Demystifying IP Migration for IT Professionals

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Author:
Timothy M. O'Neil
Corporate, Director of Technical Marketing
Polycom, Inc.

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Overview

The application and transport convergence that is occurring today in modern IP networks is significantly impacting Collaborative Networks (CN) and Collaboration Applications (CA). It is the intent of this paper to discuss all of the pertinent variables that must be considered in order to cost effectively and efficiently deploy or migrate CN and CA to IP networks.

The panoply of technologies, standards, and applications are discussed and proven methodologies are recommended. Every effort is made to aid the designer in developing a best of breed, scalable, and cost effective CN.

The migration from ISDN based CN to IP based CN requires in-depth knowledge in order to rapidly and smoothly deploy. Deploying IP based CN extends beyond simply connecting video communication terminals to the LAN. A successful implementation of a CN requires a well thought out architectural design and a clear concise plan for components to facilitate the ease of use and reliability required by end-users and IT administrators.

In order to properly identify all of the variables in deploying IP based CN/CA, network administrators must understand CN architecture, as well as CA and IP networking variables. These concepts can then be applied to the deployment of CA. Whether the deployment is a small work group, multinational CA service, or carrier based CA service provider the concepts are the same. This guide helps to simplify the process of selecting the right components and processes required to properly deploy.

Figure 1:
Unified Collaborative
Network Architecture Model

Polycom Accelerated Communications (PAC) is the industry's leading standards-based Unified Collaborative Network (UCN) architecture. See Figure 1 for a reference architecture model. Polycom is the only supplier of a completely integrated system of both CN infrastructure and applications. Most importantly Polycom is the leader in Unified Collaborative Networks.

Return on Investment

Most modern networks are now built out to support real time collaboration and communication applications. One major driver for moving to a converged network is the ability of a company to run the applications that makes the business more efficient, effective, global, and competitive. After a UCN is deployed, that is the time to count return on investment (ROI). Unless it previously deployed a CN a business cannot effectively take full advantage of routing and switching purchases. Polycom continually hears the following from our customer base "I am running at less than five percent capacity in my core" laying a great foundation is not enough. The user community also needs tools and services that actually affect their daily work processes, thus the true total cost of ownership (TCO) and ROI; or as Gartner Research defines it Total Value of Opportunity (TVO).

[1]According to Gartner Research: "In more than 80 percent of the projects we followed, after the business initiative was launched, the project was not monitored or bench marked against the original projected benefits."

If you have not fully considered collaborative applications prior to a network or application convergence project and subsequently measured their impact on TVO, how can you be satisfied that you have made the right decision about the true business value?

Investment Protection

You can rest assured that Polycom is partnered with the leading routing, switching, and communication vendors. This ensures that you can seamlessly deploy our solution on your network. As you can see within the UCN architecture model in Figure 1, PAC is optimized to overlay on your state-of-the art Cisco, Avaya, Nortel, or 3Com network. Additionally,
Polycom provides investment protection through efficient downloadable upgrade software for all products. Unlike our competitors, a forklift is not required to upgrade to the latest version. In addition, our modular infrastructure systems allow for a simple swap out/snap in of legacy cards for the newest processing capabilities. These are field upgradeable; again, no forklift required.

The Polycom Office™

Polycom's Unified Collaboration suite of voice, video, Web, scheduling, and network management products and applications is called The Polycom Office. Polycom Unified Conferencing is a revolutionary

[1]Gartner Research, Research Note, Decision Framework, DF-17-0235, Author; A. Apfel, August , 2002.

conferencing system. In a Polycom Unified Conference, all voice and video conference participants have access to all of the conference features expected in a video or voice conference. Conference entry queue, roll call, question and answer (Q&A) sessions, and voting and polling sessions are examples of features that can be utilized in a unified Web, voice and video conference.

Polycom delivers the widest depth and breadth of collaborative applications and services on the market today. The Polycom Office and our accelerated communication architecture meet the demands and feature requests of our global installed base and the analyst community. Independent verifications abound with accolades about our recent 5.0 release coming from all sectors.

Network Computing, August 21, 2003
"Polycom KOs Proprietary VoIP Woes"

Network World Fusion, August 18, 2003
"Simplicity key to videoconferencing success"

CRN, February 24, 2003
"Polycom Unleashes Conferencing Product Blitz"

eWeek, February 24, 2003
"Unifying Video and Audio Conferencing"

True Thought and Technology Leadership

There are partial competitors that talk about having all or most of the requisite pieces of a UCN. Be cautious when evaluating the width of these so called competing solutions without first carefully investigating the depth of features and services that are supported. Do not simply investigate the data sheets or only pieces of the solution physically try the solution in your environment and independently measure the truth for yourself.

Recent Leadership Examples

It is Polycom's position that true technology leadership requires more than just participating in standards. At Polycom we drive those standards from invention to participation, and then to implementation in products. Leading in technology and standards is part and parcel to building a market leading UCN. Polycom's commitment to technology and standards provides you with an assured path to UCN IP migration making IP migration simpler, more cost effective, and certainly allowing for the leveraging of a greater ROI/TVO.

The most recent standards work completed includes:

- ITU-T H.264 (A.K.A MPEG AVC): Video algorithm and flexible macro block ordering (FMO); error concealment. The best video.
- ITU-T H.239 (H.AMC): Role management and additional media channels for H.300-series terminals. The newest and best collaboration standard.
- ITU-T G.722.1 (Siren™ 7): Wide band audio algorithm. The best audio.

