WHITE PAPER

Deploying IP Telephony in the Enterprise

Enterprise

In November 2003, Extreme Networks® and Avaya announced a strategic partnership that will accelerate the move toward convergence in the enterprise.

Avaya is the undisputed leader in enterprise TDM and IP telephony solutions deployed in 90% of the Fortune 1000. With 18,000 employees and over $4 billion dollars in annual revenue, Avaya's robust MultiVantage™ Communication application suite will take full advantage of the network intelligence enabled by the award winning Extreme Networks' Alpine®, BlackDiamond® and Summit® families of switches. Extreme is widely recognized as the leader in high–performance, highly resilient enterprise and metro solutions. The leaders in delivering network infrastructure and converged applications are cooperating to tightly integrate, distribute, develop and support critical components required for successful implementation of convergence application solutions.

The multi-year, strategic alliance involves:

- Joint Marketing and Distribution Agreement
- Global Sales/Service Reseller Agreement
- Converged Solutions Joint Development

The alliance will broaden the choices available to customers looking for convergence solutions. For the first time, end-to-end, best-of-breed converged application and infrastructure solutions based on industry standards will be available from a single accountable source. In addition, 7,000 professionals in Avaya's Global Services Organization are committed to helping you with your network assessment, optimization, security, continuity planning, network deployment, training, technical support and maintenance needs.

The Move Towards Converged Networks

What is convergence? Simply put, converged networks are about extending your IP network to leverage a common infrastructure for voice, video, data and all other converged communications. In 2003, PBX line shipments were close to 7 million lines, with traditional TDM–based PBX's declining by 11% to 4.6 million lines and modern IP lines doubling to 2.4 million lines. In 2007, Gartner expects the number of IP lines will surpass TDM–based lines for the first time. The move toward implementing converged communications applications and infrastructure is afoot.

There are many benefits gained from convergence including enabling a new class of enterprise applications that are "communication enabled" resulting in productivity gains through collaboration and un-tethered access to corporate resources from anywhere from any device. However, benefits enabled through convergence can only achieved if all of the components and business processes fit together from the beginning. Networks need to be more application–aware to handle the incremental stress and unique requirements of converged communications applications.

There are many compelling reasons for implementing IP telephony:

- Empower a new class of communication enabled enterprise applications that enhance productivity
- Gain operational efficiencies from managing a single converged network
- Bypass long distance toll charges for inter-office calls within the enterprise
- Least cost routing of off-network calls via the enterprise network
- Give remote workers full featured wire-line and wire-less access to corporate enterprise applications
- Compliment or replace legacy PBXs at a much lower cost and reduced maintenance
- Create next generation contact centers for improved customer relationship management (CRM)

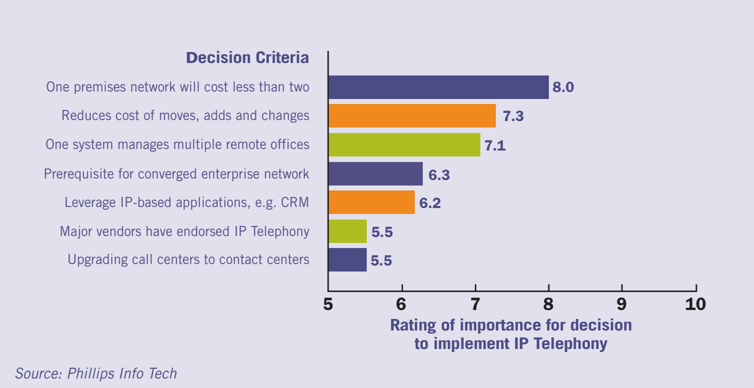

Figure 1. Important factors in considering IP Telephony

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.

This paper addresses issues that enterprises need to consider when deploying internet telephony and converged communications applications into their network infrastructure. The move toward a converged communication infrastructure should be gradual and protect your investments in existing legacy infrastructure to ensure a smooth, low risk transition.

Convergence Applications

Enterprise applications are changing and becoming more accessible through the web and more integrated with convergence applications like voice, instant messaging and video. Consolidating infrastructure for distributed call centers, remote access for sales people at customer sites, video conferencing coupled with computer based training curriculum and full featured IP telephony solutions with access to enterprise applications at corporate headquarters are just some of the examples of the proliferation of distributed converged communications.

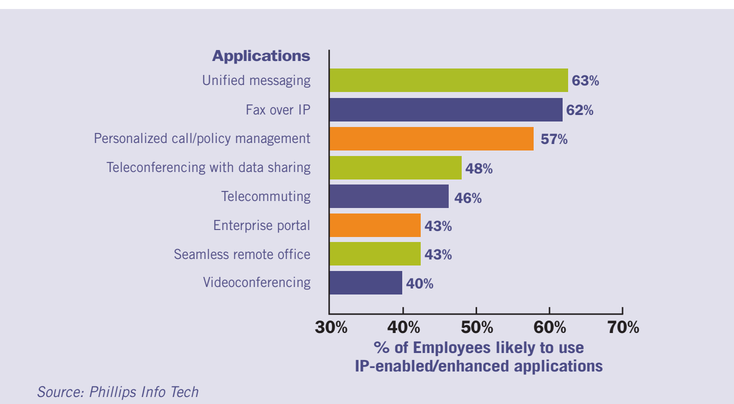

Figure 2. New Class of Converged Applications

Deploying IP Telephony

In the most basic form, IP telephony is about taking phone conversations and converting them into a stream of IP packets and sending them over a packet switched network as opposed to the public switched telephone network (PSTN).

