technology to work with IP software. An application at the U.S. Commerce Department dubbed "Informacast," supports an emergency-broadcast system. The tool simultaneously sends audio streams and text messages to multiple IP phones so that hearing and vision impaired workers will not miss important alerts, like fire alarms.

Manufacturers and access providers need to shape the infrastructure to make VoIP perform for those whose vision or hearing is impaired as well as it does for others. Then, vision and hearing impaired workers can experience what everyone else does: ubiquitous access to information, and finally, a functional option for screening their calls.

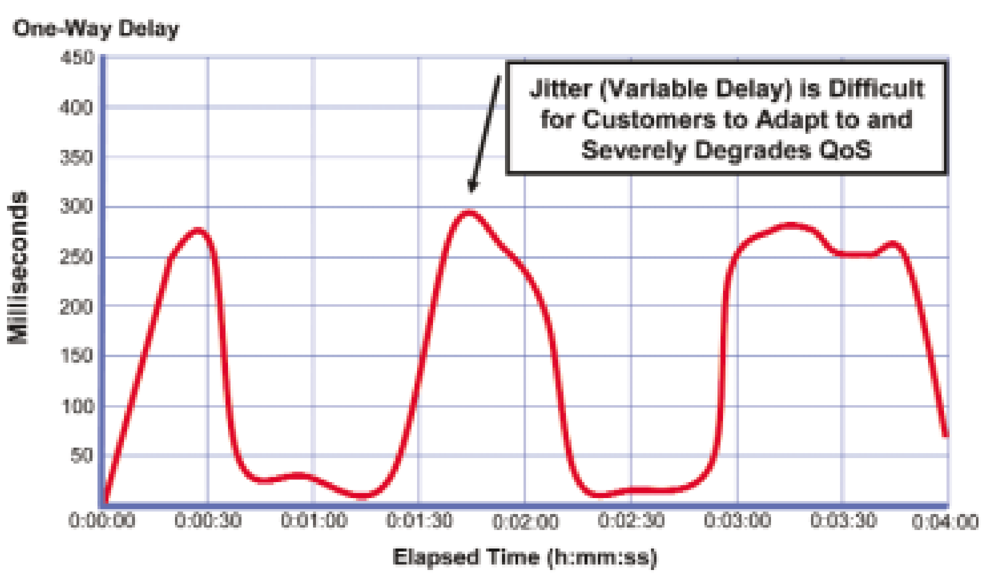
Regardless of which deployment model is chosen, the design of the IP Telephony network should ensure that telephony features are more accessible to users with disabilities, in conformance with Section 255 of the Telecommunications Act and 29 U.S.C. 794 Section 508. IP Communications can provide superior capabilities to provide system wide accessibility to users with disabilities because it supports unified messaging, additional desktop tools, and web based portals for facilitating communications to not only a few users who have special equipment provisioned, but for everyone on the system without significant additional cost.

VoIP implementation technical considerations

Network Infrastructure—Advanced Intelligent Network Features

A converged network requires a complete set of tools and integrated technologies for deploying and managing network service that support IP Communication services. These tools are used to provide end-to-end QoS for voice, video, data, and wireless applications, to ensure security and manageability, and to minimize network costs with support for resiliency and flexibility in layer 2 and layer 3 protocols and services that enable converged IP Communications services to work together with other network applications. Today's converged network technologies primarily consists of high speed layer 2 and layer 3 10/100/1000 switched networks and high-performance gigabit or terabit integrated switching and routing platforms. The design of converged networks has recently become more automated with sophisticated automation technologies to streamline the deployment of these requirements using auto-QoS and self-defending networking technologies, as well as improved management tools.

Figure 12. *Jitter*



Source: www.Voiptroubleshooter.com

Cabling

Although IP Communications services can be deployed over Category 3 wiring, a minimum requirement of Category 5E or above is recommended, since Category 5E wiring is a voice and data grade cable and has also been tested for Gigabit Ethernet. Category 5 wiring can also be used, but is less preferred; Category 6, the newest standard for cabling, can be used, but is not required. Generally IP Telephony and related IP Communications applications use a single Ethernet drop deployment model. This means that a single Ethernet port provides for desktop and IP phone support from a single outlet jack. In-line power is delivered to the IP phone, while a switch port in the IP phone allows data traffic to be logically and physically separated from the voice traffic, yet still support the desktop.

Network Infrastructure

IP telephony places strict requirements on IP packet loss, packet delay, and delay variation (or jitter). (See Figure 12.) Therefore, IP Telephony requires QoS mechanisms available from switches and routers throughout the network. For the same reasons, redundant devices and network links that provide quick convergence after network failures or topology changes are also important to ensure a highly available infrastructure for LAN, WAN, and the wireless network infrastructure.

LAN Network Design

Properly designing a LAN requires building a robust and redundant network from the top down. By structuring the LAN as a layered model using core, distribution and access layers that are typically used for designing campus networks, a highly available, fault tolerant, and redundant network can be designed.

Power over Ethernet (PoE)

PoE (or inline power), although not a strict requirement for IP Phone deployments, is highly recommended due to the network and administrative simplifications resulting for IP Phones and for other devices such as wireless access points, IP video cameras, and other IP devices supporting IEEE "802.3af" power standards.

PoE is 48 volt DC power provided over standard Ethernet unshielded twisted-pair (UTP) cable. Instead of using wall power, IP phones and other inline powered devices (PDs) such as Wireless Access Points can receive power provided by inline power-capable switches or other inline power source equipment (PSE). Inline power is enabled by default on all inline power-capable switches.

Deploying inline power-capable switches with uninterruptible power supplies (UPS) ensures that IP phones continue to receive power during general power failures. Provided the rest of the telephony network is available during these periods of power failure, IP phones will be able to continue making and receiving calls. Inline power-capable switches are deployed at the campus access layer within wiring closets to provide inline-powered Ethernet ports for IP phones, thus eliminating the need for wall power.

The use of Category 3 cabling is supported for IP Communications under the following conditions:

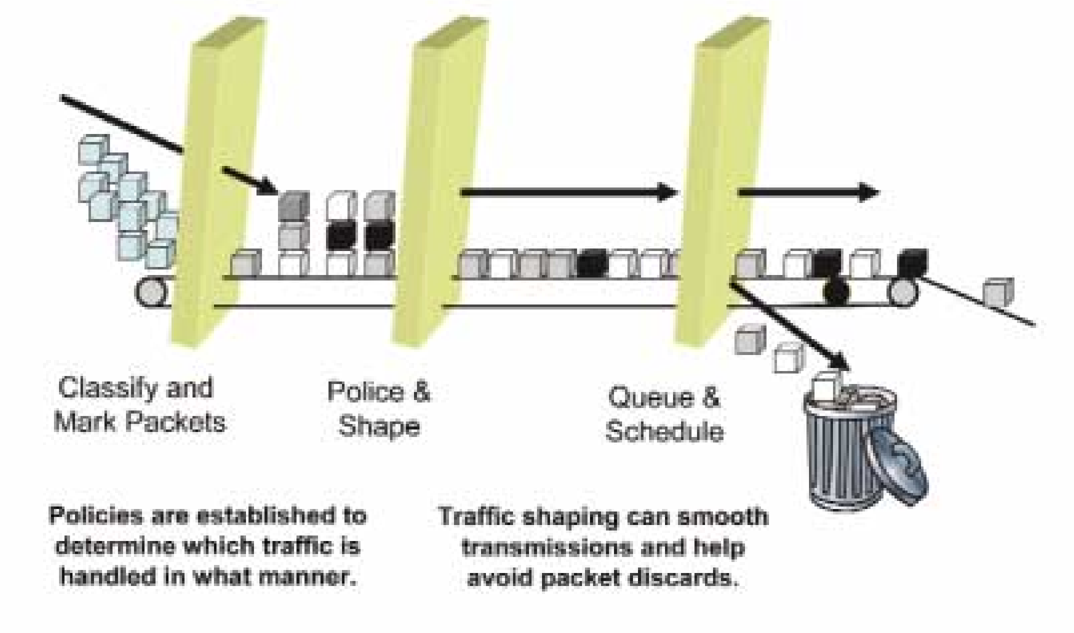
- The access layer switch port must either support 10 Mb only or be hard-coded to 10 Mb. When hard-coding switch ports to 10 Mb, it is also recommended that the port be set to full duplex.
- All devices plugged into the IP phone PC port should be configured manually to 10 Mb - full duplex.
- Devices on the PC port should not be allowed to run at 100 Mb because the uplink port to the access layer switch will only be 10 Mb.

Quality of Service (QoS)

The following types of QoS tools are needed from end to end on the network to manage traffic and ensure voice quality:

- **Traffic classification**—Classification involves the marking of packets with a specific priority denoting a requirement for class of service (CoS) from the network. The point at which these packet markings are trusted or not trusted is considered the trust boundary. Trust is typically extended to voice devices (phones) and not to data devices (PCs). In general, the recommendation is to classify or mark traffic as close to the edge of the network as possible. Traffic classification is an entrance criterion for access into the various queuing schemes used within the campus switches and WAN interfaces. The IP phone marks its voice control signaling and voice RTP streams at the source, and it adheres to requirements for voice, video, data, and signaling and control traffic. (See Figure 13.)

