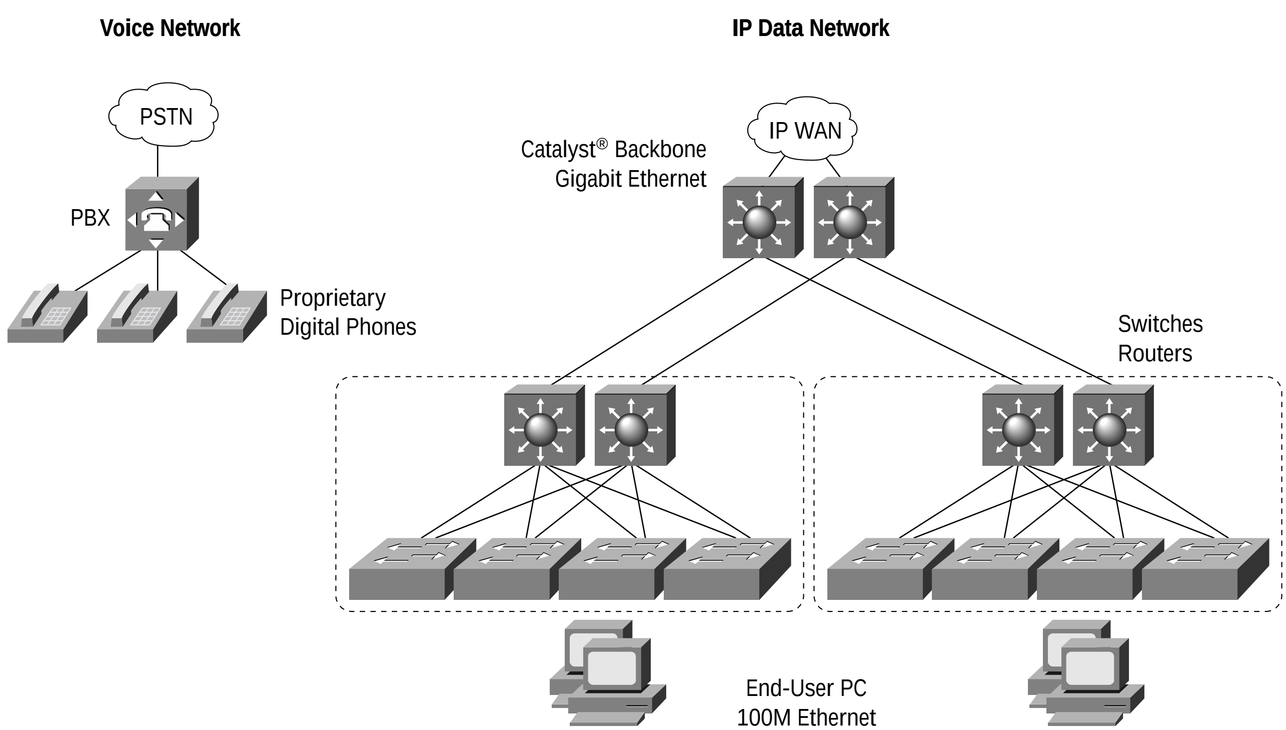
# Technical Considerations for Converging Data, Voice, and Video Networks

## Introduction—Getting Started

In networking today, corporations are looking for strategies to integrate disparate networking technologies over a single common network infrastructure. This trend started many years ago in the internetworking area when corporate networks began the migration of delay-sensitive, mission-critical Systems Network Architecture data traffic across an IP infrastructure. Now, people are examining their existing and often separate data, voice, and video network infrastructures and determining the most efficient ways of bringing these together.

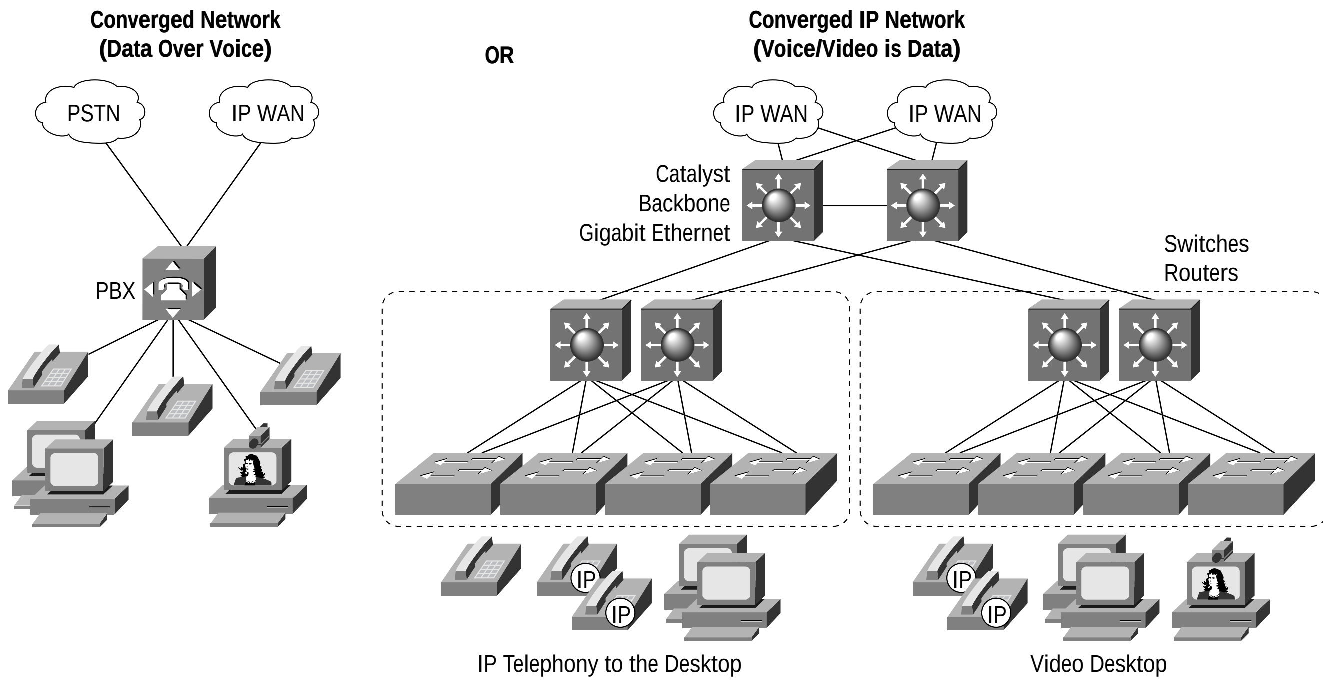
In today's environment, most corporations have voice networks anchored around private branch exchange (PBX) systems, while most data networks are IP-based and anchored around switches and routers. Figure 1 illustrates this concept, where traditional telephony in fixed 64-kbps bandwidth increments is carried across a circuit-switched PBX infrastructure while other dynamic data applications are consolidated over an intelligent IP-based backbone.