Voice traffic is converted to IP telephony packets by different devices, depending on the architecture of the solution. In toll bypass applications, gateways convert voice between the PBX and IP network. In most IP telephony deployments, packet conversion occurs at the IP phone as well. And in the case of architectures involving analog phones and phone hubs, IP telephony conversion occurs at the phone hub.

Most IP phones uses industry standard protocols such as SIP or H.323 to communicate with other IP devices. However, similar to TDM–based PBX and proprietary handsets, not all IP phones will work with any vendors' IP call manager due to proprietary client/server protocols like Cisco's SKINNY and Nortel's Unistem protocols.

For a business 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 show themselves. 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. A significant portion of IP telephony deployments thus far have been toll-bypass applications.

Implementing an IP telephony infrastructure needs to be an evolutionary step with proper consideration for legacy TDM PBX systems.

Considerations for Voice Quality

The characteristics of TDM–based telephony that we have come to expect is guaranteed dial tone, "pin drop" voice quality, and a robust suite of features. An impediment that slowed the acceptance of first generation IP telephony was that the network infrastructure had been optimized for store and forward type data applications like email, which had very different quality of service (QoS) requirements as compared to voice traffic. Also, many early IP telephony call manager features were lacking compared to their TDM competitors and were not as reliable.

Latency – is simply measured as the amount of time that it takes a packet to traverse the network from sender to receiver. High latency results in speakers talking over each other as they wait for delayed packets to arrive as a response from a sender. When planning your network infrastructure you need to be aware that there are some delays that you can and can't control. For example, on your private LAN or leased lines between sites, you have the ability to manage and control bandwidth to ensure that voice packets experience minimal latency resulting in high voice quality. However, by definition voice packets over the WAN take unpredictable routes across many routers and switches all with different store and forward requirements, buffer/queuing mechanisms etc., which results in uncontrollably high latency. Most experts agree that voice packets can sustain 150 to 300 milliseconds of delay before there is a noticeable impact on voice quality.

Extreme switches normally introduce only 8 to 12 milliseconds per switch when forwarding 64byte voice packets, considerably below the voice quality threshold for latency.

Jitter – is a measure of the variation in latency over time. In data-only networks, jitter is normally not measured as long as the packets arrive in a reasonable timeframe at variable rates. In a converged communications network, jitter can have a big impact on the quality of voice applications resulting in truncated sentences or very choppy dialog. It is critical that the network infrastructure has QoS built into the switches to enable them to buffer packets with minimal overhead to ensure smooth packet streaming rates. Jitter rates of less than 1 millisecond is generally considered accepted by most experts.

Extreme switches normally introduce a jitter of only 10 microseconds, considerably below the jitter threshold.

Bandwidth availability – bandwidth needs to be available and granularly allocated so that voice conversations are not starved due to congestion on the network. Symptoms of poor bandwidth availability result in dropped packets or out of order packets that need to be resent. The end result is voice clipping, skips and dropped calls.

Extreme switches allow for wire-speed switching at Layer 2 and Layer 3 and all have non-blocking switch fabrics ensuring that the switch backplane will never be a congestion point allowing for maximum bandwidth availability.

Echo – often packet switched voice conversations can be impacted by a reverberation of speech back through the handset and re-transmitted causing distracting echoing. Echo cancellation techniques are being implemented in DSP chips that reside in IP telephones to minimize echo.

Codecs – customer premise equipment (CPE) often comes equipped with voice codec software on the device. The purpose of the codec is to convert analog signals into digital packets often using compression before transmission. There are a number of different codecs ranging from the lightweight, lower voice quality of G.711 to G.729 which is much higher voice quality but has higher bandwidth requirements.

Avaya endpoints utilize the highest quality components including codecs and DSPs that support industry standards like SIP and H.323.

Elements of a Converged Network

There are a number of common steps in deploying a converged network that are covered below:

- Create and assign QoS profiles for all devices that might have to support phones
- Configure the logical topology for IP telephony
- Set up your call server clusters or IP PBX to manage features and dial plans for the phones
- Set up a DHCP service for address assignment
- Set up power handling for the phones
- Implement security for your dial plans and to prevent unauthorized network access
- Implement an E911 plan
- Install and set up the phones
- Test the system and dial plan

A related issue is the ongoing maintenance of the IP telephony system. One of the promises of IP telephony on a converged network is reduced cost of Moves, Adds and Changes which is difficult to realize without tools to simplify and automate deployment. The following figure shows estimates on the average relative time it takes to fulfill the requirements of these areas:

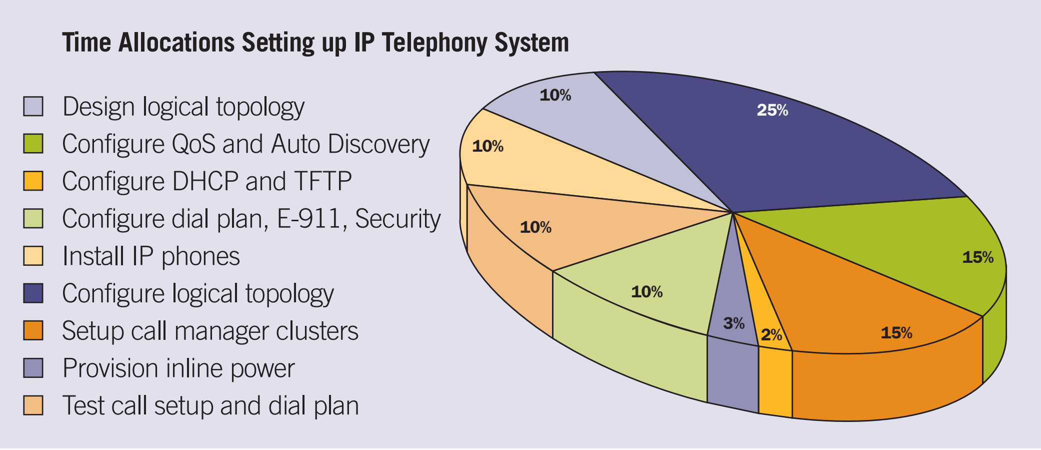

Figure 3. Key IP Telephony

Elements of a Converged Network

These topics and their related issues are discussed in more detail in the following sections.