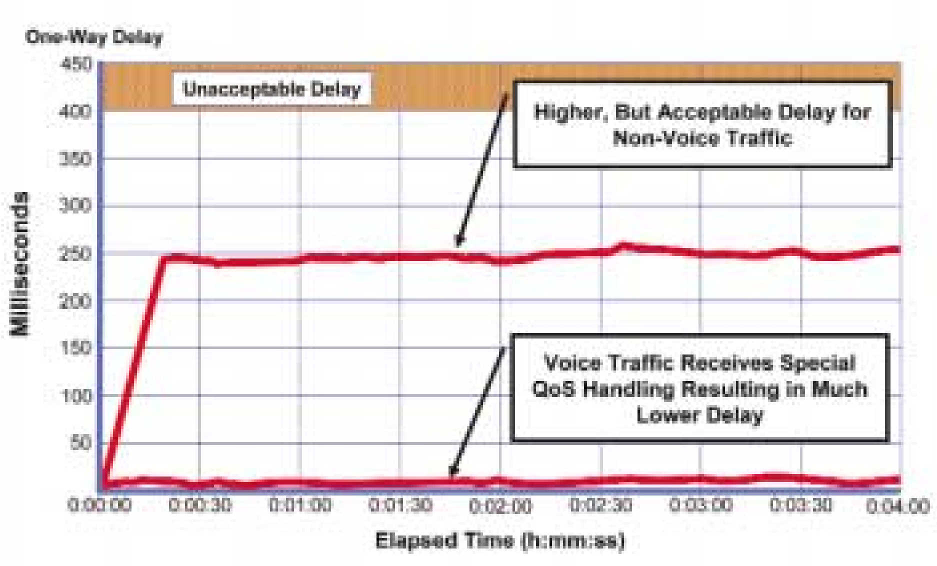
Figure 13. *Conditioning Packets to Ensure Quality of Service*



Source: www.voip-info.org

- **Queuing or Scheduling**—Interface queuing or scheduling involves assigning packets to one of several queues based on classification for expedited treatment throughout the network. After packets have been marked with the appropriate tag at Layer 2 (CoS) and Layer 3 (DSCP or PHB), it is important to configure the network to schedule or queue traffic based on this classification, so as to provide each class of traffic with the service it needs from the network. By enabling QoS on campus switches, you can configure all voice traffic to use separate queues, thus virtually eliminating the possibility of dropped voice packets when an interface buffer fills instantaneously. Although network management tools may show that the campus network is not congested, QoS tools are still required to guarantee voice quality. Network management tools show only the average congestion over a sample time span. While useful, this average does not show the congestion peaks on a campus interface. For converged networks, switches that have at least two output queues on each port and the ability to send packets to these queues based on QoS Layer 2 and/or Layer 3 classification are recommended. (See Figure 14.)

Figure 14. *Segregating Packets by Delay Characteristics*



Source: www.voip-info.org

- **Bandwidth Provisioning**—Provisioning involves accurately calculating the required bandwidth for all applications plus overhead. In the campus LAN, bandwidth provisioning recommendations can be summarized by the motto, *over provision and under subscribe*. This motto implies careful planning of the LAN infrastructure so that the available bandwidth is always considerably higher than the load and there is no steady-state congestion over the LAN links.

If QoS is not deployed, packet drops and excessive delay and jitter can occur, leading to impairments of the telephony services. When media packets are subjected to drops, delay, and jitter, the user-perceivable effects include clicking sounds, harsh-sounding voice, extended periods of silence, and echo.

When signaling packets are subjected to the same conditions, user-perceivable impairments include unresponsiveness to user input (such as delay to dial tone), continued ringing upon answer, and double dialing of digits due to the user's belief that the first attempt was not effective (thus requiring hang-up and redial).

Until recently, quality of service was not an issue in the enterprise campus due to the asynchronous nature of data traffic and the ability of network devices to tolerate buffer overflow and packet loss. However, with new applications such as voice and video which are sensitive to packet loss and delay, buffers—not bandwidth are the key QoS issues in the enterprise campus.

Auto-Management Functions

In many third and fourth generation converged network switches, the automation of QoS capabilities is provided by integrated switch software, in hardware and in network management tools. This simplifies the provisioning and on-going support for QoS and security configuration and management tasks. Additionally, automation of these functions improves consistency, and performance of the converged network due to the elimination of manual processes.

UPS and Power Backup

In general, two to four hours of UPS backup is desired for most locations. A secondary level of backup can be provided for major network operating center locations by providing backup generators for these locations in addition to UPS power. This is recommended for larger sites and centralized sites that power IP Communications servers and provide centralized gateway services to the PSTN or service provider networks with integrated switches or router modules, or use stand-alone gateway devices.

Multi-Layered, End-to-End Security Technologies

A systematic approach to securing a converged network is the best defense against attacks on the network infrastructure. Organizations must develop and employ comprehensive tools to rapidly identify, prevent, and adapt to security threats.

The dramatic spread of worms such as Slammer and Blaster highlights the need for multiple levels of defense, where many security functions interoperate to mitigate threats throughout the network.

A security policy defines who can do what, where, and when, while a security posture assessment can identify vulnerabilities and suggest fixes.

Security solutions and tools can be organized into the following three categories:

- Threat defense—watches for improper behavior in the network; examples include firewalls and network- and host-based intrusion detection/prevention systems (IDS/IPS).
- Trust and identity management—permits or denies services to devices and users based on policies; an example is a Remote Access Dial-In User Services (RADIUS) access control server.
- Secure connectivity—provides confidentiality across public links such as the Internet; for example, a virtual private network (VPN) with encryption.

A comprehensive security strategy involves approaching security solutions in "layers" so that exposure in any one layer or set of technologies is effectively mitigated by security protections and tools in other layers.

A comprehensive security solution requires management of network devices as well as security devices and solutions that span both. Generally, to provide a comprehensive solution, the following components are found in the converged IP Communications network:

- **Host Intrusion Detection / Prevention**—Host Intrusion Detection/Prevention software on each server can mitigate virus and worm attacks, as well as ease the patching and maintenance burden.
- **Firewalls**—Placing integrated firewall modules between zones efficiently enhances threat defense for the different application and server environments against directed and indiscriminate security attacks.
- **Network Intrusion Detection, Network Admission Control, Virus Protection**—Network access devices enforce admission control policy by demanding host security "credentials" from endpoints requesting access and relaying information to the policy servers; access devices then permit, deny, quarantine, or restrict access according to policy. Policy servers evaluate endpoint security information and determine the appropriate access policy to apply.
- **Identity Based Admission Control w/ Digital Certificates**—Enforce strict password rules with frequent password rotation. While the most secure options are one-time password or digital certificate systems, not all enterprises can afford them. Certificates, encryption and authentication tools can also be used for IP Applications software on the converged network to ensure that all endpoints, servers are running authenticated software.
- **Access Control Lists for Protecting Layer 2/3** —For Layer 2 networks use 802.1X port-based authentication. For Application servers, close unused ports on all devices and expire user passwords after a specified period; enforce strong password rules. In wireless

deployments, go beyond WEP when securing the wireless LAN, and regularly look for rogue access points. Conduct periodic security audits of the network, preferably by qualified, independent internal security specialists or a reputable third party. Configure access control lists (ACLs) to conform to security policies.

- **Network Segmentation**—Compartmentalize the network into security "zones" and define policies for each one, including access rules and rules for how zones interact. Use of VLANs enables zones to be created for the different tiers (for example, Web, application, and database) within multi-tier applications and between different applications.
- **Remote Access**—Where users require remote access to data center applications, use IP Security (IPSec) VPNs or Secure Sockets Layer (SSL) VPNs for partners. Also use VPNs for interconnecting remote data centers for backup and replication.
- **Other Security Network Technologies**—Using quality of service (QoS) features, you can minimize the impact of distributed denial of service (DDoS) attacks. This is done by configuring network-based application recognition (NBAR) to filter worms and unpermitted HTML traffic and by using Layer 2 security features such as Dynamic ARP Inspection (DIA) and Dynamic Host Control Protocol (DHCP) snooping to prevent spoofing. You should also disable router services that hackers commonly exploit such as Finger, BOOTP, and Proxy-ARP.

Implementation Considerations—Centralized, Distributed, or Autonomous?

One question that must be addressed is the extent of deployment of VoIP. Is this a centralized, distributed, or autonomous deployment, or is it a combination? If agencies manage their own LANs, it could well be that a VoIP system could be autonomous and deployed agency by agency. If this is the case, it would be in the best interests of all if established standards were followed so that the systems are compatible across the enterprise.

A distributed deployment would be the best option to provide the toll savings that is so often given as a reason to deploy VoIP. In this instance, numerous distant locations are networked together, and can achieve toll bypass when calling between any of the local calling areas on the network. A good application of this would be networking all of the offices for a single agency throughout the state.

A centralized deployment would aggregate the greatest number of individuals on the platform, hubbing the traffic to specific locations. This would be advantageous in passing VoIP traffic to a provider, which would take advantage of the bulk minutes used to get any decrease in toll rates.

The latter two options may require a phased approach. The larger the location, the longer it will take due to funding issues or deployment schedules. This should certainly be considered when making decisions; the option with the least impact to the end user should have an edge in that factor of the analysis.

Mix of Broadband, WAN, MAN, LAN, and PSTN

In evaluating the support for convergence networking options, the various WAN, MAN, LAN, PSTN and Wireless networking options should be considered. Typically, support for both digital and analog PSTN voice interfaces is needed for IP to PTSN networking gateways requiring data rates up to or exceeding T1/E1 speeds. The range, type and flexibility of Voice and Data Interface cards, modules within dedicated or stand-alone gateways should be evaluated. In additional to providing robust support for PSTN networking, routers, gateways and switches should be evaluated in terms of how many, and what types of wide area network, broadband network, and Local Area Network Interfaces they support. For WAN networking, are ATM, Frame Relay, IP, and MPLS supported? For the MAN (Metropolitan Area Networking), can metro dark fiber be used? Do the switches or routers support termination of fiber for GigE or metro-Ethernet transport equivalents?

With dark fiber and MPLS services frequently available within existing government owned or leased facilities, and often available from local and national service providers, the incremental cost of supporting many sites with common high speed networking infrastructure can be significantly reduced. Some approaches that governments are successfully using include having their own dark fiber, with optical connections now readily available in next generation intelligent networking equipment. Another approach is to use an existing spare lambda (light wavelength) on an existing fiber plant from a service provider. Many governments are also adopting a cost effective and flexible MPLS service that connects all of their sites together with quality of service guarantees. All of these approaches can dramatically lower the cost of converged networking infrastructure support costs, while enabling improved quality and increased flexibility to deploy progressive productivity gains from new applications over time.

Within the LAN, consider if the network is currently prepared to handle voice traffic. The rule of thumb for QoS capability is a 100 MB Ethernet backbone, but ideally, GigE uplinks and priority queuing capabilities within switching infrastructure should be present to properly address latency, jitter, and delay for larger multi-site and campus networks.

The cable plant will need to be professionally installed and certified for Category 5E or greater wiring. This will provide a physical layer that meets the minimum VoIP requirement. If utilizing GigE, all legacy Category 5 wiring will need to be tested and certified to support GigE. A service provider that supports Multi-Protocol Label Switching (MPLS) managed services will need to be considered if a QoS protocol (such as TDM/ATM) is not available.