Figure 1 Today's Typical Network Infrastructures



With today's separate infrastructures, two strategic directions toward the convergence of data, voice, and video are possible. These approaches are illustrated in Figure 2.

Figure 2 Strategies for Convergence




One approach calls for the use of the PBX to support the kinds of interfaces and bandwidth required by today's IP-based data applications. This would mean interfaces such as Ethernet, Fast Ethernet, Gigabit Ethernet, and others on PBX platforms. Most data desktops are now connected at speeds of 10 or 100 Mbps, making this a strategy without merit. Fundamentally, the PBX is a time-division multiplexed system built on 64-kbps time slots with aggregate capacity in the tens of megabits per second. Although this is adequate for voice only, any attempt to drive data bandwidths shows that a whole new breed of platform would be required beyond the PBX.

It is also interesting to look back in history at other attempts to accomplish this integration. In the 1980s, when a 2400-bps modem was standard and a 9600-bps modem was almost unheard of, ISDN at 64 kbps per channel was being touted as the next high-speed desktop technology. Naturally this was well-suited to be terminated on the PBX, given its conformance to the bandwidth hierarchy of the PBX systems. Interestingly, this still did not take the marketplace by storm, even before technologies such as Ethernet were on the radar screen. Vendor investment in traditional PBX technology has not scaled to meet the demands of a converged network.

Most of the networking innovation during the last ten years has been developed around IP networks because of the pervasiveness and rapid development of the Internet. This includes World Wide Web technology, streaming video, IP telephony, higher-speed IP transport mechanisms, and more intelligence in the network to cope with new classes of delay-sensitive, mission-critical applications. This has all lead to the future of integrated networks—one where data, voice, and video share a common IP data network. Naturally, things will need to be done to prepare for this eventuality. The first step is to understand the architecture that will make this convergence a reality and some of the more specific considerations in the data networking infrastructure that will make this migration easier.

### The Cisco AVVID Architecture

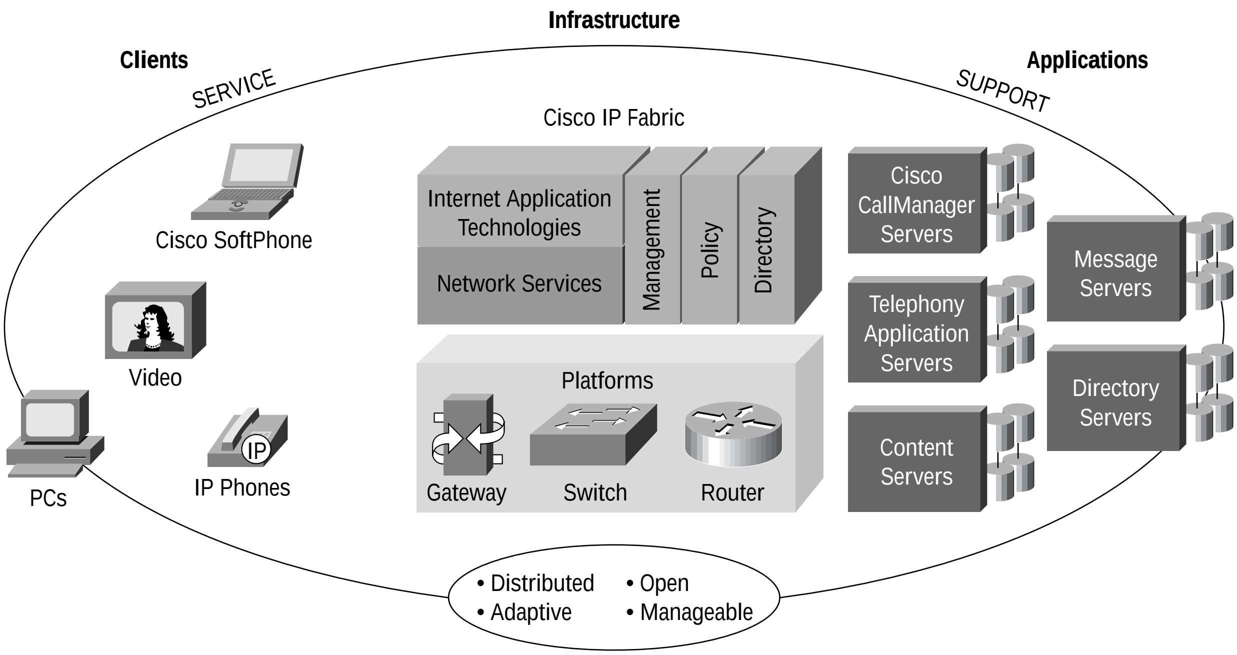
The Cisco solution for the convergence of data, voice, and video, shown in Figure 3, is called Cisco AVVID (Architecture for Voice, Video and Integrated Data). Cisco AVVID has three fundamental components:

- Clients are the user end stations or appliances that are used to communicate with the network or other users. Some examples are PCs, telephone devices, and video cameras.

- Applications for Cisco AVVID are written to an open-standards environment and, as such, will be supplied both by Cisco and third-party application developers. Some examples are interactive voice response (IVR), call center, and unified messaging.
- Infrastructure is the network upon which both the clients and applications reside. The network is IP-based using intelligence inherent in the platforms to provide the flexibility and scalability to support the convergence of different media. Examples of network devices are Catalyst switches, Cisco routers, and voice gateways.

The remainder of this paper explores this intelligent infrastructure and guides the network manager in preparing for the opportunities that lie ahead in the creation of a Cisco AVVID network.