You should note that logical topology and virtual local area network (VLAN) configuration, along with QoS provisioning, comprise approximately 50% of the Moves, Adds and Changes task set in deploying IP telephony.

VLAN Configuration

A typical campus IP telephony implementation involves the creation of at least one VLAN for the voice service on every switch that may have an IP phone attached to it. The end-to-end QoS needs of voice are very different from those of data and require that each switch be configured with QoS profiles for voice.

Configuring the logical topology of your enterprise network for high availability and IP telephony includes designing the Layer 2 and Layer 3 topologies including VLANs and subnets. It also includes setting up your Spanning Tree (Layer 2) and routing rules (Layer 3).

VLANs are essential in a converged voice/data network. Creating a voice VLAN provides the easiest configuration and management options for IP phones. Separate VLANs for voice and data is highly recommended as it allows you to logically separate and prioritize IP telephony traffic over data transactions.

Voice VLANs are configured as overlays across a network that can be either routed, VLAN-separated, or a hybrid. The telephony VLAN supports only the telephony application components and traffic. However if it is enabled, it provides application isolation for QoS and administration, and also allows specific DHCP service to keep Moves, Adds and Changes easy and cost effective within the call server zone. Ideally, an IP telephony VLAN should also be its own subnet, fully routable across Layer 3 links or to interconnect voice VLANs that have been separated for administrative or other reasons.

Phone Connectivity Options

A significant portion of the cost of a implementing a converged communications infrastructure in large enterprises is the cost of deploying a large volume of IP handsets. Depending on functionality, wired IP phones or wireless IP phones using 802.11b standards and can range in price as low as $99 or expensive as $500 with many features such as color LCDs. Most IP handsets generally need to support the often proprietary signaling protocols of the IP call manager. In the TDM PBX/proprietary handset world, the "proprietary razor/razor blade" pricing model has existed for generations and it appears that IP handsets will continue down the same path.

As part of your converged network, you have to make some choices as to how your phones are going to be configured at the desktop and how they will be connected to the network. One configuration is to have the IP phones use a dedicated switch port connection; this option requires two drops to each desktop, one for the phone and one for the PC. This option provides a level of physical redundancy at the cost of an extra switch port.

In order to reduce costs, a common alternative is to daisy-chain the desktop PC to the IP phone. This option essentially shares the network switch port and cable between the PC and IP Phone. This aids in rapid deployment and facilitates maintenance later with Moves, Adds, and Changes but does requires a more intelligent switching infrastructure. To achieve this, the IP phone basically has a three port switch built in, so that there is a 10/100/1000 port connection from the phone to the network switch, a 10/100/1000 port connection that connects to the desktop PC, and the internal port that connects to the phone itself. Most vendor IP phones support this mode.

Of course, if you are using a soft phone (driven by software in the desktop PC), the only connection is to the PC.

You do not need to use the same configuration at each desktop, but the connectivity option you choose may affect the feature sets available to you and the maintainability of the network at the edge.

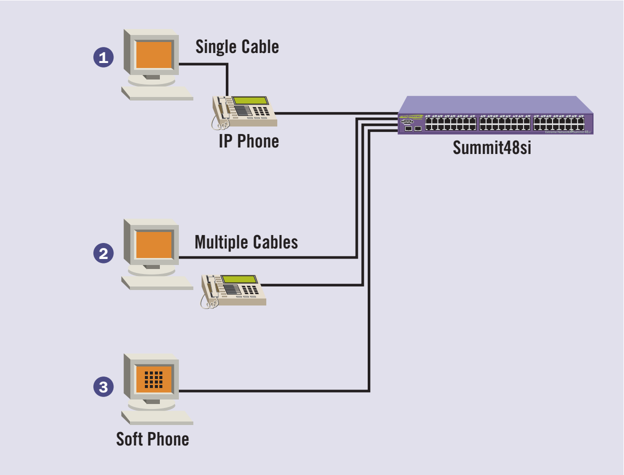

Figure 4. Three different IP phone configurations

Analog phones, which you may still use in some areas, must connect to an IP network via a voice hub. The voice hub will digitize and packetize the analog signals, and additionally provide control and/or call signaling to other voice hubs, call servers, other SIP or H.323 endpoints as well as gateways to the PBX or PSTN. Alternatively, depending on which telephony vendor you select, you may be able to re-use your analog phones by connecting IP call servers or a related component to legacy peripheral cabinets. In this case the conversion to IP occurs at the device aggregating the peripheral cabinets.

QoS Profiles

IP telephony VLANs contain much more, of course, than the edge switches connecting the phones. They also must include all aggregation and core switches end–to–end between all the IP phones throughout the enterprise.

All relevant devices that might have to forward voice traffic must be configured with QoS profiles. Doing this once across the network may mitigate the need to make changes at the edge for each IP phone Move, Add or Change. However, as more phones are added at the edge, or if large numbers of phones are moved from one switch to another, QoS profiles at the aggregation layer and core will need adjustment to guarantee more aggregate bandwidth.