Flexibility, Scalability, Resiliency, and Availability

Another aspect of VoIP service will be the how the network is designed and placement of servers in order to maintain a high degree of reliability. Diverse PSTN connections, meshed network access, diverse physical paths, backup power, database mirroring, are a few of the issues that must be addressed prior to any implementation.

Security

As touched upon in the Value Proposition section, having multiple agencies with partial ownership of the underlying routers and network provides complexity that will impact deployment. Security is one large area, as all of the agencies would need to work together. Each firewall encountered can cause variations in the output, some to the point of a failed call. Each installation should include a full analysis of the existing security that is in place, how that is provided, and how any VoIP equipment or traffic will react to that situation. Similarly, the agencies involved would need to determine policies and procedures to deal with troubleshooting, how, when, and who performs upgrades on the routers, so that the network as a whole can be maintained with the utmost of integrity. These steps would have to be done even if one department maintained ownership of the network, but in that event, the determination would be much easier as many of those processes would already exist.

Training, Support and Maintenance

Because converged communications applications tightly rely on close integration with the underlying intelligence in the network, it is a good idea to have a business continuity plan in place. Application developers and IT staff responsible for infrastructure need to be sensitive to the ripple effect that even a small change may have. Cross discipline training is always a good idea. Once fully deployed, your converged network will need to have 24x7x365 global support to avoid costly down time.

E911 Considerations

An E911 service has to provide automatic number and automatic location information (ANI and ALI) to a 911 public safety answering point (PSAP) when an emergency call is made. Most traditional PBX's are only able to provide this support with third-party assistance and a lot of administration. Local numbers must be updated regularly as new codes are assigned to phone companies, wireless carriers, etc. Some states have laws mandating private switch enhanced E911. The National Emergency Number Association (NENA) has a current list of such laws <http://www.nena9-1-1.org/9-1-1TechStandards/state.htm>.

To appreciate the unique issues of E911 support in IP telephony, consider how emergency calls are handled with a traditional PBX. When an emergency call is made, information is typically sent to a security staff relating the caller's physical location on the campus with their extension number. When an emergency team (e.g., police or fire department) arrives, an employee can meet the team and direct them to the emergency.

The issue to be solved here is that an IP phone can be moved without any centralized administrative intervention. A database has to be maintained to map the IP phone's unique Layer 2 addresses to a physical location now being served by a port on a switch. This is not the case with a traditional PBX, which just maps a port to a phone.

IP telephony offers two basic approaches to handling emergency calls: on-net to campus security or off-net to the carrier Point of Presence (POP). With on-net campus security, usually an individual in the company assists the 911 respondent; with off-net approach, the number and location of the individual in distress is made available to the PSAP.

Dial Plans

The dial plan architecture includes dial plan groups, calling restrictions, and on-net route patterns. This includes defining which gateway to use when someone makes a long distance call, or which PSTN trunks to use for domestic versus international calls. For a "least cost routing" example, consider the following: If someone in Dallas wishes to call off-network to a person in San Antonio, the call can be routed over the enterprise network and get handed off to the public phone network via a gateway in San Antonio, thereby being charged only for a local call instead of a long distance call from Dallas to San Antonio.

VoIP Vendors

Major telephony equipment vendors now have stable solutions to complement their traditional TDM voice switching gear. Network infrastructure vendors have optimized their products specifically for converged applications and are working closely with application vendors to fully exploit the underlying intelligence in the network. Enterprises now have a choice of proven solutions and no longer have to be locked into a single vendor in order to enjoy an end-to-end solution. According to August 2004 Gartner Research, the following vendors have established a credible presence in the North American IP Telephony market:

Leaders (Product portfolios and market strength affecting overall market trends)

Avaya
http://www1.avaya.com/SEO/IPTelephonyA/

Cisco Systems
http://www.cisco.com/en/US/products/sw/voicesw/index.html

NEC
http://www.neaxnet.com/

Nortel Networks
http://www.nortelnetworks.com/products/voip/pureip.html

Siemens
http://www.siemens.com/index.jsp

Niche Players (Strong product offerings for specific market segments)

3COM
http://www.3com.com/products

Inter-Tel
http://www.inter-tel.com

ShoreTel
http://www.shoretel.com/STCorp/

Toshiba
http://www.toshiba.com/taistsd/pages/prd_cti_main.html

Vertical Networks
http://www.verticalnetworks.com/

Visionaries (Innovative companies with the potential to influence the market)

Alcatel Networks
http://www.alcatel.com/

Mitel Networks
http://www.mitel.com/

VoIP regulatory environment

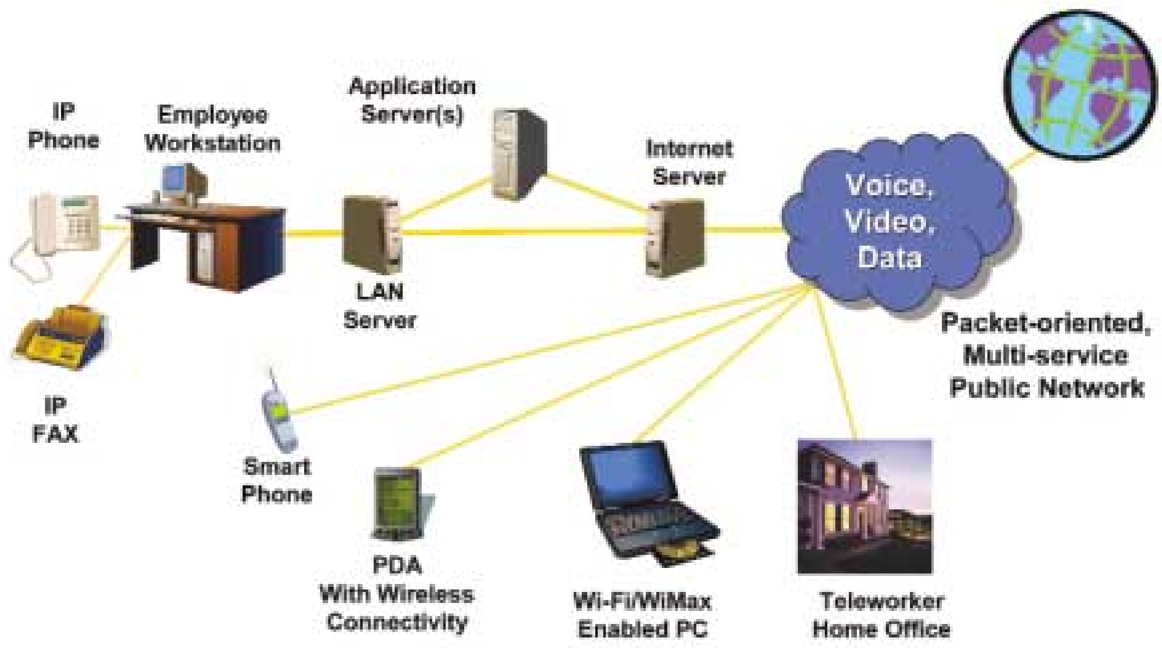
The Federal Communications Commission (FCC) under former Chairman Michael Powell took the stand that the regulation of VoIP would not be subject to traditional state public utility regulation. The FCC has set up a working group to identify policy issues addressing the migration of communications services to Internet-based platforms. The most hotly contested debate is over whether VoIP technology should be regulated in an approach similar to traditional telephone services, which leads to taxation issues regarding calls flowing between states. The FCC has taken the position that moving more communications to IP is in the public interest. Chairman Powell stated publicly, ". . . The policy environment must begin with the recognition that the Internet is inherently a global network that does not acknowledge narrow, artificial boundaries. I (absolutely) believe in maintaining an Internet free from government regulation and firmly support the idea that VoIP should evolve in a regulation-free zone."

In fact, in November 2004, the FCC effectively preempted the states from having jurisdiction on Internet telephony. The commission voted 4-1 to exempt Vonage Holdings Corp.'s DigitalVoice VoIP phone service from Minnesota telephone taxes and certification standards, including 911 emergency call capability. In taking Vonage's side in its dispute with Minnesota, the FCC set a standard it says is essential to free the VoIP industry from state laws that can hinder growth and prosperity. The decision adds to the regulatory certainty the Commission began building with orders adopted earlier this year regarding VoIP by making clear that this Commission, not the state commissions, has the responsibility and obligation to decide whether certain regulations apply to IP-enabled services. The Commission has the power to preempt state regulations that conflict with, or impede, federal authority over interstate communications.

the vision of VoIP

The "connected" office worker in the near future will be able to remain connected regardless of where they are located. Wi-Fi / Wi-Max enabled portable devices and VoIP enabled applications will extend the enterprise to them anywhere, any time. In moving to a converged voice, data, and video network, organizations will need to "future-proof" their infrastructure in order to support multi-media interactions with constituents and support the connected teleworker. (See Figure 15.)

Figure 15. *Technology Support for the "Connected" Teleworker*



Source: Tom Shepherd, ITE, IA

state CIO VoIP/IP telephony questions

Section I—General

1. **What is VoIP? Is it the same as IP telephony or is it different?**
   It is helpful to think about IP Telephony separately from IP transport for a number of financial and operational reasons. VoIP uses an IP transport layer between voice switches to complete phone calls instead of the traditional dedicated circuitry. VoIP is being used today by many of the Interexchange Carriers to transport calls throughout the nation, with the traditional PSTN being used to originate and terminate the calls. This makes the change to VoIP transparent to the end user as long as proper bandwidth and QoS are allocated to the voice calls. IP telephony is different in that it extends the IP network all the way to the desktop, bypassing the PSTN completely. (Also, see pages 6-9.)

2. **Is VoIP the right technology?**
   In the traditional voice deployment, dedicated facilities are required. In a VoIP deployment, where IP is the transport portion only, this is called a hybrid application. The transport is IP, which can save money on toll charges between intra- or inter-LATA locations, but the desktop handsets would be unaffected. In a pure IP deployment, or IP telephony deployment, there is no conversion to TDM at any point in the network. The decision as to which technology is the right one for any situation will depend on the existing infrastructure. In some instances, the existing cable plant is sufficient and replacing that with a pure IP solution would be too costly, since that may include upgrading existing routers to be QoS capable, as well as replacing existing handsets. If these items are nearing end of life anyway, it may be cost effective to examine an IP telephony deployment.