When it comes to state of-the art technology, Polycom leads the charge. This leadership even extends to the licensing of our intellectual property free to our [2]competitors; as was the case with our People+Content™ data sharing specification which ultimately became ITU-T H.239 (H.AMC) standard. This leadership also extends to Polycom's patented technology in open standards. FMO error concealment in H.264 is based on Polycom's patent pending technology. Both the [3]MPEG Licensing Authority and [4]Via Licensing Corporation, which manage the business of intellectual property in standards work, have stated that FMO is recognized as an essential part of H.264 and not an optional component; it is core to H.264.

It is this kind of true industry leadership that brings features like H.264 video to the market sooner. Polycom was the first to bring H.264 technology to market, (in February, 2003). This work is beginning to improve the quality of every communication session that uses H.264. Polycom technology leadership is found even in competing vendor products. H.239, H.264, and G.722.1 are just the tip of the iceberg when it comes to Polycom technology and thought leadership. Please view the [5]Polycom and Standards white paper for more details.

So if it looks better, makes you sound better, improves your hearing of remote participants, and improves collaboration' ease of use then you know Polycom is involved.

The Bottom Line

Polycom is the only company that provides a fully integrated Unified Collaborative Network (UCN) offering. A true UCN supplies unencumbered access to every feature, yet does not require all of the components to make it work. Some vendors only allow a Web conference that is used exclusively in conjunction with video conferencing, while others provide Web conferencing exclusive of video conferencing. Be wary of a vendor that touts its inflexibility and limited solution set as an optimized solution for your conferencing needs.

At Polycom, our tag line "Connect Any way you want" is really our design mantra. You can choose Web and audio; audio only; or unified Web, audio, and video. Because of our unique PAC architecture, The Polycom Office is the most flexible solution available, and still the only true standards based UCN. Migrating collaborative applications to a single architecture and application suite has never been easier. So go ahead start the process of migrating some or all of your ISDN videoconferencing to IP, move that outsourced Web conferencing in-house and roll those audio calls over to your in house Unified Conferencing resources. Polycom delivers a single administrative and user interface that provides all services seamlessly and securely on your network.

[3]Press release, Development of Joint Patent License for H.264/MPEG-4 AVC Makes Progress, www.mpegla.com/news/n_03-07-07_avc.html
[4]Via Licensing Corporation, www.vialicensing.com
[5]White paper, Polycom: Market Leader and Industry Leader, free download from www.polycom.com/common/pw_item_show_doc/0,1449,2038,00.pdf

The Standards

Originally, Collaborative Networks were based solely on the International Telecommunications Union (ITU) H.320 standard. The H.320 series of standards defines Integrated Switched Digital Network (ISDN) connection-based video communication.

This technology leverages the existing Public Switched Telephone Network (PSTN). The cost of deployment and inability to scale to large numbers of cost effective communication sessions led to the development of the ITU H.323 standard. It covers packet-based multimedia communications over Transmission Control Protocol/Internet Protocol (TCP/IP). H.323 is a logical extension of H.320, which was developed to enable corporate intranets and packet-switched networks to transport multimedia and video communication traffic. H.323 recommendations cover IP devices that participate and control H.323 sessions, as well as video specific infrastructure that interacts with the PTSN. In common with other ITU multimedia teleconferencing standards, H.323 applies to either point-to-point or multipoint sessions. The ITU has ratified these core protocol components for audio, video and communication for H.323 sessions:

- H.225: Specifies messages for call control including signaling, registration and admissions, and packetization/synchronization of media streams.
- H.245: Specifies messages for opening and closing channels for media streams and other commands, requests and indications
- H.263: A video codec that adds picture formats over H.261 (4CIF and 16CIF)
- G.723: Audio codec, for 5.3 and 6.3 Kbps modes
- H.264: The newest video codec, designed to replace both H.263 and MPEG
- H.239 (H.AMC): Integrated video and data collaboration
- G.722.1 (Siren™ 7): Wide band audio algorithm

These ITU protocol components were previously defined in H.320, but also apply to H.323.

- H.261: Video codec for audiovisual services at p x 64 Kbps
- G.711: Audio codec, 3 KHz at 48, 56, and 64 Kbps (normal telephony)
- G.722: Audio codec, 7 KHz at 48, 56, and 64 Kbps
- G.728: Audio Codec, 3 KHz at 16 Kbps

Polycom's Accelerated Communications Architecture

The Polycom Accelerated Communication Architecture (PAC) is a three tiered architecture. See Figure 2.

I. Tier one

Consists of **Terminals** (end points) that users interact with from a user interface (UI).

II. Tier two

Belongs to our **MGC** media processing network infrastructure. MGCs are responsible for processing audio and video media streams, transcoding, and media quality optimization.

III. Tier three

Belongs to our **Application Server** software, which spans network aware scheduling, device and directory management, Web conferencing, and bandwidth management/call processing.

IP Basics for Implementation

When deploying an IP-based network that will be used to support Collaborative Applications, an understanding of how video specifically differs from other IP-based applications is required. The three key differences are:

1. CA are real- time applications.
2. CA can use higher bandwidth.
3. Firewalls can be an obstacle to the traversal of video CA traffic.