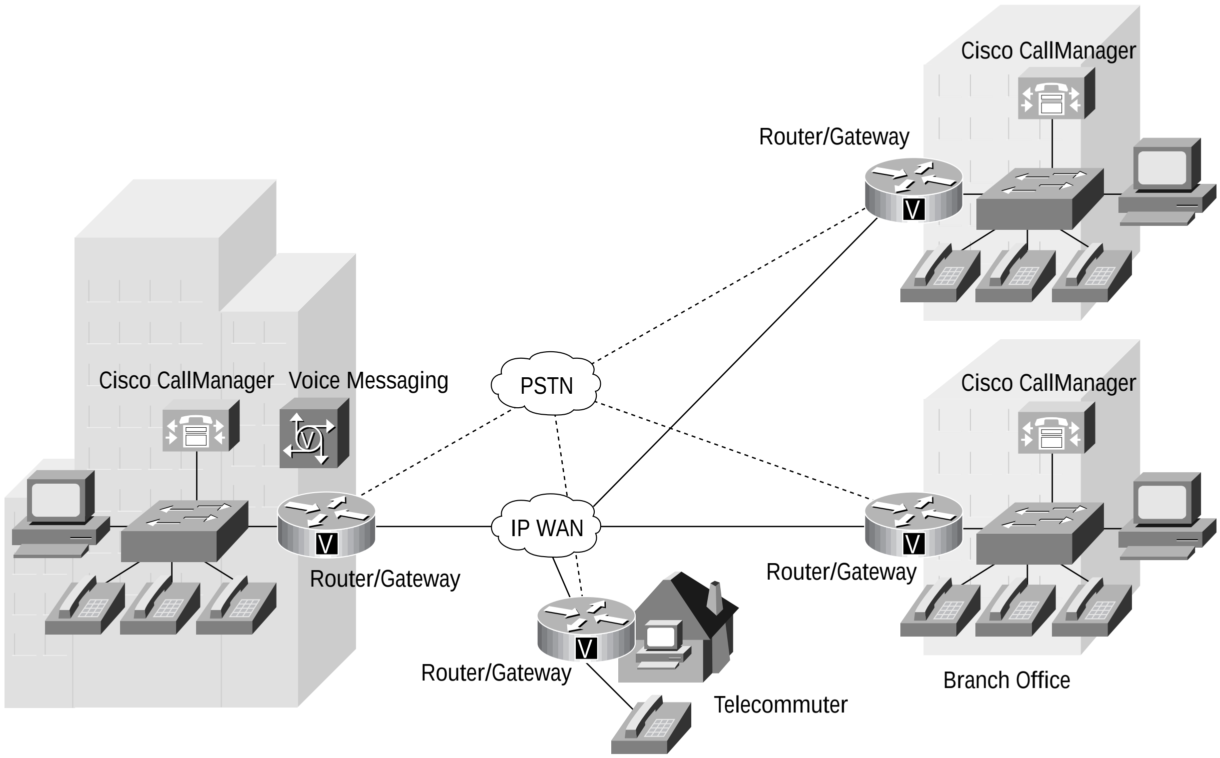
Figure 3 The Cisco AVVID Architecture




**Typical Cisco AVVID Network Deployments**

Figure 4 illustrates a typical deployment of a converged network combining data, voice, and video over a single infrastructure. The network comprises a single or several large headquarters sites, a larger number of branch offices, and individual telecommuter sites. From a telephony point of view, the telephone devices, whether a physical handset or a soft-phone application on a PC, are IP endpoints on an IP infrastructure typically serving Ethernet to the end user. Voice calls inside a site traverse the IP local-area network (LAN) until the exit point from the site where, as a first choice, it would cross the IP wide-area network (WAN) backbone to save in public network toll costs. If for some reason this IP WAN is "full" or unavailable, the call would automatically be rerouted across the Public Switched Telephone Network (PSTN). Either way, the path of the call is transparent to the end user. Gateways are used at the edge of each site to provide a path to the PSTN, which is also the primary path for local inbound and outbound calls. This level of integration allows for very simple coordinated dial plans where an entire enterprise can use four- or five-digit dialing, as appropriate, for internal calls.

Figure 4 End-to-End Converged Network




The technology used in the IP WAN allows for tremendous flexibility in the design of the intranet. Because the IP protocol is the underlying carrier of data, voice, and video, any technology such as Point-to-Point Protocol (PPP), Frame Relay, Asynchronous Transfer Mode (ATM), dark fiber, digital subscriber line (DSL), Synchronous Optical Network (SONET), and wave-division multiplexing (WDM) may be used to construct the network. This allows for maximum flexibility and agility on the part of a network manager to take advantage of new high-speed WAN services from carriers, without making major changes to the converged IP network. However, routers at the edge of the WAN must employ the proper quality of service (QoS) techniques to ensure that voice traffic remains high quality, even in congested conditions. Techniques such as Low-Latency Queuing, Weighted Fair Queuing, and Priority Queuing are examples of necessary technologies in a converged network.

Familiar applications such as voice messaging are also easily integrated into this converged networking environment. Since the voice being carried across the network is encoded in IP, a new breed of IP-enabled unified-messaging solutions allows users to access both voice mail and e-mail from a common server. This can be accomplished with either the e-mail application or through a next-generation IP telephony handset.

While the nirvana of a converged network has widespread appeal to many, the reality is that every corporation already has existing voice and data networks, a scenario that requires finding simple ways to migrate to a converged network.