3. **Does VoIP support transformation of the business of serving the public?**
   Using the definitions provided in no.1 above, VoIP could improve serving the public as it could very well lower toll costs. The Applications associated with IP Telephony could certainly improve services to the public by increasing the accessibility of people at all levels of government, improving responsiveness, and facilitating collaboration between the public and government employees and improving the efficiency of public employees.

4. **How does all this VoIP activity relate to my relationship with the various Telco's in my state?**
   Telco's will see inter- and intra-state access charges decline as VoIP is implemented. Some of the larger telcos have opted to provide VoIP on their own network to retain the customer base. While a considerable amount of money can be saved by implementing a VoIP solution, this may not prove to be a financially sound model for every location, and there will still be a number of calls that will go to the PSTN or to an alternate VoIP provider for termination. Balancing the savings achievable with the on-going relationships will be a challenge if the Telco's see their revenue streams decreasing. The adjustment will be that more bandwidth is needed to do VoIP, so the overall loss of voice origination/termination may be limited somewhat by additional circuit or bandwidth requirements. The value proposition of VoIP for individual organizations is in the different trunking and transport options. (See section on Major IP Communications Solutions, page 9.)

5. **Are we better off waiting for the LECs to provide VoIP or should we build our own "phone system?" How do we make this decision?**
   Looking at this from the two perspectives of VoIP and IP telephony might help clarify this. The answer depends on the level of expertise available. If you currently manage your own phone system, and have IP knowledge within the organization, it probably makes sense to maintain an inhouse system, particularly with a VoIP deployment—meaning transport only. IP telephony may require a more significant overhaul of the system,

and a cost-benefit analysis will have to be done to determine the best path forward for each organization. There are different drivers for the implementation of VoIP Transport and IP Telephony. You will need to assess the value proposition of ownership vs. leasing or renting from the carrier.

6. **What is the regulatory climate surrounding VoIP? How may that eventually impact my operation?**
   The regulatory climate is beginning to clarify itself. One of the largest areas of disagreement is the area of access fees, and it is anticipated a revamping of the current system will be required in the near future to ensure universal coverage. The E911, Communications Assistance for Law Enforcement Act (CALEA), and disability access are areas currently being addressed by the industry, which are critical for state applications.

7. **What is the future of VoIP?**
   Call centers are a great use of VoIP; the technology is here to stay. More and more residential customers are moving to VoIP as a primary means of communication, though there is still less than one percent of homes that have gone to VoIP. The business community has embraced using VoIP more completely, as there are more dollars to be saved by connecting their largest centers together without having toll charges or having to maintain dual data and voice facilities. According to TEQConsult Group in Hackensack, N.J., new VoIP lines are on track to hit 40 percent of all U.S. lines installed in 2004 and should pass the 50 percent mark in 2005.

Section II—Reliability/Quality of Service/ Security

1. **What are the reliability expectations of government users?**
   Users have more communications options today, so expectations for any one technology are dependent upon the availability of other options. However, the expectation is that departments with public contact should maintain the highest level of telephone availability, where departments with contacts mainly within the system would have e-mail as a back-up means of communication. Obviously, if the LAN and voice networks are combined, an issue in one may very well indicate an issue in the other. Generally, VoIP will be expected to have the reliability people are historically used to.

2. **Can we create a solution that is reliable enough to approximate the current phone system?**
   Yes, a solution can be implemented that is very reliable. These solutions are vendor dependent, and as with any application, the analysis of reliability versus implementation and maintenance costs will have to be examined to ensure the best solution is selected. These are all part of the QoS equation and will need to be addressed. Generally speaking, the higher the reliability requirements, the higher the cost will be in supporting the VoIP service.

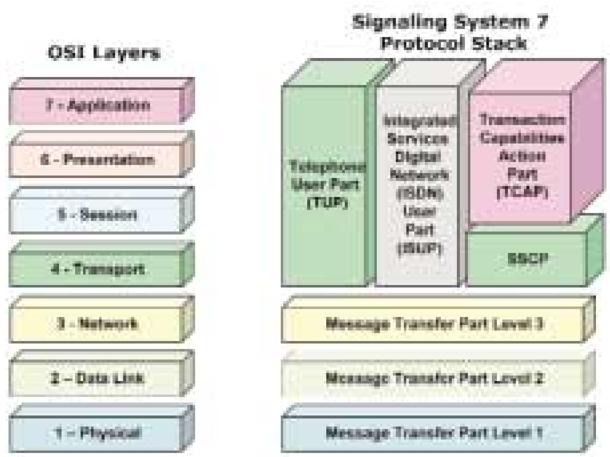
3. **Can you give me a checklist of what I need to do to facilitate a good QoS?**
   See VoIP Implementation and Planning Section, pages 24-28.

4. **What QoS protocols and policies are available in the IP network and how will we ensure QoS and traffic engineering?**
   The specific QoS protocols are dependent upon the vendor or service provider you choose.

   PSTN QoS is handled by the Message Transfer Parts (MTP) 2 and 3 in the SS7 Protocol Stack. MTP Level 2 ensures accurate end-to-end transmission of a message across a signaling link. Level 2 implements flow control, message sequence validation, and error checking. When an error occurs on a signaling link, the message (or set of messages) is retransmitted. MTP Level 2 is equivalent to the OSI Data Link Layer.

   MTP Level 3 provides message routing between signaling points in the SS7 network. MTP Level 3 re-routes traffic away from failed links and signaling points and controls traffic when congestion occurs. MTP Level 3 is equivalent to the OSI Network Layer. (See Figure 16.)

Figure 16. *OSI Layers—Signaling System 7 Protocol Stack*



Source: Intel Corporation

5. **How secure are IP service offerings?**
The level of security depends a great deal on the network design and vendor equipment selected. If the network is designed with limited connectivity to the public internet, those connections can be monitored to protect or minimize against a DoS (Denial of Service) attack or SPIT (SPam over Internet Telephony). Links to a couple of recent articles on this topic can be found at:

"Watchguard Brings Firebox Model to Small Nets"
http://www1.commsworld.com.au/NASApp/cs/ContentServer?pagename=commsworld/home&var_el=art&art_id=1084814734726&var_sect=NEWS&from=home

"Spam, DoS Headed VoIP's Way"
http://www.enterpriseitplanet.com/security/news/article.php/3398951

6. **Since VoIP necessitates "always on" Internet connectivity, what security implications are there?**
Any device with an IP address will have security implications. To mitigate that impact, VLANs, internal IP numbering, etc, should be considered. Managed networks would be the most secure to use instead of using the public Internet.

Section III—Cost Benefits/Funding

1. **What are the real (visible and hidden) costs associated with VoIP?**
   VoIP transport will require properly trained support staff and new routers, switches, power, increased bandwidth, and new servers. VoIP can be implemented into the transport layer of the network and not affect the end devices. IP Telephony would require the wholesale change out of handsets and switches as well as the addition of properly trained support staff. IP Telephony can be deployed slowly to the desktop, which would spread out the costs of deployment. Some users would have IP handsets, and some would have digital or analog handsets. The assumption is that the digital sets in use would be compatible with the IP PBX.

2. **What are the documented cost benefits of moving to VoIP and is there a true ROI model I can utilize to determine when and how to move to VoIP?**

   <u>VoIP Savings for Organizations</u>
   - Eliminate or reduce PSTN toll charges.
   - Reducing of eliminating service and support contracts on existing PBX hardware.
   - Eliminate the requirements for Centrex services and charges.
   - Collapsing voice and data infrastructures, support, and management resources into one that represents savings in staff, maintenance, upgrades and reduced costs associated with moves, adds and changes.
   - Reduce the on-going costs for the support and maintenance of separate voice messaging systems.
   - Provide productivity benefits for remote and traveling workers by extending the enterprise to them (e.g. providing them with the same integrated capabilities as office workers).
   - Reduce user training and improve the learning curve on phone and messaging systems.
   - Cost-effectively implement unified messaging. (Unified messaging is the integration of several different communications media, such that users will be able to retrieve and send voice, fax, and e-mail messages from a single interface. Unified messaging technology provides the power to reach people almost anywhere, at any time and the flexibility to allow people to control when they can be reached and enable them to interface with messages how and when they want. Subscribers reduce the number of places they must check for incoming voice, fax, and e-mail messages).
   - Improved security.
   - Reduced systems downtime and improve performance.

   <u>Benefits for Call Centers</u>
   - Enable the implementation of "Virtual" call centers, allowing more flexibility in the center's configuration. This can facilitate call center consolidation efforts and provide enterprise capabilities to telecommuting call center workers.
   - Improve customer support services and reduce abandoned calls and call times.
   - Improve customer satisfaction and reduce customer turnover via improved call center services.

   <u>Cost Considerations</u>
   - VoIP telecommunication hardware and software.
   - IP phone sets or soft phones.
   - Network upgrades to ensure quality of service and performance.
   - Implementation labor and professional services.
   - On-going support and administration labor.
   - Support and maintenance contracts.
   - Increased support calls and potential user downtime losses on initial deployment.
   - IT training.
   - User training.
   - Write-off, write-down and disposal costs for existing telecommunication assets.

   <u>Potential VoIP Project Risks</u>
   - Quality of service and network performance.
   - User training and adoption.
   - Administration and support skill levels and

resources.
- Proprietary vs. open systems interoperability.
- Potential regulatory issues affecting the costs associated with VoIP and IP Telephony.

In addition to direct benefits from VoIP, it is important to consider the costs associated with the limitations in the current PSTN. The inability to support certain features (or do so cost effectively) reduce the business value of the PSTN over newer, more robust technologies. For example:
- Growth in distributed staffing by many organizations require an adaptable system for call routing and toll avoidance. A traditional PBX cannot do this (or may do so in a very limited manner). VoIP can deliver.
- Automatic reconfiguration during staff moves. If you pick up an IP telephone and move it into another office, the phone number moves with the phone. The costs of staff moves for the legacy phone system can run up to $100 per person with a PBX, depending on the number and complexity of the moves.

Each traditional phone call uses a 64 Kbit/second circuit for the full duration of the call. During a VoIP call, the periods with sound take up that same bandwidth but periods of silence or a constant tone (assuming data compression) would take much less. Because of this, VoIP can carry more calls in the same bandwidth as a circuit based system.