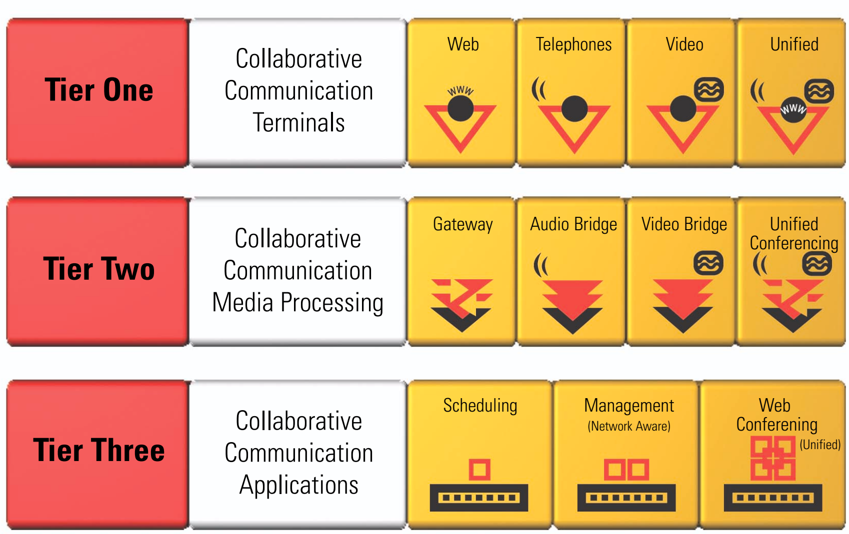

Figure 2: Polycom Office and UCN Architecture

Collaborative Applications Are Real-Time Applications

Video is a new media type for most IP networks. Although real-time applications exist today, CA, unlike e-mail or typical database applications, requires limits on total end-to-end delay (latency), and variability of the delay (jitter). Figure 3 shows how the latency and jitter are linked. Jitter is a prime contributor to packet loss, which is responsible for degrading audio and video quality and usability. It should be noted that the overall delay budget for a one-way video or voice conversation is approximately 150 milliseconds.

A New Protocol for Real-Time Data Transport

TCP, the Layer 4 protocol which serves as the data-transport mechanism for most packet-switched networks (including the Internet), was developed to guarantee the reliable delivery of information in the proper sequence from sender to receiver. However, TCP's error and flow-control mechanisms may result in indeterminate delays and disrupt data delivery. This approach does not fit the needs of real-time CA, which requires a relatively tight delay characteristic.

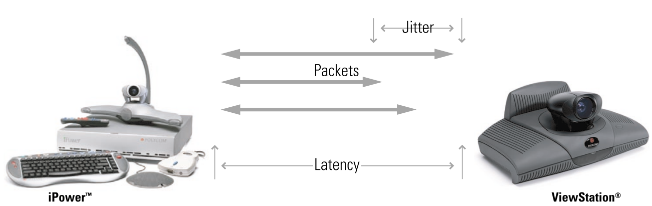

Figure 3: Jitter Diagram

Facts about packet loss as quantified by Polycom Labs

Network jitter can result in packet loss:

- A one-percent packet loss may produce blocky video and/or audio loss
- A two-percent packet loss may make video unusable, although audio may sound somewhat acceptable

While packet loss above two percent is unacceptable for H.323, one to two percent is considered poor and should be resolved.

Real-Time Protocol (RTP) and its adjunct Real-Time Control Protocol (RTCP) work alongside TCP to carry video media over the network. RTP uses packet headers that contain sequencing information and time stamps required to time the output (for example, display of frames) and synchronize different data streams (for example, audio and video) so that the remote end receives video media in the correct order.

Quality of Service (QoS)

CA presents the network with two significant challenges. The first is the additional bandwidth that is consumed during conferences. Although CA conferences can create a significant bandwidth demand, the good news is that this demand is both very predictable and is also manageable. Unlike data applications, video and audio have consistent bandwidth requirements. And unlike data applications, CA can be scheduled. This means that if a bandwidth limit (for conferencing) must be imposed, scheduling can insure that this resource limit is not exceeded. The more difficult challenge of IP-based CA is that it is real-time traffic employing the User Data-gram Protocol (UDP). Organizations that have implemented voice over IP (VoIP) will already be familiar with the demands of real-time traffic, but for those that have not, this is a new requirement. Real-time traffic asks the network to deliver its packets with very low loss rates and in a timely manner. Congested links that cause packet loss also cause data applications to slow down, but these same links will cause CA to fail. QoS techniques are often employed to insure that real-time traffic gets through on schedule. The network must be tested to insure that either sufficient bandwidth exists throughout the work day, or that QoS mechanisms provide sufficient priority for real-time traffic to insure a successful implementation.

iPriority™

Polycom's iPriority initiative consists of the Quality of Experience (QoE) features listed below. QoE is the combination of network based QoS (NQoS) and application-based QoS (AQoS). This combination provides for the highest quality user experience possible.