The next section of this paper describes how to accomplish converged networking using Cisco AVVID solutions. For simplicity, the solutions are divided into the following three sections:

- Large-campus site
- Wide-area network considerations
- Legacy migration

**Campus Considerations**

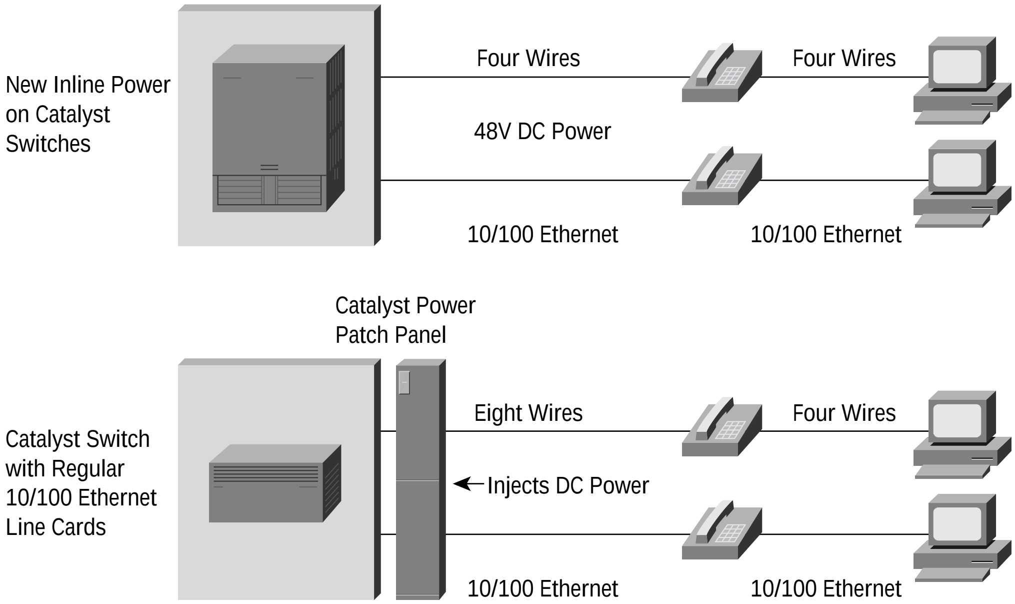
In the converged network where telephony handsets are IP-attached Ethernet devices, the following new considerations have emerged:

- Use of a single wiring closet connection for both phone and PC

- Choices in the application of power to the phone
- Assignment of IP addresses without wholesale renumbering
- Ease of adds, moves, and changes
- Guarantee of voice quality, even under congested conditions
- Overall redundancy and reliability of the network

The typical interconnection of a desktop in a converged network is shown in Figure 5. While there are many ways of doing this, including separate ports for voice and data, the illustration shows a single-wire connection that supports both the telephone set and desktop PC at 10 or 100 Mbps over copper Ethernet. This configuration is possible due to integrated 10/100 switch ports on a Cisco IP phone. On the port of the phone that connects to the IP network switch, 48 volts of power can be delivered in two ways. The first method utilizes new inline power 10/100 Ethernet switch ports on some Cisco Catalyst switches that automatically detect the presence of the phone and apply power on the same four wires that carry the Ethernet signals. The second method works with any existing switching platform and uses a Catalyst switch-powered patch panel that inserts power on the other four wires of the eight-wire bundle to the desktop. Also (not shown) is a third option to use an AC power adapter and plug the phone into a wall socket. The method chosen to power the phone set is a function of the amount of copper wire to the desktop, the type of switching equipment installed in the wiring closet, and the overall power backup strategy of the corporation.

Figure 5 Connectivity of IP Telephones




Increasingly, the data network is the mission-critical part of the business that needs to be protected during power failures. In this case, method 1 of inline power lends itself to achieving this by having a backup uninterruptible power supply (UPS) in the wiring closets and gaining the benefit of backing up both the voice and data networks.

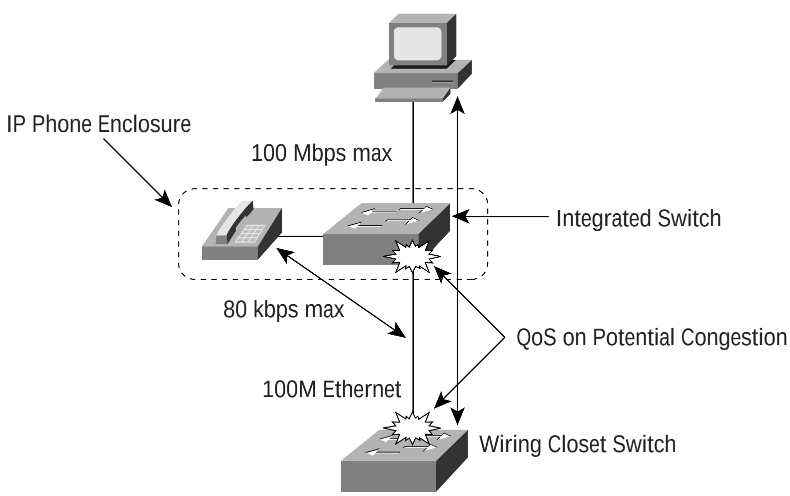
IP addressing can be a concern when adding new IP devices to the network. Elaborate IP addressing schemes have often been created to accommodate a certain number of IP endpoints in a network design. This typically results in a network with multiple IP subnets of sizes in the 128 or 256 range. Irrespective of what subnet size was chosen, adding new IP phone devices into the same subnet would cause an approximate doubling of the number of hosts in a subnet. The result could be a complete renumbering of all existing devices on the network. Fortunately, the ability to incrementally add IP phones onto a new set of IP address ranges while keeping the existing IP address structure intact has been built into the Cisco AVVID architecture. This is done by using standards-based 802.1q virtual LAN (VLAN) technology on the phone and the Catalyst switches so the phone can be easily and automatically addressed with a new address without impacting the current scheme. This is another tool of Cisco AVVID that allows for ease of migration into the converged networking world.

All the functions of IP address assignment, the application of power, and the use of the Dynamic Host Configuration Protocol (DHCP) for addressing the phones allow for another extremely powerful feature: the ease of adding, moving, and changing a phone. Gone now are the days of having to keep complex spreadsheets to keep track of people moving or paying for the programming of a PBX for a simple move. With the Cisco AVVID architecture, all that is required to move a user is to unplug the phone set and plug it into the new segment elsewhere in the building or campus. The user maintains his/her original phone number and features at his/her new location, with no intervention by the networking staff. Also for additions of new phone sets, it is possible to take a "fresh" phone out of the box, plug it into the Cisco AVVID network, and have a new number assigned to it from within a previously defined pool. All this happens with no manual programming to the voice system. These features alone can justify a converged networking infrastructure for some customers.

The QoS issue can be summarized by one simple principle: If you are running both voice and data on a common network where the voice traffic is steady and delay sensitive relative to the data traffic, proper tools are required to ensure that the voice traffic satisfies both its loss and its delay parameters. This applies both in the campus, where the links are high speed, and in the wide area, where every bit of bandwidth matters.