Quality of Service (QoS) is required to ensure that your calls are not adversely affected by your available network bandwidth. If jitter (e.g. variations in the delay in the phone call) becomes excessive, people are going to be annoyed with the telephone system. If you have a dedicated VoIP network and your load is always less than your network capacity, you will never need QoS. You need QoS when you have to prioritize (or ration) the capacity to the benefit of some protocols and to the detriment of others. In other words, by turning on QoS capabilities in your network equipment, you prioritize the Voice messages to the detriment of Data messages. The data users won't notice they get delayed (unless your network is extremely inadequate) and the voice users will be satisfied with the service.

3. **Will anticipated cost savings be eaten up by trying to ensure greater redundancy and reliability on data lines?**
   Once you implement VoIP, you can reduce costs since you are only maintaining one network instead of two. If your data network is not already redundant and reliable, the QoS requirements, even though they may prove to be costly, will eventually result in better service for agencies at a lower cost. It depends upon your individual situation.

4. **How will the transition to VoIP impact the number and cost of Centrex lines?**
   Most of the LECs have tariff pricing based upon annual commitments of lines. Partial implementation of VoIP may leave remaining lines with higher costs. You will have to do the analysis.

5. **There is a significant installed base of expensive traditional voice equipment that has not reached end-of-life. How do we implement VoIP when there are limited funds for replacement?**
   Use the projected operational savings from a converged network to assist you in mapping the migration path to IP telephony.

6. **Are there sufficient funds to build the IP network to a required level of availability?**
   It depends upon the strength of your data network. What is the current reliability and availability of your current IP network and does it meet your requirements today? Remember, you are currently maintaining 2 networks. What are the economics when you transition the telephony costs to the IP network? In the short term, there will probably be additional costs—in the long term, you should see cost savings from a converged network. You should explore various financing options that spread the costs out over a reasonable period.

Section IV —Implementation/Management

1. **I have existing PBXs and voice switching equipment. How can I phase in VoIP and leverage my existing investments?**
   VoIP, at the transport layer only, can be implemented fairly easily while still maintaining the existing PBXs, handsets, etc. This merely allows the toll savings and the combination of data and voice traffic onto one or multiple circuits which should decrease costs compared to maintaining separate trunking for each. IP telephony through IP to the desktop might be a natural progression as the existing equipment is ready for replacement. Remember, many PBXs can be IP enabled.

2. **Do I have to buy new telephones?**
   A VoIP implementation (transport layer) would not affect handsets at all. If you wish to implement IP telephony, then new telephones would be required.

3. **Can our existing systems (PBXs, Centrex, etc.) and network co-exist with IP switching? Will network be scalable and adaptable to our present and future needs?**
   It depends on the engineering and capacity of your existing network. Routers must be QoS capable and of adequate capacity. The RFI or RFP process can help determine the best solution for the specific situation.

4. **Are there any future improvements that would cause me to hesitate with my implementation today?**
   The standards today for IP telephony are incomplete and still being determined in the industry, so most vendor solutions will be proprietary. This does not impact the transport layer, however, so VoIP can be deployed on the transport level to reap the benefits. The drivers for IP telephony are dependent upon your costs for implementation and the urgency of implementing IP Telephony-based applications.

5. **Will the VoIP strategy and infrastructure allow the VoIP to be rolled out on an opportunistic basis?**
   Simply put, yes. Each office may not have the level of voice traffic to make a VoIP investment worthwhile. Financially, it makes more sense for a remote office environment at this time and much less sense for a campus environment. A migration plan spanning any length of time may cause differences to the end user that cannot be avoided depending upon the current technology and the solution being implemented. It is important to understand these implications before starting the project so the users can be adequately informed of any changes.

6. **How do I manage the fact that ownership of the various routers, switches and components of my network are distributed among the agencies in my state?**
   If you are trying to go to IP telephony, it MUST be a standards-based implementation. All agencies involved must comply with those standards. This is a converged network and must be effectively managed for quality, reliability and security. Otherwise, a VoIP implementation is still an option to save on the circuit or toll costs, while not impacting the individual LANs/WANs of the individual agencies. It may be desirable to establish common resource or funding pools and standards-based procurement requirements for VoIP implementations.

7. **Does VoIP necessitate centralized management of voice infrastructures and strategies? Consistent voice network architecture? Converged voice and data networks in the WANs/MANs?**
   VoIP doesn't require centralized management. As long as the routers can pass the traffic, the transport layer isn't a difficult conversion. However, in terms of IP telephony, there would have to be some attention to being consistent across the network. A converged network isn't necessarily required, but to maintain two separate networks is costly in terms of duplicate routers, cabling, and all environmentals associated with that extra equipment.

8. **Can a VoIP solution be effective if we don't operate the whole network, including the LANs?**
   As in question no. 6 above, a VoIP solution can

be implemented, but an IP telephony solution would be complex. To do IP telephony, all LANs would have to be managed identically, any software upgrades would have to be coordinated, and trouble-shooting rules would have to be in place to be able to adequately address any issues. Without consistent management, strategies, and service level agreements (SLAs) of converged network stakeholders, VoIP will have a hard time approaching the five-nines reliability of the current circuit-switched phone systems.

9. **What organizational changes must be made to effectively implement IP telephony using a converged network?**
   In order to be responsive to the requirements of IP based phone services, a role and response matrix must be established for LAN technicians. You should also provide additional training and designate technicians with a primary responsibility for IP telephony.

Section V—Emergency Services/Disaster Planning

1. **What are the implications of VoIP in the area of enhanced 911 (E911) coverage?**
   Some VoIP systems (no firm estimates on the number) will be able to offer E911 services by the third quarter of 2005. E911 service refers to the ability to route calls directly to an emergency dispatcher with the caller's location and phone number appearing automatically.

   VoIP 911 calls are currently routed through public safety access points (PSAPs)—facilities where an operator can alert emergency response agencies. These PSAPs don't have Data Management System/Automatic Location Identification (DMS/ALI) systems, which allow operators to immediately identify the caller's location. Since VoIP carriers aren't regulated, they don't have access to E911 selective routing. So, for the foreseeable future, each provider must come up with a solution and pass the cost on to consumers.

   Some VoIP providers and cable operators offer E911 as a subscription service. One VoIP provider with 1,700 subscribers charges a $9.95 setup fee plus $3 a month. Even with this E911 service, the VoIP provider urges consumers to seek emergency services elsewhere. User agreements from VoIP providers 8x8 Inc., Verizon Communications Inc., and others advise consumers to "maintain an alternative means of accessing traditional 911 services." VoIP users can't dial 911 during a power, Internet, or other network outage.

   A selling point of VoIP services is that it's portable. Once you move to a new location and reconnect, any location information registered with your VoIP service provider is invalid and 911 won't work correctly. You have to register your new geographic location, which currently takes days to be processed.

   VoIP service providers are working to develop a smarter E911 system that would identify a caller's location and route calls to the nearest PSAP regardless of the caller's pre-registered address. This functionality will probably not be generally available until 2006 or after.

2. **What network disaster plans should be in place?**
   PSTN Reliability is typically associated with an ability to get a dial tone, which is much different than actually being able to complete a call with an acceptable QoS. Plain old telephone service (POTS) subscribers in densely populated areas with very old local wiring loops serving the apartments, houses and businesses can all recount stories of phone outages or degraded call quality on rainy days. The same holds true for rural customers served by aging infrastructure. In contrast to their circuit-switched counterparts, packet-switched networks fundamentally enable fault tolerance, adaptive routing, and disaster recovery. It is quite possible—and can be quite cost-effective—to build IP-based voice systems that are more reliable than circuit-switched PBX platforms. The key is to start with the right foundation. Today's VoIP solutions fall into three basic categories:
   1. Systems evolved from traditional PBX platforms;
   2. Systems evolved from traditional data-switch platforms; and
   3. Systems designed from the ground up for VoIP.

   All three of these architectures can be used to deliver VoIP systems with five-nines reliability, but they involve different degrees of complexity and cost.

   Legacy PBXs and key systems are hierarchical voice silos that operate independently at each location in a multi-site company. They cannot back each other up or be managed as a single voice network, and create a single point of failure at each site. IP networks are inherently distributed and resilient. VoIP architects starting with a blank slate can exploit this fundamental strength to create a self-healing voice platform. A truly unified voice system can be distributed across multiple sites by using a simple peer-to-peer architecture that has no single point of failure. Your VoIP/IP Telephony implementation must take advantage of these design features.

Additionally, there are product and service offerings from vendors, such as Cisco's Survivable Remote Site Telephony (SRST) feature. The Cisco IP Communications Solution utilizes Cisco CallManager in combination with SRST in to enable organizations to extend high-availability IP telephony to their small branch offices.

http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_data_sheet09186a00800888ac.html

3. Will IP equipment suppliers support us in a disaster?
This is dependent upon your relationship and contract with the provider(s).

Section VI—Technical

1. **I have a large number of devices on my network. At any given time, I have some isolated power outages. What can I do in this scenario to avoid any voice outages? What about power failures?**

   In regard to using VoIP to transport voice packets, the issues are the same as they are with your current network reliability. The complexity increases dramatically when it gets to IP telephony. Redundant power availability should be built into any solution if up-times are to rival the current standard 99.999 percent PSTN availability. This will also impact the environmentals (e.g. HVAC) of the equipment location.

   The primary reason for the switched phone network's superior reliability lies in the reliability and redundancy of network components, both in hardware and software. Phone network switches are designed for reliability. Because their hardware components represent only a small fraction of their price, and physical size is not a dominant consideration, switch hardware engineers can afford to be conservative. But the real story regarding hardware reliability is redundancy—just about anything that could cause significant outages upon failure is backed up. Critical processors typically are in "hot standby," meaning that if one fails, a second one immediately assumes all functions without dropping a single call. And power systems are backed up with batteries and generators so that the phones keep working indefinitely in the event of a widespread commercial power outage.

   It has been reported that for every hour spent writing switch software code, almost 200 hours is spent testing it. When it comes to the phone network, it's not good enough to find and fix software bugs after they cause an outage.