Network-Based QoS:

- IP Precedence: Packet marking for (QoS)
- Diffserv (DSCP): Packet marking for QoS
- RSVP: Packet marking for QoS

Application- Based QoS:

- Dynamic bandwidth allocation- for network congestion
- Packet and jitter control- for network congestion
- Asymmetric speed control– for dissimilar speeds of transmit and receive for example ASDL (384 Kbps up and 128 Kbps down)
- Fixed port firewall capabilities for simplifying deployments of video that traverses firewalls
- Network address translation (NAT) support- for security
- On screen diagnostics- for rapid problem resolution
- PVEC video error concealment- to conceal packet loss
- White noise insertion-for a more comfortable audio experience
- Packet commander for Polycom's MGC infrastructure products- QoE
- Intensive H.323 Inter-operability testing to preserve your investment in standards based environments
- Cisco verification to assure a compatible deployment of Polycom and Cisco technologies

Video Communication Bandwidth Basics

Video communication over IP can use more bandwidth than traditional applications. A typical business-quality call over IP requires the following bandwidth:

Audio (64 Kbps) + Video (320 Kbps) + IP

Overhead of approximately 20% = 460 Kbps for each call

Figure 4 illustrates that an IP video call made at 384 Kbps needs 20 percent more bandwidth to produce the same quality result as an ISDN call made at 384 Kbps.

Call Quality

The Internet does represent a quality concern to video-based communications due to the lack of QoS available on the open Internet. Some Internet service providers are currently offering Service Level Agreements (SLAs) that address latency and jitter issues; however no single provider can guarantee the quality of every communication session from all ISPs over today's end-to-end Internet. Internet conferencing does represent a way to connect with other IP domains outside of an individual organization. Cost advantages over traditional point-to-point connections are also attractive to designers. However, traversing through firewalls and NAT devices poses additional obstacles in using the Internet as a medium for passing video traffic today.

Firewalls

It should be noted that the problem of traversing firewalls with UDP and addressing issues associated with both H.323 and SIP are not manufacturing issues. Both the H.323 and SIP standards suffer from these same issues. Polycom is committed to working on these issues, and currently offers simple workarounds as well as full solutions via partnerships to overcome firewall and addressing issues. The following sections provide insight into the problem and make recommendations to overcome them.

What Is the Firewall Problem With H.323?

One of the reasons why firewalls are problematic is the heavy use of dynamically allocated ports within H.323. The dynamic nature of port assignment makes it nearly impossible to pre-configure firewalls to allow H.323-signaled traffic without opening up large numbers of ports in the firewall. As an example, Microsoft recommends configuring firewalls for use with NetMeeting, an H.323-based conferencing application as follows:

"To establish outbound NetMeeting [an H.323 application] connections through a firewall, the firewall must be configured to do the following:

- Pass through primary TCP connections on ports 389, 522, 1503, 1720, and 1731.
- Pass through secondary TCP and UDP connections on dynamically assigned ports (1024-65535)."

For more information, read the article How to Establish NetMeeting Connections Through a Firewall at support.microsoft.com/support/kb/articles/Q158/6/23.asp.

This represents a somewhat lax firewall policy than would be acceptable at many sites, but it still does not address the problem of receiving incoming calls. The other workaround for firewalls is to employ an H.323 application proxy, a software component of a UNIX or NT-based firewall that actually takes part in the protocol. In the H.323 context, a proxy would take part in the H.323 conversations, terminating the call on the firewall and creating a second call to the final destination, and finally plugboarding the two calls together. These steps may cause a delay in voice and video transmission that could disrupt the communication session. In addition, the enterprise firewall must handle all H.323 call setup and tear down work, as well as processing all of the video traffic for all of the endpoints attempting to communicate across the firewall. This presents significant security and performance/scalability challenges for H.323 and any data traffic traversing the enterprise firewall. H.323 support within the firewall in addition to the traditional role of firewalls in securing com-

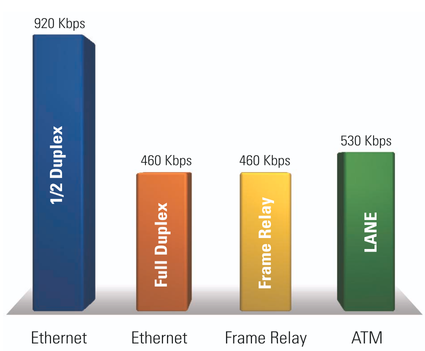

Figure 4: IP Bandwidth Over Different Transports

For greater detail please refer to the Step Three section of this document for Polycom's recommended transition plan. There you will find greater insights into how bandwidth calculations affect your deployment.

Figure 5: 384 Kbps ISDN Call Quality Over IP

mon protocols such as HTTP and FTP, makes firewall design more complex and (by definition) more vulnerable to attack. Also, H.323 support has the potential of degrading overall firewall performance and scalability.

Polycom recommends using proxies coupled with accelerated processing as is the case with market leading firewall products such as Check Point and PIX. In the event that your network contains firewalls from multiple vendors or for cost reasons upgrading or standardizing would be the antithesis of cost effectiveness, Polycom recommends third party proxy products such as Ridgeway Systems (http://www.ridgewaysystems.com/).

What Is the NAT Problem With H.323?

NAT is a method by which IP addresses are mapped from one IP domain to another, in an attempt to provide transparent routing to hosts, specifically from private non-routable addresses to publicly routable addresses. Traditionally, NAT devices are used to connect an isolated IP domain with private unregistered addresses to an external IP domain with globally unique registered addresses. NAT is generally used for two purposes:

1. As a mechanism to work around the problem of Internet Protocol version 4 (IPv4) address space depletion

2. For security purposes (to hide hosts at unroutable addresses)

NAT works by having a NAT device, often implemented as part of a firewall application, rewrite IP headers as packets pass through the NAT. The NAT device maintains a table of mappings between IP addresses and port numbers. The problem with NAT from an H.323 perspective is that H.225 and H.245 make heavy use of embedded IP addresses. If NAT is being used, addresses in the protocol stream are addresses in the private address space (behind the NAT device), rather than addresses at which hosts have established public, routable interfaces. For example, a host may have its address in a private address space, 172.18.0.51, which when traversing a NAT is translated to 207.126.235.233. When that host attempts to place a call, the calling party information element in the H.225 signaling stream contains the private, non-routable address 172.18.0.51 Attempts to make an H.225 connection back to that address will fail.