Figure 6 shows an example in the wiring closet where QoS could be used in a campus environment to ensure safe passage of voice.

Figure 6 QoS for Voice Traffic in the Wiring Closet



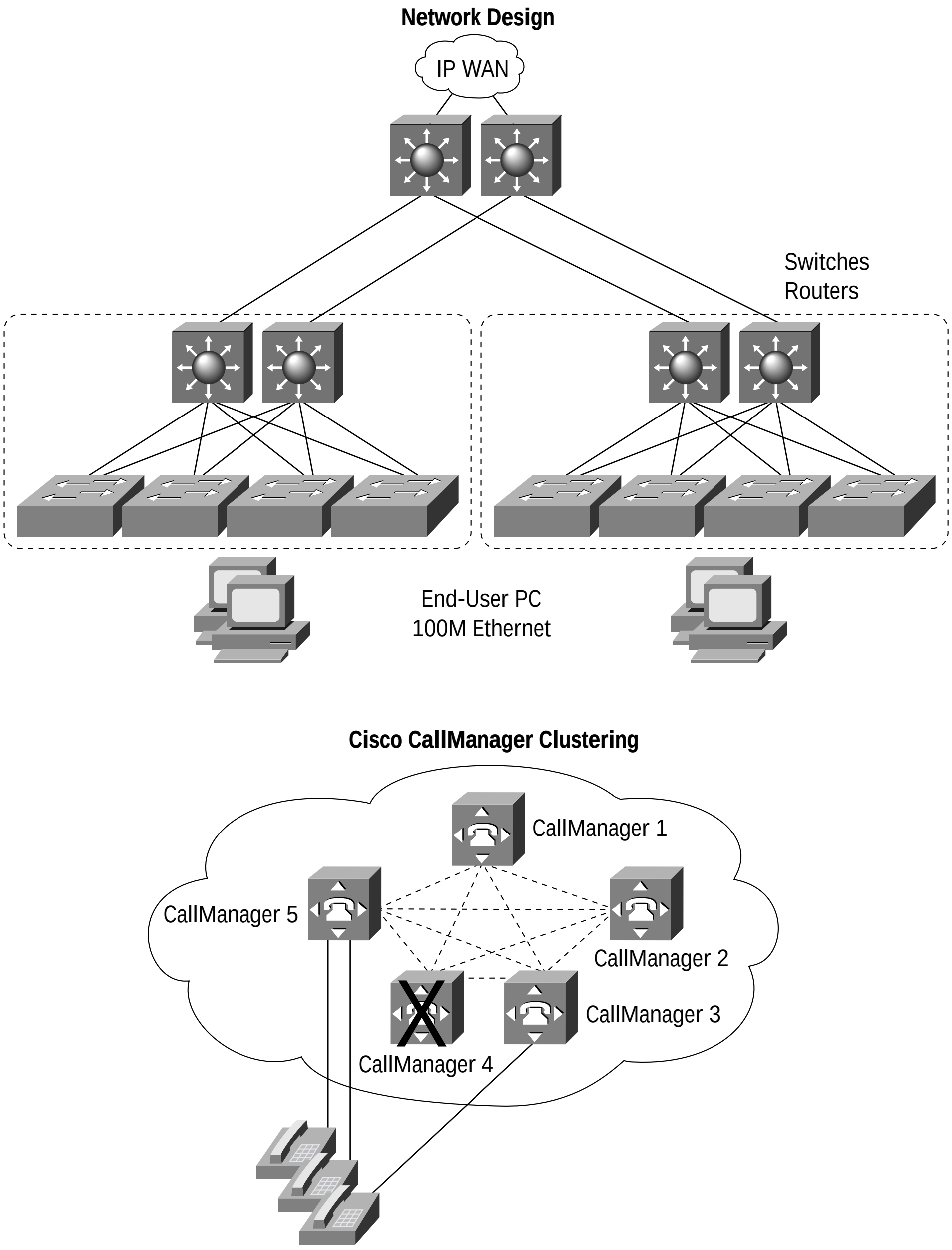

First consider traffic from the phone and PC arriving at the integrated voice switch. The bandwidth of the PC traffic could overrun the 80-kbps (64-kbps uncompressed voice plus IP overhead) voice stream being generated by the phone. For this reason, the integrated Ethernet switch in the phone has queuing mechanisms built in to ensure that the 80-kbps voice stream will pass through unimpeded while any dropped traffic will occur on the data stream that is tolerant to such events. Conversely, for traffic leaving the wiring-closet switch port bound for the phone and PC, a potential congestion point could be on the Catalyst switch port. Again, a number of Catalyst switching platforms have the appropriate queuing mechanisms to guarantee safe passage of the voice traffic.

Equally as important as queuing, however, is a QoS mechanism called classification. Later, as will be illustrated in the WAN section, it is important to differentiate at all points in the network between voice and data traffic so that queuing techniques can be applied in places such as the WAN edge. Classification therefore "marks" voice and data frames differently so that they can be treated differently by the other switches and routers that make up the intelligent IP network. Two standards-based methods are utilized. At Layer 2, 802.1p defines classes of service (CoSs) pertaining to that Ethernet segment. At Layer 3, the IP layer, the IP type of service (ToS) field can carry different values end to end through the network.

The ability to change the Layer 2 and Layer 3 classification parameters to values desired by the network designer is a function inherent in the Cisco AVVID architecture and available in the IP phone integrated switch and some Catalyst switching platforms. The ability of the network manager to manipulate these values at different points in the network defines a parameter called the "trust boundary." For example, if you trust that the end stations and phones will set their QoS values correctly so that end-to-end QoS can be maintained, the trust boundary includes the end stations. If, however, you cannot guarantee that all end stations behave properly and set their own priority values correctly, then the ability to properly manipulate these parameters is available in the Catalyst switch infrastructure.

Finally, in the campus, the requirement for a robust, fault-tolerant infrastructure is inherent in any converged network. The first thing to consider is the design of the network itself. Cisco offers many tools in its switches and routers that enable network designers to build truly fault-tolerant, fast, converging networks. While the details of how to do this are beyond the scope of this paper, an example of what this might look like can be see in Figure 7 on the top part of the diagram. Most data-only networks today already employ the intelligent features available in Cisco switches and routers to build a redundant architecture to support their mission-critical data applications.