   VoIP is architecturally sound and cost-effective, but the servers and routers that provide connections to individual users currently are less reliable than telephone switches in both hardware and software design. Until that gap is reduced, VoIP will not achieve the same levels of overall reliability as consumers now expect from the circuit-switched telephone network. This will take considerable attention to engineering your VoIP and IP telephony solution.

2. **How do we ensure 99.999 percent uptime for voice?**
   For IP telephony, each vendor will have a solution based on the design of the network and the individual customer's needs. The important items to keep in mind are uninterruptible power and security. Additionally, most vendors encourage a connection to the PSTN at each location to ensure call completion. Good design practice will allow you to achieve 99.999 percent availability. Reliable LAN and WAN infrastructures supporting power protected clustered enterprise-class servers currently deliver five nines availability. Reliability can also be extended to remote sites by deploying features such as Cisco's Survivable Remote Site Telephony, ensuring telephony services remain functional even in the event of a WAN failure.

3. **What are the interoperability issues around VoIP? What components do we need to concern ourselves with regarding interoperating with multiple vendor equipment?**
   There are uniform industry standards at all levels, IP telephony interoperability can be provided by the vendors themselves. Most do testing with competitor's products to determine where interoperability is an issue and where it isn't. On the VoIP side, where a transport solution only requires the routers to pass IP traffic, standard procedures would apply here.

4. **How does the network design handle congestion?**
   This really does depend on how the network is designed. Most vendors encourage maintaining connectivity to the PSTN so that when calls are blocked on the IP side, they can use the PSTN as a backup route.

5. **Are network devices compatible or upgradeable to support VoIP?**
   This really depends on the devices that are

currently deployed. Some devices will be—some won't be. Some devices may already be VoIP enabled.

6. **How are lost packets handled?**
   Proper LAN, WAN, and Gateway design and enabling QoS on the voice trunks will be needed to address these issues up front. VoIP packets are not retransmitted.

7. **How will we handle loss of network synchronization?**
   In a VoIP solution, the network synchronization would not change as the existing data routers would take over this function. In terms of an IP telephony deployment, the handsets will be timed off of the switch. Each vendor should provide information on how the equipment reacts to loss of network synchronization and what is required to restore.

8. **Can we accurately predict traffic volumes?**
   This information should be available either in-house or through your current provider. The anticipated call volume, minutes of use, and number of seats using any given trunk would be incorporated into sizing that trunk appropriately.

9. **How well do carriers understand IP switching?**
   Most of the Inter-exchange Carriers and major Incumbent Local Exchange Carriers have been using IP switching on some level for several years. The smaller the carrier, the less likely they are to have a deployed IP network, though there are certainly exceptions.

10. **What are the technical support considerations that we need to be aware of?**
    For a VoIP solution, the issues will be the same as that of regular IP traffic. You will want to ensure QoS is available on the routers.

Section VII—Applications

1. **What applications will be supported for VoIP and can you give me specific case studies?**
   See page 15 for a list of IP Telephony applications. See the following page for case studies.

links to articles and case studies

"Localized Mobility"—March 2005, *Government Technology Magazine*
http://www.govtech.net/magazine/channel_story.php?channel=19&id=93338

"Nortel Leads the Way in VoIP Applications That Support Teleworkers"—December 2003, *Network World Fusion*
http://www.nwfusion.com/reviews/2003/1208rev.html

"IP Telephony Applications"—Nortel Networks
http://builder.itpapers.com/abstract.aspx?scid=6&dtid=2&docid=81001

Web cast—"IP Voice from the Inside: The CNET Case Study"—Shoreline Communications
http://builder.itpapers.com/abstract.aspx?scid=6&dtid=2&docid=37375

Cisco Customer Video Case studies are located at:
http://newsroom.cisco.com

Additional Cisco case studies can be found at the following links:
http://www.cisco.com/en/US/strategy/government/all_local.html
http://www.cisco.com/go/ipc

VoIP/IP telephony glossary of terms

NUMBERS

802.3af	An IEEE standard for powering network devices via Ethernet cable. Also known as "Power-over-Ethernet," it provides 48 volts over 4-wire or 8-wire twisted pair. The 8-wire cable uses one twisted pair for the power, while the 4-wire transmits the power over the same pair as the data, but uses different frequencies. Designed with IP phones and wireless access points in mind, it allows devices such as these to be placed in locations that have no electrical outlets. Only the Ethernet cable needs to be connected to the device.
802.11 a/b/g	802.11 refers to a family of specifications developed by the IEEE for wireless LAN technology. 802.11 specifies an over-the-air interface between a wireless client and a base station or between two wireless clients. The IEEE accepted the specification in 1997. <u>There are several specifications in the 802.11 family:</u> 802.11 - applies to wireless LANs and provides 1 or 2 Mbps transmission in the 2.4 GHz band using either frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (DSSS). 802.11a - an extension to 802.11 that applies to wireless LANs and provides up to 54 Mbps in the 5GHz band. 802.11a uses an orthogonal frequency division multiplexing encoding scheme rather than FHSS or DSSS. 802.11b (also referred to as 802.11 High Rate or Wi-Fi) - an extension to 802.11 that applies to wireless LANS and provides 11 Mbps transmission (with a fallback to 5.5, 2 and 1 Mbps) in the 2.4 GHz band. 802.11b uses only DSSS. 802.11b was a 1999 ratification to the original 802.11 standard, allowing wireless functionality comparable to Ethernet. 802.11g - applies to wireless LANs and provides 20+ Mbps in the 2.4 GHz band.

A

ACD	See automatic call distributor.

ALI	Automatic Location Identification provides for an address display of the subscriber calling 911. With ALI, the PSAP receives the ANI display and an ALI display on a screen. The ALI display includes the subscriber's address, community, state, type of service and if a business, the name of the business. The PSAP will also get a display of the associated ESN information (police, fire and rescue).
ANI	Automatic Number Identification. A PSTN system that transmits the billing number of the calling party for accounting and billing purposes.
ARP	Address Resolution Protocol. Internet protocol used to map an IP address to a MAC address. Defined in RFC 826. Allows host computers and routers to determine the data link layer address corresponding to the IP address in a packet routed through the LAN. Although the packet is addressed to an IP address, the LAN hardware responds only to data link layer addresses. The host or router with the destination IP address replies with its own data link layer address in an ARP response, which the forwarding host or router will use to construct a data link layer frame. The result is stored in cache memory so subsequent packets addressed to the same destination can be routed without an explicit ARP process.
Asynchronous Transfer Mode (ATM)	A 53-byte cell switching technology well suited for carrying voice, data, and video traffic on the same infrastructure. It is inherently scalable in throughput and was designed to provide Quality of Service (QoS).
Auto Discovery	The process by which a network device automatically searches through a range of network addresses and discovers the known types of devices that are present.
Automatic Call Distributor (ACD)	A specialized phone system that handles incoming calls or makes outgoing calls. An ACD can recognize and answer an incoming call, look in its database for instructions on what to do with that call, play a recorded message for the caller (based on instructions from the database), and send the caller to a live operator as soon as the operator is free or as soon as the caller has listened to the recorded message.

B

B-ISDN	Broadband Integrated Services Digital Network. A network that employs switching techniques independent of transmission speeds, and that allows a network to expand its capacity without major equipment overhauls. B-ISDNs support gigabit speed circuits in the public network and high speed switching of all traffic types in public and private networks. B-ISDNs also provide bandwidth-on-demand capabilities. Contrast with N-ISDN. See also BRI, ISDN, and PRI.
BOOTP	Bootstrap Protocol. A TCP/IP protocol that enables a network device to discover certain startup information, such as its IP address.
Broadband	High-speed voice, data, and video networked services that are digital, interactive, and packet-based. The bandwidth is 384 Kbps or higher, and 384 Kbps is widely accepted as the minimum bandwidth required to enable full-frame-rate digital video.

C

Call Agents	Intelligent entity in an IP telephony network that handles call control in an MGCP model voice over IP network. Also known as a Media Gateway Controller (MGC).
CAS	1. Centralized Attendant Service. One group of switchboard operators answers all incoming calls for several telephone systems located throughout one city or region. 2. Channel Associated Signaling. In-band signaling used to provide emergency signaling information along with a wireless 911 call to the Public Safety Answering Point (PSAP).
Category 3/5/5e/6 Wiring	The type of twisted pair copper cabling used in phone and data networks to connect a device to a jack, and the cabling used in the ceilings/walls to connect the jack to the local equipment closet. Different category cable supports different network connection speeds - generally, the higher the value, the faster the service. A network connection is only as fast as its slowest link. Category 3 wiring can carry a maximum rate of 10 Mbps (megabits per second), category 5 can support Fast Ethernet (up to 100 Mbps), category 5e and 6 have the potential to carry Gigabit (1000 Mbps) service. Category 6

	wiring is currently the standard for new data jack installations except in a limited number of locations on campus that currently cannot support it. Outlet color indicates wiring: beige is category 3 wiring, black is category 5 wiring, blue is category 6 wiring.
Central Office	See CO.
Centralized Call Processing	Refers to a processing construct where all call processing is performed at a central site, or hub, and no call processing is performed at branch sites.
CLEC	See Competitive Local Exchange Carrier.
CO	Central Office. Local telephone company office to which all local loops in a given area connect and in which circuit switching of subscriber lines occurs. Central office can also refer to a single telephone switch, or what is known as a "public exchange" in Europe.
Codec	Coder-decoder. 1. A device that typically uses pulse code modulation to transform analog signals into a digital bit stream, and digital signals back to analog. 2. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a software algorithm used to compress/decompress speech or audio signals.
Common Channel Signaling (CCS)	A communications system in which one channel is used for signaling and different channels are used for voice/data transmission. Signaling System 7 (SS7) is a CCS system, also known as CCS7.
Competitive Local Exchange Carrier (CLEC)	Created by the Telecommunications Act of 1996, a CLEC is a service provider that is in direct competition with an incumbent service provider. CLEC is often used as a general term for any competitor, but the term actually has legal implications. To become a CLEC, a service provider must be granted "CLEC status" by a state's Public Utilities Commission. In exchange for the time and money spent to gain CLEC status, the CLEC is entitled to co-locate its equipment in the incumbent's central office, which saves the CLEC considerable expense.