What Is the External (or Incoming) Call Resolution Problem With H.323?

Some vendors have begun to create firewalls with H.323-aware NAT, which allows outgoing calls (as described above) to function correctly. These firewalls translate the IP address in the H.323 signaling protocol stream as well as the IP address in data packets themselves. However, NAT cannot allow an incoming call from an external H.323 endpoint to an endpoint behind a NAT. If an external endpoint tries to call an internal endpoint using an internal endpoints' private IP address, the incoming call contains an unroutable address in two places and the call's packet is discarded by the first router it reaches. If the external user tries to use the public IP address of the NAT device, the NAT device has no way of knowing who the intended recipient is for the call.

IP Deployment Recommendations

A network that is optimized for real time applications over IP ensures quick, smooth transfer of data. Before deploying CA, upgrading your organization's network to meet the minimum IP-video requirements is strongly recommended. The next section details a plan to help define the essential application, network, and infrastructure requirements.

"During the design phase of a migration from ISDN to IP, consulting services are often required; customers need to know help is out there."

Geno Alissi, Vice President and General Manager, Polycom Global Services

Four-Part Transition Process

Polycom recommends a four-part transition process.

First, the best-practices step reviews the current state of the CN/CA environment, use and processes, and makes recommendations for changes that can be addressed during the transition.

Secondly, create an overall system architecture for IP-based UCN. This creates a coherent roadmap for the deployment, and insures the system will scale in size as needs change.

Third, review the demands that CA will put on the IP network, and insure that the network is ready. This insures that the network will properly support the expected CA load, and e existing mission-critical applications running on the IP network will not be adversely impacted by the introduction of CA.

Finally, deploy IP-based CN equipment, establish management processes, and train both support staff and end-users on how to use the new system. This transition requires a careful, staged plan to insure the technical infrastructure is ready, the administrative team is trained, management processes are in place and working, and end users know how to use the new equipment as it shows up in their conference rooms and offices.

Step One – Best Practices Review

A best practices review captures the current state of the CA deployment and use, and presents it to management in ways that quickly brings issues to light. The best practices review should include:

- An environmental scan to determine the specific equipment, revisions, software, and transports currently in use
- A cost review to determine the cost of CA services, including ISDN and POTS line charges, equipment, maintenance and outside services such as multipoint audio, video and Web conferencing
- A process review to determine how CA are currently managed, including scheduling, initiation, multipoint conferences, maintenance, upgrades, user support and troubleshooting
- A user review to determine how users view the service, their perceptions of how easy or hard it is to use, and their predilections for using it in future collaboration

The results of this effort should be presented in two ways. First, the data itself should be shown in a number of formats (tabular and graphic) to show the specific results. Second, a comparative report should be generated that shows your company's data compared to a peer-group of companies with similar sized networks. This latter comparison helps place the data in context, showing what the average peer-group values are, as well as the best in class.

This analysis can be used to determine specific changes that you would like to implement during the ISDN-to-IP transition, in order to improve those areas that need attention.

Benchmarking

Many companies conduct yearly surveys to users with questions on CA use and quality. Polycom recommends that such surveys contain questions relating to:

- Track equipment utilization
- Failed calls
- Interrupted calls
- Calls that start late due to start-up problems with ISDN based conferencing systems

Problems typically identified during such surveys include:

- Call failures due to ISDN drops
- People unfamiliar with system
- Users call minutes before a meeting for help on how to use the system or connect laptops
- Meeting rooms are booked back-to-back, so there is no time available to set up a call before a meeting begins

It is strongly recommended that a best practices review be conducted. Polycom's Global Services groups can assist you in conducting a complete best practices review. The output from the review can be used in presentations relating process changes and budgeting for the migration project.

Internal Support Process

Polycom recommends that customers establish a consistent, global strategy for supporting CA users. A three-tier strategy is envisioned:

- Tier 1 – Utilize your existing IT help desk services
- Tier 2 – Escalate to your in house CN/CA specialists and administrators
- Tier 3 – Escalate to external support vendor for maintenance and warranty work for example 24 hour turnaround break fix SLA

Escalation to external support is a growing trend occurring in the marketplace today. In particular, companies want to implement a global maintenance agreement so that any system worldwide can be maintained by one vendor, managed under one contract, with one point of contact. Polycom Global Services organization assists in this trend, while bolstering our value added partners service organizations, where required to bring the best possible services to your deployment.

Step Two – CA System Architecture

A CN systems architecture should be developed to encompass the full vision of your CN/CA deployment. All variables — from terminals to MGC and application server software — that can be envisioned in the foreseeable future must be considered. The goal of this architecture is to insure that the system can expand easily as your CA needs grow. The architecture prevents common mistakes like choosing a numbering plan that is not consistent with all of your expected locations, or not determining a management process that can insure efficient use of the CN resources.

Polycom recommends that a set of terminal deployment reference architectures be designed to fit the needs of small, medium, and large group deployments within your company. Each terminal's reference architecture should describe the terminal and corresponding CN infrastructure components/services that are required. These architectures should be considered as the standard supported by your company and the definitive recommendation for each implementation. These reference designs, when deployed, should tie back into the CN core architecture and provide full CA functionality. Network requirements (capacity) should be derived from terminal requirements and expected utilization.