Figure 7 Highly Available Networking Techniques



The other high-availability feature of Cisco AVVID is the ability to provision multiple Cisco CallManagers in a cluster so failure of one Cisco CallManager can be easily tolerated by the telephony end stations to provide even higher overall system availability. The distributed nature of IP networking lends itself to such a scheme as do highly available Web commerce sites.

#### Wide-Area Network Considerations

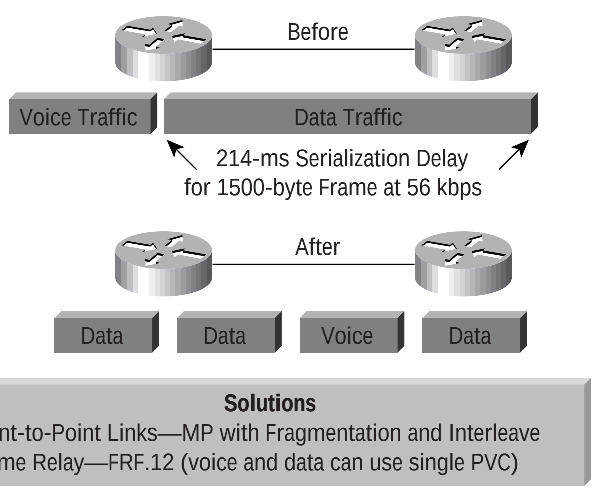
In the wide-area network, numerous key infrastructure considerations work together to provide a seamless, efficient wide-area transport of data, voice, and video. They include:

- Quality of service
  - Queuing
  - Slow link efficiency tools
  - Traffic shaping
- Admission control (protecting voice from voice)
- Compression of voice and IP headers
  - Digital signal processing "farms"

The need for QoS in the wide area is mandatory in a converged network. As discussed in the campus section, any time there is a potential congestion point in the network, proper queuing techniques need to be applied so that delay- and drop-sensitive traffic such as voice and real-time video pass through unimpeded relative to the drop-tolerant data applications. This is typical at the WAN edge router, where hundreds of megabits of potential traffic are aggregated into slower-speed links in the kilobits or low megabits-per-second range. It is beyond the scope of this paper to detail all the mechanisms available, other than to say that these mechanisms are inherent in the Cisco intelligent network and support all wide-area interface types such as PPP, Frame Relay, ATM, Packet over SONET, and many others. Whatever the queuing mechanism used, it likely will take advantage of the classification of data, voice, and video packets that could have been achieved earlier in the campus wiring closets.

Special consideration needs to be placed on low-speed links. Even if all the voice traffic has been sent out an interface and it is now time for a data packet to get sent, the data packet could possibly be in the order of 1500 bytes long. That size data packet would take more than 200 milliseconds to get clocked out a slow-speed interface. This may not sound like a lot, but the overall delay budget for a one-way voice or video conversation is in the order of 150 milliseconds. Fortunately, for both point-to-point and Frame Relay networks, Cisco routers employ standards-based fragmentation techniques, as illustrated in Figure 8.

Figure 8 Techniques for Dealing with Slow-Speed WAN Links



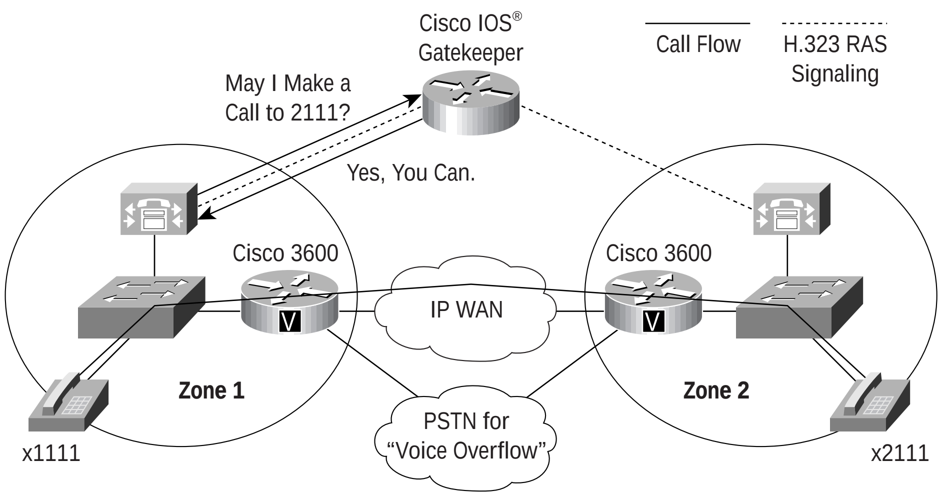
These link fragmentation tools are used on links typically less than 768 kbps, where the delay of large data packets can be problematic. By fragmenting the data packets into sizes that are equivalent to a voice packet, the delay incurred by the voice frame becomes insignificant. At the other end of the link, the data fragments are reassembled by the router and passed on as if nothing happened. On point-to-point links, Multilink PPP (MPPP) is used (similar to that on multiple ISDN lines to aggregate two 64-kb links into a virtual 128-kb stream). On Frame Relay networks, the Frame Relay Forum specification FRF.12 is used.

Many WAN networks are deployed using Frame Relay and, in this case, one more function is required to satisfy the QoS demands of a converged network. With a high-speed connected central site feeding many low-speed remote sites, the central site could overrun both the physical bandwidth and the committed-information-rate (CIR) bandwidth of the remote site. To prevent this, a feature called traffic shaping is used on the central-site router; traffic shaping effectively prevents too much bandwidth from being sent in the first place to any of the remote sites.

Another type of QoS called admission control is required in the wide-area network. This effectively prevents too much voice or video (which typically have fixed bandwidth requirements) from being sent down a wide-area link that does not have enough bandwidth to handle it. In the case of voice, for example, if there is enough bandwidth to carry five voice calls between locations A and B, and a sixth call is placed, this call should not be admitted across the link because the quality of all calls could suffer, regardless of the QoS techniques used. If this call is not admitted, it would be automatically rerouted via some other path, such as the PSTN links that are available between sites.