D

Dark Fiber	Refers to unused fiber-optic cable. The dark strands can be leased or sold to individuals or other companies who want to establish optical connections among their own locations.
DHCP	Dynamic Host Configuration Protocol. A TCP/IP protocol that enables PCs and workstations to get temporary or permanent IP addresses out of a pool from centrally-administered servers. Like its predecessor, BOOTP, DHCP provides a mechanism for allocating IP addresses manually, automatically and dynamically, so that addresses can be reused when hosts no longer need them. For Cisco CallManager, a DHC server is queried by a telephone or gateway device upon booting to determine network configuration information. The DHCP server provides the device with an IP address, subnet mask, default gateway, DNS server, and a TFTP server name or address. With Cisco IP Phones, DHCP is enabled by default. If disabled, you must manually enter the IP address and other specifications manually on each phone locally.
DID	Direct Inward Dialing. A method of directly dialing the directory number of a Cisco IP Phone or a telephone attached to a PBX without routing calls through an attendant or an automated attendant console, such as Cisco Web attendant. Compare to DOD.
Digital Signal Processor (DSP)	A specialized digital microprocessor that performs calculations on digitized signals that were originally analog, and then forwards the results. The big advantage of DSPs lies in their programmability. DSPs can be used to compress voice signals to as little as 4,800 bps. DSPs are an integral part of all voice processing systems and fax machines.
Direct Inward Dialing	See DID.
Direct Outward Dialing (DOD)	Direct Outward Dialing. The ability to dial directly from Cisco CallManager or PBX extension without routing calls through an operator, attendant or automated attendant functions. Compare to DID.

Dual-Tone Multi-frequency (DTMF)	A way of signaling consisting of a push-button or touch tone dial that sends out a sound consisting of two discrete tones that are picked up and interpreted by telephone switches (either PBXs or central offices).
Dynamic Host Configuration Protocol	See DHCP.

E

ESN	Emergency Service Number. Assigned to the subscriber's telephone number in the tandem office translations. The ESN represents a seven digit number by which the tandem office routes the call to the proper PSAP. PSAPs with ALI capabilities also receive a display of the ESN information that shows which police, fire and rescue agency serves the telephone number calling 911. An ESN is a unique combination of police, fire, and rescue service for purposes of routing the E911 call.

F

Finger	A program that goes to a computer running the finger daemon (service) and returns information about a particular user, if available. Part of the information displayed is the .plan and .project files. Some people update these files often, allowing others to find information about them easily. Originally the finger client was a UNIX program, but now versions are available for other operating systems.
Frame Relay	ITU-T-defined access standard. Frame Relay services, as delivered by the telecommunications carriers, employ a form of packet switching analogous to a streamlined version of X.25 networks. Packets are in the form of frames that are variable in length with the payload being anywhere between zero and 4,096 octets. Frame Relay networks are able to accommodate data packets of various sizes associated with virtually any native data protocol.

G

Gatekeeper	A component of the ITU H.323 "umbrella" of standards defining real-time multimedia communications and conferencing for packet-based networks. The gatekeeper is the central control entity that performs management functions in a Voice and Fax over IP network and for multimedia applications such as video conferencing. Gatekeepers provide intelligence for the network, including address resolution, authorization, and authentication services, the logging of call detail records, and communications with network management systems. Gatekeepers also monitor the network for engineering purposes as well as for real-time network management and load balancing, control bandwidth, and provide interfaces to existing legacy systems.
Gateway	The point at which a circuit-switched call is encoded and repackaged into IP packets. A gateway is an optional element in an H.323 conference and bridge H.323 conferences to other networks, communications protocols, and multimedia formats.
Gateway Controller	Coordinates setup, handing and termination of media flows at the media gateway.
GigE	Gigabit Ethernet, a transmission technology based on the Ethernet frame format and protocol used in local area networks (LANs), provides a data rate of 1 billion bits per second (one gigabit). Gigabit Ethernet is defined in the IEEE 802.3 standard and is currently being used as the backbone in many enterprise networks.

H

H.323	ITU-T standard that describes packet-based video, audio, and data conferencing. Allows dissimilar communication devices to communicate with each other using a standardized communications protocol. H.323 is an umbrella standard that describes the architecture of the conferencing system, and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol. For example, the Cisco IOS integrated router gateways use H.323 to communicate with Cisco CallManager. See also gateway.

H.323 Terminal	Endpoint on a LAN. Supports real-time, 2-way communications with another H.323 entity. Must support voice (audio codecs) and signaling (Q.931, H.245, and RAS). Optionally supports video and data e.g., PC phone, videophone, Ethernet phone.
H.GCP	Breaks up a gateway into a decomposed architecture whereby the media gateway (MG), and the media gateway controller (MGC) are treated as separate components.
Hot Desking	Using a set of cubicles for mobile workers who come into the office from time to time. It is similar to hoteling, but reservations are not required. People come in and sit down at the next available seat, plug into the network and go to work, which means a vice president might sit next to a junior trainee at any given time.
Hoteling	Using office space on an as-needed basis like a hotel room. Telecommuters reserve office space ahead of time for trips to the office.

I

ICM	Intelligent Contact Manager. The Cisco software system that implements enterprise-wide intelligent distribution of multi-channel contacts (for example, inbound and outbound telephone calls, e-mail messages, Web collaboration requests, chat requests) across contact centers. Cisco ICM software is an open-standards-based solution that provides contact-by-contact pre-routing, post-routing, and performance-monitoring capabilities.
IEEE	Institute of Electrical and Electronics Engineers. Professional organization whose activities include the development of communications and network standards. IEEE LAN standards are the predominant LAN standards today.
Incumbent Local Exchange Carrier (ILEC)	Typically the carrier that was granted the right to provide service as a result of the breakup of AT&T. These providers are also referred to as RBOCs (Regional Bell Operating Companies) or Baby Bells.

Interactive Voice Response (IVR)	Links callers with information in databases. This technology allows callers to complete transactions or queries over the phone. Automatic Speech Recognition (ASR) is fast replacing the DTMF method of activating IVR services and is one of the most important recent innovations in telephony-based self-service.
International Telecommunications Union (ITU)	An organization established by the United Nations to set telecommunications standards, allocate frequencies to various uses, and sponsor trade shows every four years.
Internet Engineering Task Force (IETF)	One of two technical working bodies in the Internet Activities Board. It meets three times a year to set the technical standards for the Internet.
Internet Protocol (IP)	Internet Protocol. Messaging protocol that addresses and sends packets across the network in the TCP/IP stack, offering a connectionless internet work service. To communicate using IP, network devices must have an IP address, subnet, and gateway assigned to them. IP provides features for addressing, type-of-service specification, fragmentation and reassembly, and security. Standardized in RFC 791.
IP	See Internet Protocol.
IP Contact Centers	A contact center that does not use circuit switching. All calls are IP or converted from PSTN to IP. Also called: IPCC, IP-based Call Centers, Converged IP Contact Centers, and IP-based Contact Centers.
IP Telephony	Technology that allows voice phone calls to be made over the Internet or other packet networks using a PC via gateways and standard telephones.
Interworking Function (IWF)	Provides the means for two different technologies to interoperate.

ISDN	Integrated Services Digital Network. Communication protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other source traffic.
IVR	See Interactive Voice Response.

J

Jitter	A type of distortion caused by the variation of a signal from its reference that can cause data transmission errors, particularly at high speeds.

L

LAN	Local-area network. High-speed, low-error data network covering a relatively small geographic area (up to a few thousand meters). LANs connect workstations, peripherals, terminals, and other devices in a single building or other geographically limited area. LAN standards specify cabling and signaling at the physical and data link layers of the OSI model. Ethernet, FDDI, and Token Ring are widely used LAN technologies. Compare with MAN, VLAN and WAN.
LATA	Local access and transport area.
Layer 2 Network Technologies	At this layer, data packets are encoded and decoded into bits. It furnishes transmission protocol knowledge and management and handles errors in the physical layer, flow control and frame synchronization. The data link layer is divided into two sublayers: The Media Access Control (MAC) layer and the Logical Link Control (LLC) layer. The MAC sublayer controls how a computer on the network gains access to the data and permission to transmit it. The LLC layer controls frame synchronization, flow control and error checking.
Layer 3 Network Technologies	This layer provides switching and routing technologies, creating logical paths, known as virtual circuits, for transmitting data from node to node. Routing and forwarding are functions of this layer, as well as addressing, internetworking, error handling, congestion control and packet sequencing.

LEC	See Local Exchange Carrier.
Local Exchange Carrier (LEC)	A company that provides local telephone service.
Local Loop	1. The communication line between a telephone subscriber and the local exchange carrier (LEC) switching center. 2. A local connection between an end user and a central office (CO) or end office (EO).

M

Media Access Control (MAC)	Lower of the two sub layers of the data link layer defined by the IEEE. The MAC sub layer handles access to shared media.
MCU	Multipoint Control Unit. The combination of a multipoint controller and a multipoint processor.
Media Gateway	A generic class of products grouped under the Media Gateway Control Protocol (MGCP). A major function of the media gateway is simple IP/TDM conversion under the control of a soft-switch. Media gateways include, but are not limited to, the following types of equipment: standalone, server-based gateways, RAS-based gateways, gateway switches, traditional CO switches, and ATM switches.
Media Gateway Controllers	Also known as Call Agents. See Call Agents.
Media Termination Point (MTP)	Media Termination Point. A virtual device that allows transfer, forward, conference, and hold features on any G.711 μ-law call between an IP Phone and any H.323 gateway, gatekeeper, or client. A call using MTP will automatically convert A-law to μ-law (and vice versa), if required. As a Cisco software application, MTP installs on a server during the software installation process.

MEGACO	A signaling protocol that enables switching of voice, fax and multimedia calls between the PSTN and next-generation IP networks, allowing dramatic growth and scalability of enhanced services. An updated and enhanced version of MGCP (Media Gateway Control Protocol.)
Metro-Ethernet	Ethernet access and services across a MAN (metropolitan area network). Also called: Metropolitan Ethernet and MEN.
Media Gateway Control Protocol (MGCP)	Enables external control and management of data communications equipment operating at the edge of multi-service packet networks (known as media gateways) by software programs, which are known as "call agents" or "media gateway controllers."
Multi-Protocol Label Switching (MPLS)	MPLS is a scheme typically used to enhance an IP network. Routers on the incoming edge of the MPLS network add an 'MPLS label' to the top of each packet. This label is based on some criteria (e.g. destination IP address) and is then used to steer it through the subsequent routers. The routers on the outgoing edge strip it off before final delivery of the original packet. MPLS can be used for various benefits such as multiple types of traffic coexisting on the same network, ease of traffic management, faster restoration after a failure, and, potentially, higher performance.
Multipoint Control Unit	See MCU.
Multi-Service Network	Same as multiuse network. A network that handles data, voice, and other capabilities simultaneously.