Examples of reference architectures

1) CA User: Unified conferencing dial in number and Pin#, Web*Office*™; Web, audio, and video portal client/account.

2) CA Desktop: Web*Office* client with ViaVideo® appliance/camera, Utilizing MGC Unified Conferencing and Web conferencing, Scheduling, Management, and Call Processing Applications Server services.

3) CA Executive Desktop: iPower™ executive dual monitor system, Utilizing MGC Unified Conferencing and Web conferencing, Scheduling, Management, and Call Processing Applications Server services.

4) CA Small Group System: ViewStation® EX with a visual concert and a Premier speaker phone. Utilizing MGC Unified Conferencing and Scheduling, Management, and Call Processing Applications Server services.

5) CA Executive Conference Room: ViewStation® FX, Visual concert™, and a VTX™1000 Speaker phone, Utilizing MGC Unified Conferencing and Scheduling, Management, and Call Processing Applications Server services.

6) CA Remote Education Suite: iPower 3000 (iPower 9800 with a rear projection interactive whiteboard from Smart technologies), Utilizing MGC Unified Conferencing and Scheduling, Management, and Call Processing Applications Server services.

Capacity planning

A spreadsheet should be created to estimate the peak-hour demand on major inter-campus links of the IP network based on the terminals installed. This demand is used to determine if sufficient capacity exists in the network. Endpoint requirements drive the placement of MGC devices. MGCs provide conferencing bridge services as well as gateway services for dialing into/out to an ISDN network. The placement (location) of these devices should best fit your user calling style, IP network capacity, and management strategy.

Additional Items to Consider

I. A firewall strategy must be designed to insure calls can be made across firewalls without creating security holes in the network or degrading firewall performance. If encryption is required for calls, an encryption strategy should also be created.

II. A number and naming plan must be devised that makes calling between endpoints as simple as possible, and enables consistency with existing naming and numbering systems such as e-mail addresses and current telephone extensions.

III. A scheduling system should be proposed that allows users to easily reserve conference rooms, MGC bridge ports and bandwidth to support collaborative conferencing. Polycom's scheduling systems are designed to integrate with existing user tools such as Microsoft® Outlook® or Web browsers.

IV. A management system should be specified. It is responsible for tracking terminal status, capabilities and software versions. Polycom's Management systems provide statistics to the CN/CA management team on utilization, call set-up success rates, conference interruptions, and the quality of conference connections.

IV. Management processes should be developed during the transition plan to make use of this information on an ongoing basis to monitor and manage the quality of the end user experience.

VI. CN Core Design. As described in Figure 1, CNs reside on top of Layer 1-7 IP packet switching infrastructure. Polycom optimizes CN/CA to leverage the best of modern routing and switching protocols and architectures. As previously discussed, a Polycom CN is a three tiered architecture; consider your IP network as the fourth tier. One practical way to design a CN is to associate the CN core with major geographic service regions. Simply complete your capacity plan first; this highlights the traffic patterns that clearly suggest locations for hosting CN core technologies. Core technologies specifically refer to Layers 2 and 3 of the CN architecture model.

- CN Core Infrastructure Requirements consists of both media processing (MGC) and server software (scheduling, call processing, Web conferencing, and management). CN cores are typically attached to geographic regions.

Dialing Plan Requirements

Each terminal in the system requires an ISDN (E.164) number and an alias. Users dialing from ISDN-based systems must use the ISDN number, because they have no other dialing capability. IP-based system users can dial using an alias name instead, which is often easier and more intuitive.

A dialing/naming plan must be created to assign an ISDN number and alias to each endpoint. The dialing plan should have enough numbering space to accommodate future expansion of terminals, and be logically consistent with existing E.164 numbering systems.

The alias naming convention should also be logical, so that users can easily determine the name of the endpoint they wish to call. For more information on deploying dial plans with The Polycom Office please refer to the [6]Deploying The Polycom Office white paper.

Scheduling and Reservation Requirements

The scheduling system should be tightly integrated into your existing enterprise scheduling and calendaring system. This provides users the ability to intuitively schedule conference rooms, MCU ports, and bandwidth in the same manner as scheduling people. Configurations of the scheduling, gatekeeper and management systems can insure resources are available for scheduled conferences. Network Aware Scheduler, a Polycom patent pending technology, is available only from Polycom. Because of the amount of manpower required to manually set up each meeting, automation via scheduling and ad hoc services are must-have service capabilities. Unified conferencing is schedulable or ad hoc. Network awareness ensures that CA and your network are optimized for each other. Please refer to the [7]Polycom Management Solutions white paper for greater detail on network awareness and Polycom server applications.

Gatekeeper Issues

Gatekeepers work in either routed or direct mode. Polycom recommends using routed mode, where all control traffic flows through the gatekeeper, so that advanced features like call forwarding, conference on demand, and alternate routing can be implemented.

Rogue Detection

Rouge detection is very important to managing a CN. Rogue systems are terminals that are purchased and installed by users without working through the corporate plan or structure, and are thus not registered with the gatekeeper. PathNavigator™, Polycom's Call Processing Server and Gatekeeper can recognize and report rogue systems when they open a call with a registered endpoint.