As Figure 9 illustrates, Cisco CallManager interacts with an H.323 gatekeeper, which is programmed with the policy regarding call admission across the WAN. In this case, there was enough bandwidth for the call to cross the WAN, so the call was sent across the IP wide-area network. If the gatekeeper has answered “no,” the Cisco CallManager could append the appropriate digits in the dial plan and send the call to the gateway, which in turn would route the call across the PSTN to the destination. Whether the call traverses the IP WAN or the PSTN is completely transparent to the end user. This call-admission-control capability is inherent in the Cisco AVVID architecture.

Figure 9 Admission Control in the Wide-Area Network




The other key to successful wide-area networking of voice traffic is compression. Two types of compression can be used to make more efficient use of the WAN bandwidth. The first type is compression of the voice itself. This takes place in the endpoints (telephone sets) and termination points (gateways) of a voice call. Examples of standard voice compression include:

| | | |
|---|---|---|
| G.711 | 64-kbps | Uncompressed voice payload |
| G.729a | 8-kbps | Compressed voice payload |
| G.723.1 | 5.3 – 6.3-kbps | Compressed voice payload |

Another reality of carrying Voice over IP (VoIP) is the IP overhead added to these payloads. This overhead needs to be taken into account when calculating how much bandwidth a link can carry. The IP headers add about 16kb/s to each of the streams above including the IP, the User Diagram Protocol (UDP), and the Routing Update Protocol (RTP) headers. This, for example, would cause an 8kb/s G.729 a stream to use 24kb/s of bandwidth.

A tool called Compressed Real-Time Transfer Protocol (CRTP) available on Cisco routers compresses the IP headers on slow-speed links so the 24-kbps G.729a stream takes up only 12 kbps. Thus, CRTP can shed 12 kbps of the 16-kbps overhead of the IP stream. On slow-speed links, this translates directly into monetary savings. The use of this feature should be accounted for in the overall engineering of the network.

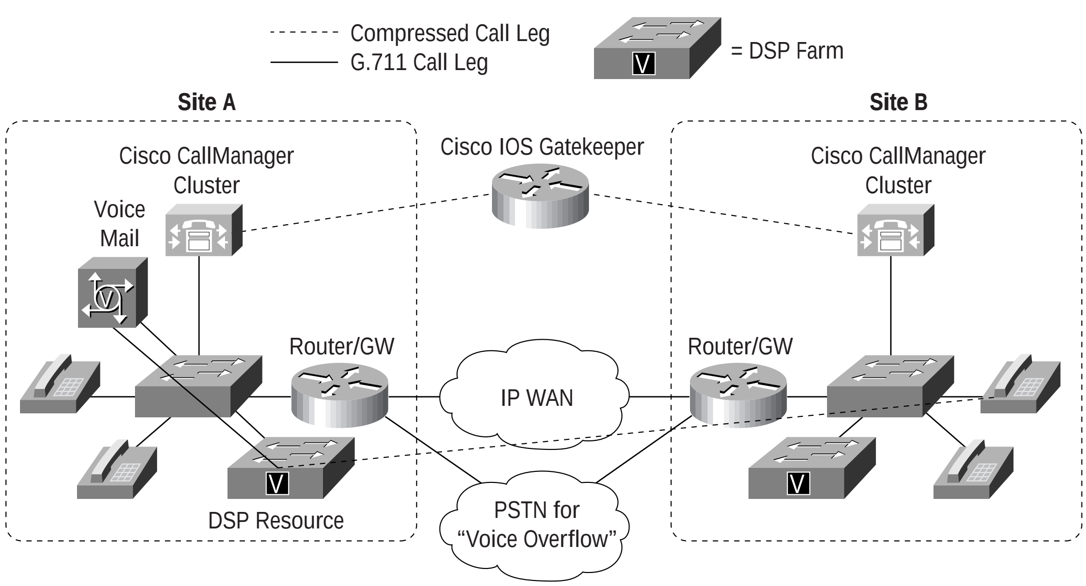
When you combine the benefits of voice compression in the wide area with admission control and examine end-to-end convergence solutions, another piece of the Cisco AVVID puzzle emerges in the form of digital signal processing (DSP) farms.

Again, with reference to Figure 9, suppose that the dial plan was set up in the network in such a way that compressed calls at G.729 crossed the WAN using 24 kbps of bandwidth. Once the call is allowed to cross the WAN by the gatekeeper, it is possible for many reasons that there will be no more bandwidth to admit any more calls. If the call is successful and terminates on the destination phone, all is well. But in the more likely case that it gets transferred to voice mail (or some other service such as an Interactive Voice Response [IVR]), the coder/decoder (codec) would have to change to G.711 (80 kbps with IP overhead) because most voice-mail devices support only this standard. If this happens, the end device could oversubscribe the WAN bandwidth allocated for voice and destroy the quality of all calls on the link.

For this reason, the call flow is altered somewhat to account for this and other scenarios.

In Figure 10, a user at Site B calls a person at Site A but the call is forwarded to a voice-mail system that supports uncompressed voice. It is important to keep the leg of the call across the WAN in an unaltered, compressed state for admission control purposes. Therefore, the call will terminate on a DSP resource, which turns the compressed voice stream into uncompressed voice for the voice-mail application. This DSP transcoding function and call-control logic is integral to the Cisco AVVID solution and critical to maintain the benefits of a converged network that offers voice compression in the WAN.