In addition to simple packet marking and classification, your infrastructure needs to be able to quickly parse and prioritize queues for congestion management. Even better, is the ability to monitor traffic and anticipate congestion and react before performance degrades or does not meet a QoS service contract.

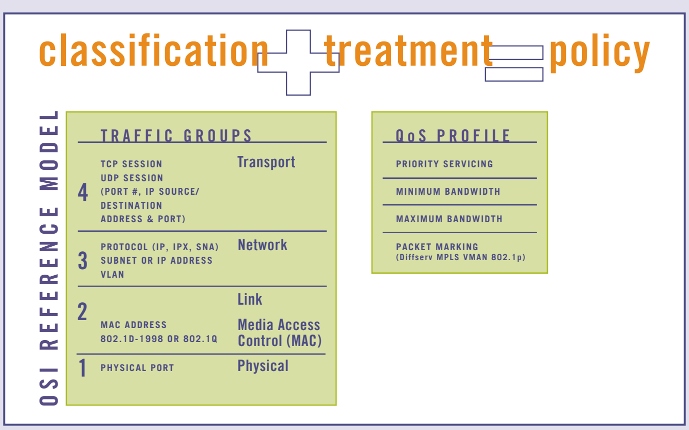

Figure 5. Extreme Classification and Treatment

Extreme's Policy–based QoS

Even though raw bandwidth capacity can be plentiful in a LAN/Campus network, you still need to provide a guaranteed level of packet throughput for IP telephony for it to function well. Without proper QoS settings, the bursty, bandwidth–hogging potential of certain data and video applications increases the latency and jitter of voice to unacceptable levels.

Extreme's Policy–based QoS relies on several components to achieve the desired results that allow network administrators to get the proper control over managing the bandwidth characteristics various applications demand. Stated simply, Policy–based QoS consists of robust traffic classification combined with bandwidth management treatments:

Classification + Treatment = Policy–based QoS

These policies allow network administrators to control the bandwidth various applications use and maintain latency and jitter control over the applications that need it. Extensive classification without robust treatment capabilities are quite useless. Imagine having a carpool lane when there is only a single lane—that's similar to no classification. Imagine no traffic laws on a multi-lane freeway, without minimum or maximum speed limits—that's similar to no treatment capabilities.

Classification can be made using either explicit information (such as 802.1p DiffServ code points) or implicit information (such as membership of a voice VLAN, or TCP/UDP ports).

Treatment refers to prioritization and bandwidth management, which is handled in Extreme Networks' switches via min and max bandwidth control, in conjunction with eight queues on every switch port (in the I series chipset switches). After traffic is classified, it is assigned to one of these priority queues for servicing. Each queue may also be rate-shaped using bandwidth controls, thereby also controlling latency.

Examples of classification (Traffic Groups) and treatment (QoS profile) are illustrated in the diagram on the left.

Call Servers

Call setup and control is a key IP telephony function. It can be provided by a centralized call server with optional backup servers, or can be distributed throughout the network, depending on implementation. Alternatively, an IP PBX, or an IP telephony-enabled interface for a standard PBX could also offer direct attachments to the IP network and provide call management services.

Call servers manage the features and dial plans of the phones. Setting up a call server involves configuring rules for communicating between different call servers, ensuring redundancy, and configuring device (phone) pools. You must build a table containing a list of IP phone Media Access Control Addresses and the phone extension that should be allocated to each phone. As new phones are added to the network they will need to be defined to the call manager in this table.

DHCP Service Configuration

You can assign IP addresses for the phones using either DHCP or static configuration. Since the phones will require a lot of IP addresses, you will probably prefer to use DHCP so addresses can be reused. This is another reason why telephony VLANs should be on their own subnet if possible—it will be easier to track IP addresses used for phones.

Phones use the DCHP servers to get information about what “call servers” exist. Some telephony vendors have implemented extensions to DHCP to assign phones to their appropriate VLANs in conjunction with IP address assignment. Features like this will continue to make deployment and configuration of IP telephony more “plug and play.”

Inline Power and Power over Ethernet (PoE)

Most traditional phones operate during power failures as a result of being powered by the PBX itself. In an IP telephony environment, not only must uninterruptible power be supplied to the call servers and gateways, but each IP phone may also need similar power protection in order to operate, in local power outages. Reliable power delivery through inline power to IP phones is critical for service survivability and E911 services support. Inline power reduces the risk of lost power to the phone and reduces the support costs associated with troubleshooting phone outages.

Inline power may be implemented as mid-span powered hubs or integrated in to Ethernet switches themselves. Extreme supports both third party mid-span power hubs as well as the recently ratified IEEE 802.3af standard for PoE.

With inline power support at every seat, enterprises avoid having to verify and possibly change power availability at the wiring closet every time an IP phone is added or moved. This long-term goal of ubiquitous power is a single piece in the puzzle of eliminating support costs associated with service deployments.

Extreme’s VLAN Manager and VoIP Manager

Extreme Networks' switches have been designed from the beginning to support network convergence. With eight hardware-based queues per port, each with the ability to offer a minimum and maximum bandwidth allocation, Extreme’s ability to support IP telephony along side other mission critical applications is unparalleled in the industry. (See the Hardware Profile for “I” Series Chipsets tech brief for more information.)

Moreover, Extreme Networks has automated as much of this process as possible. As part of the EPICenter® network management suite, Extreme Networks continues its tradition of award-winning provisioning tools with VoIP Manager, helping enterprise network admins configure IP telephony networks in the most efficient way possible.