ISDN Connections

Gatekeepers determine how a call is routed through the IP network, and also out into the ISDN network if that is required. There are a number of scenarios where using the ISDN network is advantageous.

Clearly if an endpoint is available only on an ISDN connection, the call must and will be routed through a gateway onto the PSTN for that connection. The gatekeeper can be configured to optimize the use of gateways based on demand, based on least cost routing (toll bypass) if desired.

Application Reliability

If the IP network between two geographic areas is experiencing congestion, and available CA bandwidth is already consumed, a call can either be dropped (a busy signal), or it can be routed through the ISDN network. This feature may be used to support peak demand without denying the call. Monitoring the utilization of ISDN to bypass the existing IP network can indicate when it is time to invest in additional IP bandwidth.

If the IP network is experiencing problems with its current traffic load due to partial link failure or some other issue, the gatekeeper can be reconfigured to reduce the number of CA calls using that link. For the duration of the network problem, calls can be routed via ISDN to continue to provide service.

[6] www.polycom.com/common/pw_item_show_doc/0,1449,1780,00.pdf
[7] www.polycom.com/common/pw_item_show_doc/0,1449,1911,00.pdf

Step Three

Two key concerns are raised whenever a substantial new application, such as CA, is to be implemented on an existing IP network. The first concern is whether the network has sufficient capacity to support the new application. Is there enough bandwidth between company locations to support the additional load created by conferencing? The second concern is the impact of this new load on applications currently using the IP infrastructure. Will the introduction of CA slow down mission-critical applications and make them hard to use or even render them inoperable? This step of the planning process answers these questions.

Network Review

Demand

Create a demand spreadsheet that predicts the bandwidth required between each geographic location at the peak hours of utilization. This spreadsheet is created by determining the number of endpoints, calling patterns expected from those endpoints (times, length and to what location), and call bandwidth.

By reviewing the existing network a determination can be made as to its ability to support IP-based CA. The review includes WAN link bandwidths, router devices, router OS levels and QoS capabilities in routers and switches. Endpoint connections are examined to determine if endpoints have dedicated switched connections or are on shared media.

The second step of the network review is to determine current utilization levels on key WAN links during the busiest hours. This step helps determine the impact on existing applications of CA's introduction. Key business applications and the approximate number of users are identified for modeling purposes.

A good first step to accomplish this task is a network review and test that reviews and tests a subset of the network environment. This review is intended to accomplish two goals

1. Test a designated portion of the network

2. Expose network engineers to the process so they can determine the best way to proceed with the rest of the network validation

Network Modeling

The information collected in these steps is combined with predicted demand to determine the network's conferencing capacity. A model of key inter-campus links must be built to determine CA's impact on existing data applications. The results of this stage show the trade-off between network upgrades if they are required, and the number of simultaneous CA sessions that can be achieved. At this stage of the process a plan can be formulated on how to proceed with some combination of network upgrades and planned introduction of IP-based CA to different sites.

A second key output of the network review is to create an SLA between the CA and network support teams. This SLA defines required bandwidth, packet loss and jitter requirements for the network between each of the major sites where CA is required. The SLA is a key document for helping the two organizations sort out where problems exist when they arise, either with CA equipment or the network that is carrying the traffic. The SLA lets the networking group understand the requirements of CA traffic in terms they understand. Conversely the CA group can test the network to insure they are getting the transport they requested, and understand if problems they are experiencing are due to equipment or transport.

Network Verification

Once any network upgrades have been accomplished, or it has been determined that the network already has sufficient capability, the network must be tested with synthetic traffic. Test tools must be introduced to the network to create traffic that simulates the CA load. This traffic is measured against SLA requirements for bandwidth, packet loss and jitter. Issues discovered during this phase are reviewed with the networking group. The problems are isolated and a plan put in place to correct any issues. This testing phase often finds issues that are otherwise overlooked, and insures that the network is clean and operational when the CA equipment is first installed.

Network Verification Tools:

- **Net IQ's Chariot™ Product**
  www.netiq.com/products/chr/default.asp
- **H.323 Beacon tool**
  www.itecohio.org/beacon

Step Four

A transition plan must be created that moves the organization step-by-step from the current deployment to the desired CA deployment. This plan should take into account the organization's goals and issues discovered during the best practices, architecture and network testing phases. The transition plan spells out which portions of the enterprise implement IP-based conferencing first, and to what level. It coordinates network upgrades needed to support conferencing, insure network testing is complete and schedule delivery and installation of conferencing equipment. Training sessions should be planned to insure users are familiar with use of the equipment and the new methods of call setup and operation.

Additional steps should be specified to insure that existing ISDN-based units continue to operate, and will interoperate with IP-based conferencing as it is deployed. Business units that depend on conferencing are supported throughout the transition to insure there is no interruption of service.

Also, during the transition, training is provided for network and CA personnel on setup, configuration and use of the new equipment.

Recommended Approach

Polycom recommends that the transition plan contain a number of phases to insure that system issues are resolved before the majority of users encounter an IP-based conferencing environment. The value of successful initial interactions with CA cannot be overemphasized. Early acceptance of new technology leads to increased use by a wider employee population.

I. Phase I of the conversion consists of a pilot or test setup using laboratories in geographically dispersed portions of your network. Each lab should set up the minimum components of the core network architecture, as well as sufficient terminals to test out the system's capabilities. Both labs must have ISDN capability to verify the IP-to-ISDN calling features.