N

Network Appliance	A typically inexpensive personal computer, sometimes called a thin client, that enables Internet access and some business-related activities but lacks many features of a fully equipped PC, such as a hard drive or CD-ROM. Applications used on network appliances typically are housed on a Web server accessed by the appliance. Network appliances are used to ease remote management and cut costs.

O

OSI	Open Systems Interconnection. The only internationally accepted framework of standards for communication between different systems made by different vendors. Developed by the International Organization for Standardization, OSI is a model, not an active protocol. OSI organizes the communication process into seven different categories and places these in a layered sequence based on their relation to the user. The seven layers are: physical, data link, network, transport, session, presentation and applications.

P

Packet	Logical grouping of information that includes a header containing control information and (usually) user data. Packets are most often used to refer to network layer units of data. The terms datagram, frame, message, and segment are also used to describe logical information groupings at various layers of the OSI reference model and in various technology circles.
Packet Network	Network that transmits data in packet-mode (data are broken up into packets to be routed to their destination) as opposed to circuit-mode.
Private Branch Exchange (PBX)	Digital or analog telephone switchboard located on the subscriber premises, typically with an attendant console, and used to connect private and public telephone networks. A PBX is a small, privately owned version of the phone company's larger central switching office. It is connected to one or more central offices by trunks, and provides service to a number of individual phones, such as in a hotel, business, or government office. On a PBX, an outside line is normally accessed by dialing an access digit, such as 9.
Priority Queuing	A priority queue is an abstract data type supporting the following two operations: add an element to the queue with an associated priority; remove the element from the queue that has the highest priority, and return it.
Protocol	A set of rules or conventions that govern the format and relative timing of data in a communications network. There are three basic types of protocols: character-oriented, byte-oriented, and bit-oriented. The protocols for data communications cover such things as framing, error handling, transparency, and line control. Ethernet is an example of a LAN protocol.

Proxy	A device that relays network connections for other devices that usually lack their own network access.
PSAP	Public Safety Answering Point, usually the police, fire and/or rescue groups as determined by the local municipalities. A "ring-in" will not have ANI or ALI capabilities, but just receives calls or transferred calls from another PSAP.
Public Switched Telephone Network (PSTN)	General term referring to the variety of telephone networks and services in place worldwide.

Q

QoS	Quality of Service. Measure of performance for a transmission system that reflects its transmission quality and service availability.

R

RAID	Redundant Arrays of Independent Disks. A disk system with RAID capability can protect its data and provide on-line, immediate access to its data, despite a single (some RAID storage systems can withstand two concurrent disk failures) disk failure.
Real-Time Transport Protocol (RTP)	RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services.
Rich Media Conferencing	Rich Media Conferencing is a term that is used to describe the integration of various forms of media (such as audio, video and data) into one single 'multimedia' conferencing environment. Previously we used these communication applications in stand-alone environments, e.g. the phone for audio communication, the video for visual communication, and a web conferencing service for simultaneous collaboration.

Router	1. An interface device between two networks that selects the best route even if there are several networks between the originating network and the destination. 2. A device that provides network management capabilities (e.g., load balancing, network partitioning, usage statistics, communications priority and troubleshooting tools) that allow network managers to detect and correct problems. 3. An intelligent device that forwards data packets from one local area network (LAN) to another and that selects the most expedient route based on traffic load, line speeds, costs, or network failures.
RTP	See Real-time Transport Protocol.

S

Selective Routing	The capability to route a call to the particular PSAP serving the address associated with the telephone number making the 911 call. Selective routing is achieved by building TN/ESN translations in the tandem central office. These translations are driven by the E911 data base which assigns the ESN to each telephone number based on the customer's address. Service order activity keeps the E911 data base updated. The E911 data base, in turn, generates recent change to the tandem office to update the telephone number/ESN translations in the tandem data base.
Session Initiation Protocol (SIP)	A signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. The protocol initiates call setup, routing, authentication and other features to endpoints within an IP domain.
Signaling Control Point (SCP)	An SCP is usually a computer used as a front end to a database system. It is an interface to telco databases, not usually to other, application-specific databases. Telco databases are usually linked to SCPs by X.25 links. The SCP can provide protocol conversion from X.25 to SS7, or can provide direct access to the database through the use of primitives which support access from one level of protocol to another.
Signaling Gateway	SS7-IP interface - coordinates the SS7 view of IP elements and IP view of SS7 elements.

Signaling System 7	See SS7.
SIGTRAN	SIGTRAN (for Signaling Transport) is the standard telephony protocol used to transport Signaling System 7 (SS7) signals over the Internet. SS7 signals consist of special commands for handling a telephone call.
SIP	See Session Initiation Protocol
Skinny Station Protocol	See SSP.
SMDI	Simplified Message Desk Interface. Analog data line from the central office containing information and instructions to your on-premises voice mail box. A required interface for voice mail systems used with Cisco CallManager. SMDI was designed to enable voice mail integration services to multiple clients. However, to use SMDI, the voice mail system must meet several qualifications, including providing database support for two PBX systems simultaneously and IP network connectivity to the voice messaging system while maintaining the existing link to the PBX. SMDI-compliant voice mail systems must be accessible with a null-modem RS-232 cable and available serial port.
Soft-keys	On a Cisco IP Phone, buttons that activates features described by a text message. The text message is displayed directly above the soft key button on the LCD screen.
SoftPhone	Application that enables you to use a desktop PC to place and receive software telephone calls and to control an IP telephone. Also allows for audio, video, and desktop collaboration with multiple parties on a call. Cisco IP SoftPhone can be used as a standalone application or as a computer telephony integration (CTI) control device for a physical Cisco IP phone. All features are functional in both modes of operation.
Soft-tokens	(1) In programming languages, a single element of a programming language. For example, a token could be a keyword, an operator, or a punctuation mark. (2) In networking, a token is a special series of bits that travels around a token-ring network. As the token circulates, computers attached to the network can capture it. The token acts like a ticket, enabling its owner to send

	a message across the network. There is only one token for each network, so there is no possibility that two computers will attempt to transmit messages at the same time.
SS7	Signaling System 7. A telephone signaling system with three basic functions: supervising (monitoring the status of a line or circuit to see if it is busy, idle, or requesting service); alerting (indicating the arrival of an incoming call); addressing (transmission of routing and destination signals over the network).
SSP	Skinny Station Protocol. A Cisco protocol using low bandwidth messages that communicate between IP devices and the Cisco CallManager.
Switch	Network device that filters, forwards, and floods pieces of a message (packets) based on the destination address of each frame. Switches operate at the data link layer of the OSI model. See also OSI.

T

T.120	An ITU-T standard (International Telecommunications Union) for document conferencing. Document conferencing allows two or more people to concurrently view and edit a document across a network.
T.38	Defines procedures for real-time Group 3 facsimile over IP network.
T1	Trunk Level 1. A high-speed (1.544 megabits per second) digital telephone line with the equivalent of 24 individual 64Kbps channels, which are joined via time division multiplexing. A T-1 can be used to transmit voice or data, and many are used to provide connections to the Internet. Also known as a DS1 or Digital Signal 1.
TCP/IP	Transmission Control Protocol/Internet Protocol. The two best-known internet protocols, often erroneously thought of as one protocol. The transmission control protocol (TCP), which corresponds to Layer 4 (the transport layer) of the open systems interconnection (OSI) reference model, provides reliable

	transmission of data. The internet protocol (IP) corresponds to Layer 3 (the network layer) of the OSI model and provides connectionless datagram service. TCP/IP was developed by the U.S. Department of Defense in the 1970s to support the construction of worldwide internetworks.
TDM	Time Division Multiplexing. A technique for transmitting a number of separate data, voice, and video signals simultaneously over one communications medium by quickly interleaving a piece of each signal one after the other.
Telephony	Science of converting sound to electrical signals and transmitting it between widely removed points.
Trivial File Transfer Protocol (TFTP)	A simplified version of the FTP, TFTP is an application that transfers device configuration files (.cnf files) to devices from a TFTP server.
Toll bypass	A toll-free telephony call in which the relative locations of the two ends of the connection would cause toll charges to be applied if the call was made over the PSTN.
Trunk	Physical and logical connection between two switches across which network traffic travels. A trunk is a voice and data path that simultaneously handles multiple voice and data connections between switches. A backbone is composed of a number of trunks. See also CO.
TTY	A TTY is also known as a TDD (Telecommunications Device for the Deaf).

U

User Datagram Protocol (UDP)	A connectionless messaging protocol for delivery of data packets. A simple protocol that exchanges datagrams without acknowledgements or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols.

Unified Messaging	An application that provides a single network-based access point from which users can manage all information and message types, using any number and variety of access devices (PC, web browser, phone, etc.), from anywhere, and regardless of connection path (LAN, Internet, telephone). Unified messaging solutions seamlessly integrate voice mail, e-mail, and fax in a single e-mail inbox on one server. From a central digital store, all of these message types are accessible via multiple devices and interfaces with a consistent set of features and capabilities.

V

VLAN	Virtual LAN. Group of devices on one or more LANs that are configured (using management software) so that they can communicate as if they were attached to the same wire, when in fact they are located on a number of different LAN segments. Because VLANs are based on logical instead of physical connections, they are extremely flexible. See also LAN.
Voice over Internet Protocol (VoIP)	Technology used to transmit voice conversations over a data network using the Internet Protocol (IP). VoIP primarily builds on and complements existing standards, such as H.323.

W

Wide Area Network (WAN)	A communications network used to connect computers and other devices across a large area. The connection can be private or public.
Wi-Fi	Short for wireless fidelity and is meant to be used generically when referring of any type of 802.11 network, whether 802.11b, 802.11a, dual-band, etc.
Wi-Max	A more powerful version of Wi-Fi that can provide wireless Internet access over wider geographic location such as a city.