VLAN Manager and VoIP Manager are both fully integrated into the award winning EPICenter management Solution. You can create VoIP VLANs through EPICenter’s VLAN Manager, and VoIP Manager inherits these VLAN definitions.

In addition to creating and configuring VoIP VLANs and QoS profiles end-to-end, VoIP Manager calculates end-to-end min/max bandwidth requirements based on the G.7xx compression algorithm used, number and location of IP phones. VoIP Manager also provides status reporting for all VLANs on a per-switch and per-port basis.

Extreme is working closely with Avaya to ensure tight integration of tools for management for both applications and infrastructure.

Implementing Security

Voice security is at least as sensitive a topic as data security. Users expect that all voice communications are confidential (i.e. no one is listening in), even when they don't have the same expectations of an e-mail containing the same information. Similar to your needs to protect the infrastructure against rogue access points being inserted in the network, you should also prevent unauthorized Moves, Adds and Changes of IP phones, especially while automated procedures (QoS adjustment, automated E911 tracking, etc.) are not available.

Password or access list changes may be needed to accommodate a Move, Add or Change. Even if a phone changes VLANs, some authentication updates may have to be made to accommodate its new physical location. Also, there may be some database updates needed for accounting or billing purposes. This is especially true (as with QoS) if the switch now supporting the phone was not aware of the VLAN before the move. The call server may need to generate a new security profile based on the phone's new physical location.

Configuring Phones

The phones now have to be configured. This is a task that will vary depending on your architectural choice at the desktop—single cable, multiple cable, or soft phone.

This is an area where management tools can help with VLAN assignment and QoS configuration. See the VoIP Manager Tech Brief for more information on how Extreme simplifies this process.

Testing the IP Telephony System

Before deployment of even a part of the IP telephony infrastructure is complete, extensive testing on a stand alone controlled network has to be done on the IP telephony system including on-net and off-net calls, the dial plans, E911 emergency calls, and fail-over conditions.

Training, Support and Maintenance

Because converged communications application 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.

Extreme's Infrastructure Security

When IP telephony is deployed, network ports are exposed from semi-public or public areas—for example, there may be an IP phone in the lobby or a wireless access point in a conference room.

Extreme Networks has advanced security capabilities. Access Control Lists (ACLs) filter out inbound traffic (such as HTTP, FTP, Telnet, SMTP etc.) that may not be relevant to an IP phone. Thus, thenetwork remains secure even in the case of Layer 2 address spoofing, and phones are protected against Denial of Service (DoS) attacks.

With ExtremeWare's Layer 2 Address Security, a port can be limited to a number of MAC addresses that are learned by the port (e.g., one address for a pure IP phone port). The MAC address of an IP phone can be learned dynamically when the phone is installed, and the port is then "locked down" to this MAC address, even after a reboot. This provides an excellent level of security against port abuse.

In cases where ports are allowed for data clients as well, phone discovery is combined with user level access security. Network Login is an excellent user level security feature in ExtremeWare that requires no client software, and thus works even in semi-public environments such as universities and libraries. Network Login works with any client device and operating system that supports DHCP and a web browser. In addition, Extreme Networks is following the standard for 802.1x (port based network access control) that will play an important long term role in enterprise environments. The combination of Network Login and 802.1x on the same port will allow deployment of user based security immediately, without requiring network client device changes.

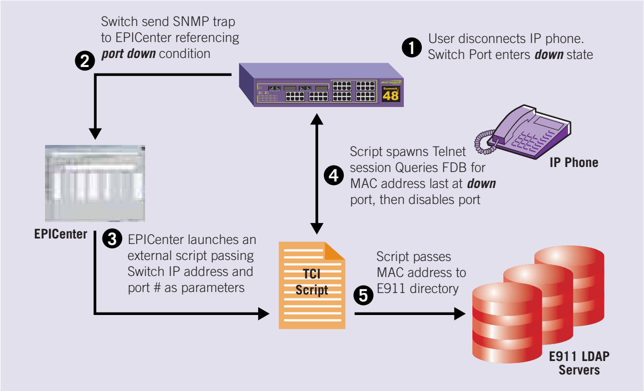

Figure 6. Extreme's 911 Phone Movement

E911 Considerations

An E911 service has to provide automatic number and location information (ANI and ALI) to a 911—operator 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.

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.

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.

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.

Dial Plans

The dial plan architecture includes dial plan groups, calling restrictions, and on-net route patterns. This area 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 Chicago wishes to call off-network to a customer in San Jose, the call can be routed over the enterprise network and get handed off to the public phone network via a gateway in the San Francisco office (the nearest office to San Jose), thereby being charged only for a local call instead of a long distance call from Chicago to San Jose.

Extreme's Rapid E911 Support

With Extreme Networks' infrastructure, E911 database information can be maintained very rapidly, whenever a phone is moved. Plugging and unplugging a phone generates "Link up" alerts (when the phone is plugged in) and "Link down" alerts (when it's unplugged). This provides information to E911 applications to detect IP phone movements allowing databases to be updated automatically.

The benefits and implications of turnkey E911 support include more rapid Moves, Adds and Changes which include two main components:

- physically relocating station equipment or new construction

- software-related updates, such as switch/router QoS or authentication configuration, E 911 database updates, and call manager-related functions such as display names, call coverage, phone numbers etc.

The process might start with a call or e-mail to a help desk. A typical Service Level Agreement (SLA) might call for software changes within half a day and physical changes within 24 hours. Many of the software changes with an IP Telephony Move, Add or Change will be similar to their PBX based counterparts, but can be automated to a greater degree with sophisticated infrastructure tools that take advantage of the intelligence of your converged network.