II. Phase II of the conversion identifies sites on a number of different campuses where early adopter users are available to help shepherd the new technology. Choose enough different geographic sites to test out QoS, bandwidth, scheduling and management strategies defined in the architecture. These sites, once they are brought on-line, can shake down the infrastructure to insure that all features are functional, the management and scheduling processes work, and they blend well with the enterprise's business flow. If modifications to these processes are made, it causes far less disruption to correct them within a small deployment than change them after a large base of users is trained on the wrong methodology.

III. Phase III is the deployment of additional end-points and the conversion of some end-points from ISDN-to IP-based systems. At this point, additional divisions or groups can join into the core network system by implementing one of the reference architectures.

IV. A strong education component is recommended as a part of the transition plan. During Phase II, training focuses on infrastructure, those users who manage conferencing environment, the day-to-day running of the equipment, solving issues as they come up and collecting data for management review. During Phase III, training focuses on end-users, insuring they have both the training and the documentation to allow successful use of conferencing environment.

Test Plans

Test plans are required for the Phase I pilot testing and the Phase II infrastructure testing. This section addresses pilot testing in Phase I.

Schedule

Phase I testing should take less than 90 days, as follows:

- 30 days to obtain equipment, labs and connections for testing, and write the test plan
- 30 days to accomplish lab-based testing, at which time the equipment is connected to your operational network
- 30 days to test on your operational network

Primary goals of pilot testing are to insure all functions work as expected within your network environment, and to create a representative environment so that problems can be found and addressed before a larger deployment is created. A few examples of services that should be tested include:

- Least-cost routing
- Bandwidth utilization
  - Reliability and fail-over

Business Case

Hard Costs

All costs must be considered as a part of the business case justification for an ISDN-to-IP conversion. Some of them are considered hard costs, while others are soft, meaning that it is more difficult to obtain a well-defined value for them, as subjective judgment is involved.

A second source of hard costs is the fact that each conference room using ISDN requires ISDN lines directly to that room, incurring a monthly charge even when the lines are not used. Because IP -connectivity is relatively ubiquitous and costs much less per connection, eliminating dedicated ISDN lines leads to substantial hard savings.

The cost of moves, adds, and changes is next on the list. Moving a conference room from one location to another, or installing a new CA system, is much less expensive when the transport is IP. This is because IP is usually available in both rooms already, and if it is not, it is very inexpensive to add. ISDN on the other hand, requires expert installation, and incurs Phone Company and monthly charges as described above. For these reasons, IP transport provides flexibility and lower cost.

Viewed on a conference-by-conference basis, IP management resources should be lower than those required to support an ISDN network. These resource reductions result from a more unified network topology, better management tools and lower inherent failure rates.

New CA systems using only IP transport are priced lower than systems that have an ISDN connection or both IP and ISDN connections. This is another hard cost savings.

Soft Costs
Soft costs are often harder to pin down, but conversely can have a much larger financial impact. The flexibility of being able to reconfigure conference rooms or offices very quickly and with minimal cost could mean much higher productivity for a team during a critical development or work flow period. The value of increased productivity may come in an earlier or cleaner product introduction, both of which have substantial financial benefits.

Other soft costs are reduced travel, increased productivity due to the use of video or use adjunct features like data sharing, and increased video use due to the ease and reliability of the IP transport method.

Rolling It Up
Per our previous discussion on TVO, Polycom recommends creating a spreadsheet that estimates both hard and soft costs, combining them to determine the ROI or payback period. Soft cost values can be reduced in this spreadsheet by a percentage factor, to indicate that these costs are soft and therefore more difficult to quantify. Thus if productivity leads to a $5,000 cost advantage, this value can be reduced by 50 percent and counted as only $2,500 in the cost sheet.

When reviewing this analysis with management, make sure they understand the components of the cost benefit analysis. If management chooses to zero out the soft costs, they can do so, but they can still see the potential benefits of those soft returns in the analysis helping to justify the case for an ISDN-to-IP transition.

Summary
Many companies are migrating their collaboration and communication applications to IP networks. This trend goes along with the application convergence occurring in many corporate, education, medical, and government networks.

> **“… in seven to ten years, video traffic on the Net will exceed data and voice traffic combined.”**
>
> *Bob Metcalfe,*
> *Founder of 3COM*
> *Feb 3, 2003*
> *Forbes Magazine*

The rate of H.320 technology deployment is predicted to slow when juxtaposed against the scale of IP CA deployments over the next three to five years.

Unifying collaborative application under a single architecture and manufacturer, and deploying with a proven process is the best path to rapidly gaining the advantage of collaborative processes within your business. We recommend selecting a Polycom solution that not only leads the market but also leads in standards and technologies that constitute true, Unified CA and CN. Partnering with Polycom brings the most business value and TVO to your deployment.

POLYCOM®

www.polycom.com

Polycom, Inc.: 4750 Willow Road, Pleasanton, CA 94588 (T) 1.800.POLYCOM (765.9266) for North America only.
For North America, Latin America and Caribbean (T) +1.925.924.6000, (F) +1.925.924.6100

Polycom EMEA: 270 Bath Road, Slough, Berkshire SL1 4DX, (T) +44 (0)1753 723000, (F) +44 (0)1753 723010

Polycom Hong Kong Ltd.: Polycom Hong Kong Ltd., Rm 1101 MassMutual Tower, 38 Gloucester Road, Wanchai, Hong Kong, (T) +852.2861.3113, (F)+852.2866.8028

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