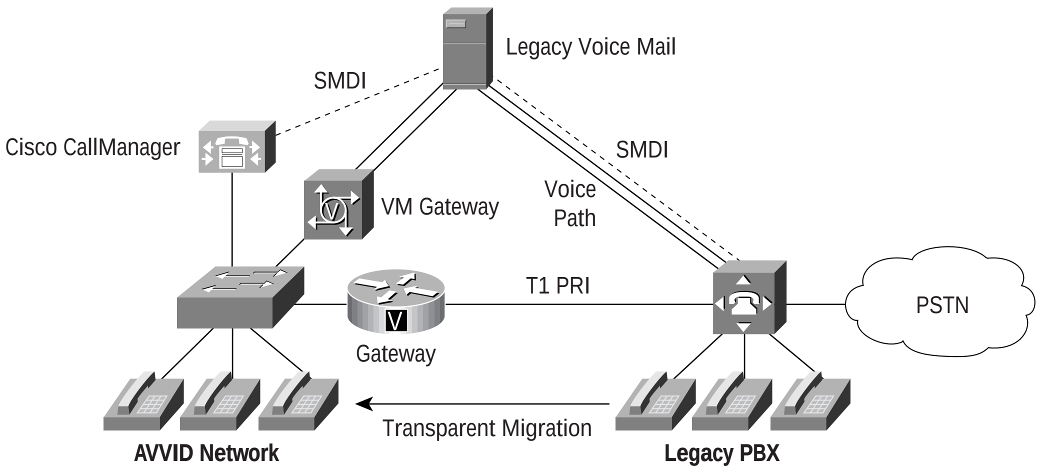
Figure 10 Using DSP Farms for Scalable WAN Networking



### Legacy Migration

Although building a converged network with Cisco AVVID has obvious advantages, some existing legacy voice and video systems require a seamless migration into the New World. One example might be a scenario where an existing PBX has reached capacity and its growth is dependent on IP telephony. In some cases the existing voice-mail system still has the capacity to sustain the additional users and it is desired maintain the existing system in production. Figure 11 illustrates one way of achieving this seamless migration.

Figure 11 Migration of Legacy Voice Network with Cisco AVVID




Here, a voice-mail system is connected to the PBX using multiple analog voice ports to carry voice traffic and a signaling line called a Simple Messaging Desktop Interface (SMDI) that passes information, such as whose mailbox greeting to play on which particular voice port, to the voice-mail system. This is one of many ways to network voice mail to a PBX.

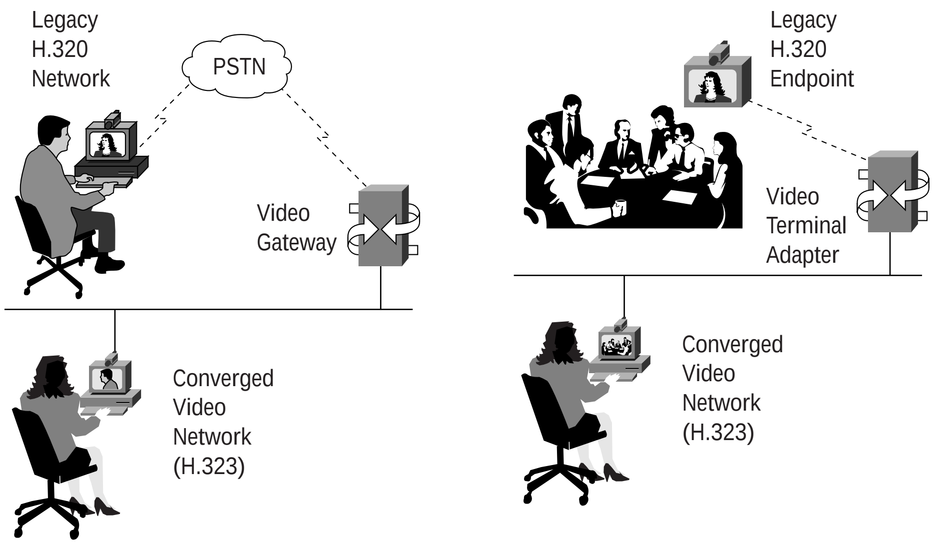
The first part of the migration to consider is establishing uniform dialing plans that are the same on both the legacy and Cisco AVVID networks. This is easily achieved using either a standalone gateway or one of the many router/gateway combinations available from Cisco. Regardless of which side of the network you are on, the incoming and outgoing calls from the outside world appear exactly the same (whether using direct inward dial or using an operator and extensions). Internal dialing using a shorter dial plan (perhaps four or five digits) can also be maintained.

From a voice-mail perspective, Cisco CallManager also supports SMDI for control of a voice-mail system. Coupled with analog gateways connecting to the same voice mail, calls on either side of the network appear exactly the same to voice mail.

In the video world, it is common to have H.320 based videoconferencing systems, each with their own PSTN network connections. LAN-based H.323 video systems also are commonly deployed in many companies, such as Microsoft NetMeeting, which utilizes H.323 for videoconferencing and shared collaboration. Again, the Cisco AVVID solution provides several different ways to preserve investment in legacy systems while investing in converged network solutions.

Two different scenarios are shown in Figure 12.

Figure 12    Migration from Legacy Video Networks




On the left side, there may be a whole network of H.320 systems deployed by your corporation or customers. Using a Cisco video gateway enables communication between the converged H.323 network and the legacy video network. It would also allow customers or vendors who have only H.320 systems to communicate with a new network of H.323-only systems. Additionally, on the right side, individual H.320 endpoints can be networked seamlessly in an H.323 network with the use of a Cisco video terminal adapter.

## Conclusions

The fact that every major networking vendor has announced plans for IP telephony solutions leads to the conclusion that the convergence of data, voice, and video networks is inevitable. The only possible convergence solution is to merge the existing voice and video networks onto the IP data network. For this to succeed, many aspects of the intelligence of the IP data network need to be exercised.

In the campus, elements of a converged network solution include:

- Tools that allow for ease of IP addressing of new devices without having to renumber
- Quality of service
- Power to the phone over the existing cable
- Setup that allows both phone and PC to use the same cable pairs at 10- or 100-Mbit Ethernet
- Simple ways to move and add new devices

In the wide-area network, a number of different queuing techniques—regardless of the link speed and technology, admission control, compression, and the use of DSP farms—all allow for converged networking technology to scale into the wide area.

Lastly, tools that aid in migration from legacy voice and video networks toward networks converged on IP make it possible for mass deployment of networks based on the Cisco AVVID architecture.

### Service and Support

Cisco AVVID support solutions are designed for one purpose—to ensure customer success by delivering a suite of proactive services. The award-winning Cisco internetworking service and support offerings provide presales network audit planning, design consulting, network implementation, operational support, and network optimization. Cisco interactive knowledge-transfer solutions enhance customer success by leveraging Cisco expertise and experience. By including service and support when purchasing Cisco AVVID products, customers can confidently deploy AVVID networks utilizing Cisco expertise, experience, and resources.

Now is the time to prepare your IP networks with the intelligent infrastructure to support the convergence of data, voice, and video.

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