End-to-end, Policy-based QoS configuration is another area where Extreme Networks is ahead of the field in both rapid deployment and rapid Move, Add and Change support – see the VLAN Manager and VoIP Manager sidebar and the VoIP Manager Tech Brief for more information.

WANTED: End-to-End, Best-of-Breed Convergence Solutions

To get the most out of deploying business critical IP telephony solutions, it is a necessary requirement that all of the applications and network infrastructure have a multiple-layer "dialog" mediated by open standards encompassing authentication, QoS, security, device discovery, auto configuration, monitoring and management.

Early attempts at convergence placed unnecessary burdens on early adopters that slowed the acceptance of IP telephony. In some cases, customers had to perform complex systems integration to make sure that hardware and software from multiple vendors actually worked together and could be supported locally and remotely in branch offices and call centers. Solutions often ended up being piece-meal with little regard for existing TDM based infrastructure. Customers became tired and frustrated being "ping-ponged" back and forth between disparate vendors when seeking support. Another route that early adopters took was to attempt to purchase complete solutions from a single vendor. However, often the customer had to settle for unbalanced, end-to-end solutions from either telephony vendors who were not well versed in the latest advancements in network architecture like QoS or from data infrastructure providers whose telephony solutions lacked some of the basic call manager functionality of their robust featured TDM PBX counterparts. These vendors who attempted to deliver end-to-end solutions sometimes offered point products that were sub-optimal in both telephony and networking. Additionally, in order to get end-to-end solutions, customers were held hostage by proprietary protocols that increased vendor lock-in and inhibited customer choice and negotiating power.[1]

Telephony and data networking are vastly complex businesses requiring vendors to make enormous investments to stay current in their own highly competitive markets where they have strong domain expertise. Customers need solutions that are sensitive to their past investments yet give them head room for growth and the opportunity to exploit new developments in areas like video-conferencing that will give them a competitive advantage. Even with the largest vendors, it is unfathomable to believe that one single vendor can develop solutions that are competitive in both IP telephony and network infrastructure.

[1] Cisco Systems and Voice: The Untold Story.
Gartner Group Research Note July 2002

Cradle to Grave Support for Convergence Solutions

Dial tone simply always has to work. Your business depends on it. The cost of downtime in a global economy can be tremendous. Implementing converged communication architectures is a lot more than just buying telephony and network equipment.

You should involve your vendors as partners in the design, pilot and customer application integration phases. End-to-end solutions need end-to-end planning, maintenance and ongoing support. When deploying a converged solutions you should expect 24x7x365 support on a world-wide basis from your vendor for the entire solution not just a single component.

Your business is too important and the opportunity cost is too great to experience vendor finger pointing. Demand accountability from your convergence vendor for the entire converged solution.

Extreme Networks: Network Infrastructure of Choice for Avaya's MultiVantage Communications Architecture

Extreme Networks was chosen by Avaya to be the infrastructure foundation for their Communications Architecture encompassing infrastructure, applications and services highlighted below:

Telephony solutions with Avaya Communication Manager use industry standards to enhance scalability and flexibility in traditional voice systems, and serve nearly any employee need with IP or traditional telephony solutions.

Contact Center solutions in the Avaya Customer Interaction Suite combine multimedia integration, self-service and process automation, efficient routing, service-level management, reporting, and more for maximum efficiency in enterprises of any size.

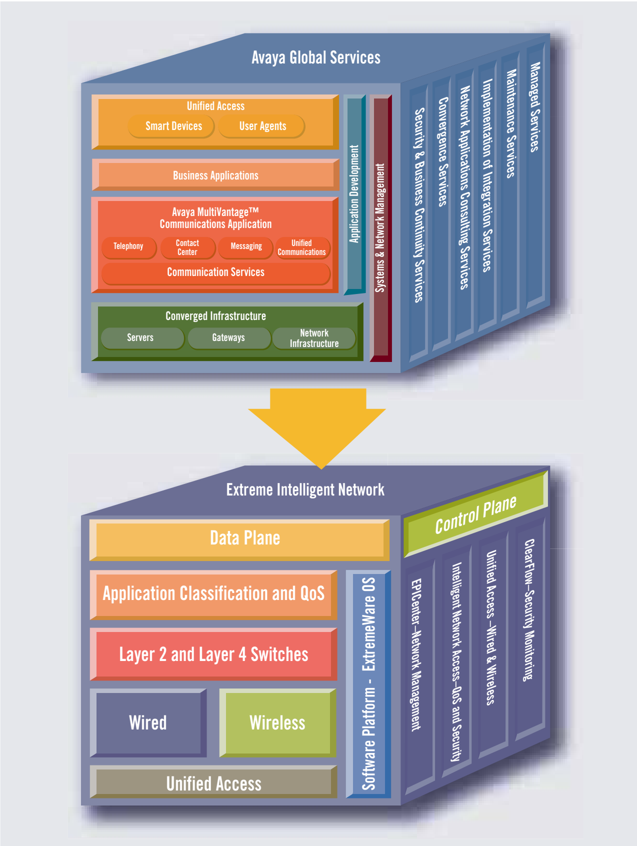

Figure 7. Extreme Networks: The Infrastructure for Avaya's MultiVantage Communication Applications

Messaging solutions with Avaya Modular Messaging and Avaya Message Networking combine scalability, reliability, and availability whenever and wherever they're needed-so employees can better manage their time and collaborate 24/7 across a virtual enterprise.

Unified Communication solutions with Avaya Unified Communication Center offer one friendly interface for all of the network services users rely on, plus speech commands that provide access to features and database information through any phone, Web browser, cellular or wireless device.

System and Network Management: When combined with Extreme's EPICenter, Avaya Integrated Management, Network and Policy Management and Security Management provide a comprehensive set of tools that make it easier to manage complex network infrastructures.

Avaya Global Services: Single point of accountability to design, build, and manage multi-vendor communications networks worldwide with full life-cycle support for your critical communications.

Security and Business Continuity Services consist of consulting services that help safeguard and control access to mission-critical infrastructure components in voice, data, and converged networks. Convergence Services provide for every aspect of planning, implementing, and maintaining converged communication solutions.

Network and Applications Consulting Services support seamless, end-to-end network operations including assessment, design, and optimization services that provide the blueprint for integrating new applications and technologies and optimize network availability and performance

Implementation and Integration Services employ proven methodology for implementing voice, data, contact center, and convergence solutions including customer evaluation, converged and multi-vendor data expertise, and value-based service pricing at the solution level.

Maintenance Services provide maintenance for voice, data, and converged needs through a comprehensive, flexible Avaya Maintenance Services Agreement. Avaya EXPERT Systems Diagnostic Tools provide unparalleled remote monitoring, diagnostics, and resolution for voice systems

Managed Services provide fault, performance, and configuration management for multi-vendor data networks and Avaya MultiVantage Communications Applications solutions.

Avaya Converged Communications Product Suite

Avaya MultiVantage™ Software: Avaya's MultiVantage is the heart of the ECLIPS platform, offering all of the features and functionality of traditional business phone systems on an IP converged network. The software operates on Avaya ECLIPS media servers and gateways as well as existing circuit switched Avaya DEFINITY servers.

Avaya Media Servers: Based on Linux® and Windows® operating systems, Avaya Media Servers deploy voice applications onto enterprise networks. Avaya offers three media servers—S8100, S8300, and S8700—each scaled for differing communications network configurations, including campus, multi-site, branch, remote, and home office.

Avaya Media Gateways: Avaya Media gateways support voice and signaling traffic routed between packet-switched and circuit-switched networks. Two media gateways—G600 and G700—provide scalable support for large or small IP telephony deployments. Both adhere to IEEE standards, allowing internetworking with a wide range of data networking infrastructures; Avaya's software permits continued compliance as IP standards evolve.

Avaya Integrated™ Management Suite: Avaya's Integrated Management Suite provides a flexible and comprehensive set of web-enabled tools designed to simplify the management of IP telephony communications networks, including IP telephony fault monitoring, performance management, policy management, and configuration.

Avaya IP Handsets and Softphones: Avaya's line of IP-enabled telephones and consoles provide a full compliment of station features, impressive voice quality, and incorporation of critical features such as E911 MAC address support. Avaya's commitment to innovative IP terminals includes the IP Softphone for Pocket PC, which converges IP telephony, Wireless LAN, and telephony features into a single PDA-enabled handset.

Extreme Networks Family of Intelligent Network Switches

The award-winning BlackDiamond 6800 product family is optimized for large enterprise network core and data centers and provides a highly resilient, highly scalable, feature-rich platform with:

- Hitless failover and hitless software upgrades, ensuring the network remains operational even under the most adverse conditions and while maintenance is being performed
- Granular QoS capabilities that provide superior support for IP telephony and other mission-critical applications
- Support for industry-standard protocols and interfaces—including 10 Gigabit Ethernet, MPLS, PoS, and ATM—that enable flexible integration into a wide variety of metropolitan or legacy networks
- Scalability to a maximum of 1,440 10/100BASE-T ports and 360 Gigabit Ethernet ports with a total switching capacity of 768 Gbps

The Alpine 3800 product family is targeted at smaller core networks and distribution applications within larger enterprise networks. It offers:

- A highly resilient platform with high–performance and non-blocking interfaces
- Many of the same features as BlackDiamond in a smaller, more cost-effective chassis
- Scalability to 256 10/100BASE-T ports or 128 Gigabit Ethernet ports, and a switching capacity of 64 Gbps

The Summit product family spans several fixed-configurations for the wiring closet and mid-tier aggregation applications and:

- Features robust, wire-speed Layer 3 switching, intelligent Policy-based QoS, and advanced security

The ExtremeWare® Operating System delivers the uncompromising management, control, and security required by today's demanding enterprise networks. This software base, common to all Extreme switch platforms, includes:

- Standards-based, multi-layer switching and Policy-based QoS to give corporate networks the best available tools for optimizing operating capacity
- Plug-and-play deployment and stable, consistent performance out of the box

EPICenter delivers an integrated network management suite that simplifies configuration and network monitoring for all Extreme products within large enterprise networks.

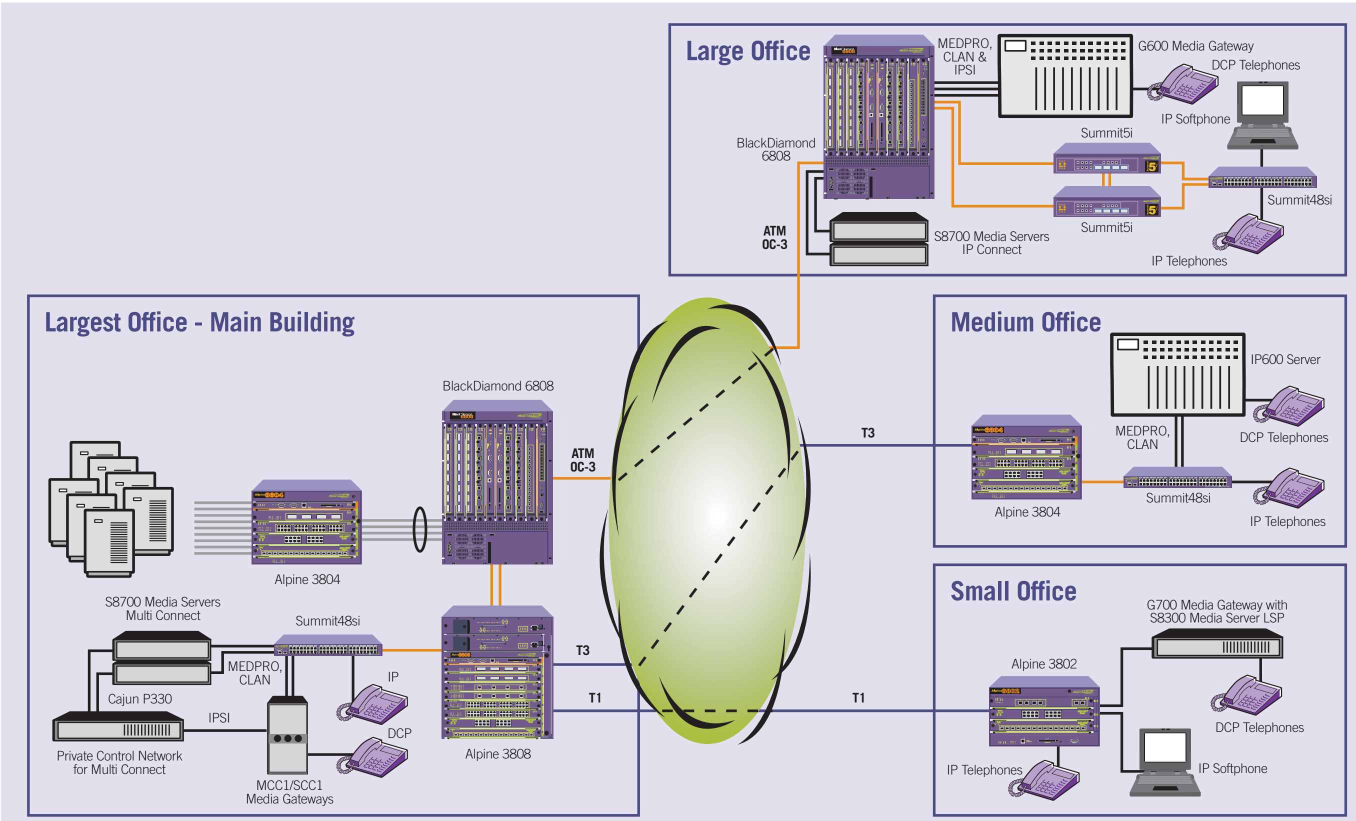

Figure 8. Avaya and Extreme Networks Delivering End-to-End, Best-of-Breed Solutions for the Distributed Collaborative

Conclusion: Success Factors in Deploying Internet Telephony

The hype over internet telephony from just a few short years ago has subsided. Despite bold predictions from entrenched data infrastructure vendors with their eye on the voice market, large enterprises did not make wholesale donations of the legacy PBX equipment. However, they have begun the process of gradually implementing plans to converge their next generation voice and data networks with emphasis on protecting their legacy TDM PBX investments. As the hype continues to subside, most enterprises realize that a single vendor regardless of whether they come from the voice or data perspective is unlikely to develop and deliver complete converged solutions that are competitive and can be supported for the long term all by themselves.

As shown in the topology diagram above, the robust platforms available from vendors like Avaya and Extreme Networks can support any size location whether it is centralized or distributed without compromising features or performance. The partnership is ultimately aimed at lowering TCO by simplifying IP telephony deployments. The partnership's stated goal is to increase uptime through collaborating on resiliency, self-healing network development and service aware routing where applications can respond in real time to available bandwidth. Integrated tools for network management, standards based device discovery, plug and play auto-configuration and enhanced security are just some of the fronts that the companies are working on together.

The benefits of convergence are real and are starting to be obtained by enterprises who enable their workers, suppliers and customers to achieve productivity gains through secure access to communications-enabled enterprise applications from a plethora of devices distributed around the globe. The tremendous benefits from communications applications can not be achieved unless the underlying network has specifically been optimized for convergence. Performance and reliability are obviously key, as phone systems must be available at all the times and voice quality has to be maintained on a policy basis with special attention to latency and jitter. But there are also some unique needs in terms of the ability to power the phones, deploy and maintain the IP telephony system, and maintain the security of the network as a whole.

Customers have spoken. They want application and network infrastructure vendors to take responsibility for integrating and supporting end-to-end solutions that allow communication applications to be able to take full advantage of the intelligence inherent in the network infrastructure. Furthermore, customers want cradle–to–grave accountability for network assessment, optimization, security, deployment, training and supporting converged networks.

The strategic alliance between industry leaders Extreme Networks and Avaya is designed to give customers exactly what they asked for – a choice of best-of-breed, end-to-end solutions specifically designed and integrated for convergence while being delivered and supported by a single accountable vendor.

For more information as to how Extreme Networks provides the best-of-breed network infrastructure for deploying IP telephony applications, see the VoIP Solutions Brief, the VoIP Tech Brief, and the VoIP Manager Tech Brief. To find out more about Avaya Unified Communication based on Extreme intelligent network infrastructure solutions, please contact your Avaya Client Executive or Authorized Business Partner, or visit avaya.com/learnmore/uc.

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