Optimizing Video Performance Across the Distributed Enterprise
High-Quality Videoconferencing to Any User, On Any Network, Anywhere

Do more with less: that’s the message echoing through corporate cultures today. One clear target is to contain costs, especially travel-related costs, which can be large enough to have a serious impact on bottom lines. But it’s more than big dollars for airfare, hotels and meals; the loss of productive time and energy can be just as costly.

Given our present economy, it’s easy to see why videoconferencing attracts attention. It puts distant people face to face without leaving their normal work environments. They stay on-site and productive. Companies have reported saving as much as $120,000 USD in travel expenses on a single meeting – not to mention the benefit of maximizing productivity. But before they can realize these benefits, enterprises need to fully understand the best way to leverage an investment in real-time video.

Video Deployment Challenges

The primary challenges in deploying a videoconferencing solution are its bandwidth requirements and real-time characteristics.

Do I have enough bandwidth?

It is important to understand how much bandwidth videoconferencing applications will require across your network and especially across your Wide Area Network (WAN). Available WAN bandwidth is less than in the Local Area Network (LAN), yet is shared by many applications. Videoconferencing requirements can range from 1-5Mbps for desktop and room-based systems, and up to the 5-8Mbps for immersive telepresence solutions. Fully understanding what your deployment requirements are relative to available WAN bandwidth is crucial to ensuring high quality video delivery.

Despite the presence of IT sanctioned modes of videoconferencing there is also a marked increase in the use of non-sanctioned, unmanaged video through applications like Skype, Yahoo Messenger, and MSN. Anyone with a Webcam can use this feed even when unsupported by IT. From the broadest view, the bandwidth available for videoconferencing is whatever is left after business applications, file transfers, e-mail, back-hauled Web traffic, and other applications consume the bandwidth they need.
The time-sensitive nature of video
Live two-way video is a real-time conversation between participating parties and can experience quality degradation if video applications encounter latency, jitter, and packet loss across the WAN infrastructure. This can result in broken images and garbled audio that can impact meeting effectiveness, decrease productivity, and even cost the enterprise as decision-makers are forced to travel to complete the meeting.

Quickly identifying the current status of these quality-affecting metrics is crucial to understanding if conditions on your WAN are changing in a way that could affect video quality. However, tracking this information in real-time can often prove challenging especially if you need this information only for specific video flows.

Is throwing bandwidth at the problem the best solution?
It’s reasonable to think that simply adding bandwidth will give you the videoconferencing performance you want. And at times, you may need to increase the size of the WAN link to support videoconferencing traffic. But without a clear understanding of what’s happening on the WAN link, adding bandwidth may simply increase your costs and enable people to increase recreational traffic over the WAN – by accessing more YouTube videos as example – without giving you the results you need. Here’s what you need to know to make decisions about videoconferencing bandwidth requirements:

» How much WAN bandwidth do I really need to deliver a high-quality videoconferencing experience?

Begin by estimating how many concurrent videoconferences you need to be support, how long the calls will be, and which codec you’ll select. Your videoconferencing vendor can help by working with you to identify deployment sites, estimate the number of users and the typical length of conversations, and choose the technology that best suits your needs.

» What applications are being delivered across the WAN?

Include all categories: sanctioned and unsanctioned, delay sensitive and non-delay sensitive, business-critical and recreational.
Do I have the budget to get a leased line into each of my sites or can I only afford to interconnect my major sites via WAN links?

With budgets being tight, the distributed enterprise has the option of leveraging existing WAN links to deliver live video. But if you use this option, you'll be back to the bandwidth question again – do I have enough bandwidth, especially since video will share that bandwidth with other applications?

How is each of the applications on my network behaving?

For example, how much bandwidth are they consuming? Does the rate vary at specific times? Could backups be timed during low-traffic periods? Applications today are more sophisticated and savvy, using dynamic ports and even sharing ports. This makes it difficult to know which applications are on your network by port or IP address information; a time consuming and potentially inconclusive way to investigate the problem. Without the ability to distinguish between SAP, web surfing, and recreational online video on Port 80, existing network solutions can't separate mission-critical traffic from non-critical traffic – it all looks the same.

The reality is that most network managers do not have full visibility into their WAN traffic and are often surprised to learn that over 50 percent of bandwidth is consumed by recreational applications such as P2P, web surfing, social networking, IM and video downloads. This lack of application-layer visibility is a major obstacle to ensuring the highest quality experience for your users.

Before you consider buying more bandwidth and risk increasing costs without improving performance, you need to make sure that your WAN is fully optimized for videoconferencing by understanding, monitoring, and controlling application traffic delivered across the distributed enterprise.

**Blue Coat: A Smarter, More Flexible Approach to Optimized Video Conferencing**

Blue Coat understands that in order to optimize your videoconferencing solution you need end-to-end application visibility, management, and control. By building on the ability to identify traffic at the application layer, you can monitor application performance proactively, optimize the WAN efficiently, and resolve problems quickly to ensure a tight service level agreement (SLA) and a high-quality user experience.
Blue Coat does this by:

1. Giving you a clear understanding of what applications are going across your WAN
2. Providing you application-layer monitoring that allows you to track application performance and bandwidth usage over time
3. Providing you with robust policy controls that allow you to:
   - Protect real-time applications such as video from other bandwidth-hungry applications
   - Optimize the bandwidth usage of other business-critical traffic while controlling recreational traffic
   - Proactively set alerts to potential issues and assist in troubleshooting problems

**Application control demands deeper visibility**

Blue Coat Application Performance Monitoring technology, built around Blue Coat PacketShaper, enables the delivery of high-quality videoconferencing by providing Layer 7+ visibility. This lets you automatically discover all application traffic on your network (over 600 applications) including custom applications. Layer 7+ visibility clearly distinguishes between critical business applications, email, FTP file transfers, Web surfing, recreational applications, and more. PacketShaper can even identify sub-applications and applications that use the same or dynamic TCP/UDP ports. And PacketShaper supports granular application classifications and detailed metrics on the user experience.

Blue Coat’s Application Performance Monitoring solution can discover, monitor and control videoconferencing protocol suites such as H.323 and SIP (session initiation protocol), giving you a complete view of all video-related applications and protocols.

**Monitoring the metrics that matter**

Three video-impacting performance metrics need to be understood when deploying videoconferencing:

1. **Latency** – the end-to-end delay in delivering the video/voice stream from the presenter to the audience
2. **Jitter** – the unpredictable, variable delays in the delivery of each videoconferencing packet
3. **Packet loss** – the dropping of individual packets because of network congestion

Each can cause significant degradation in conferencing quality and overall system reliability.
Latency
Videoconferencing includes two typical modes: one-way presentation and two-way interactive communication. Two-way interactive communication is sensitive to delays in the network. Although conferencing quality is still considered acceptable when delay reaches 300ms, users see an obvious lag, and have to resort to one-at-a-time, walkie-talkie-style conferencing to communicate.

In addition to the voice stream itself, latency must also be addressed with videoconferencing protocols (SIP, H.323, etc.) that handle the call control functions between two systems. In fact, these signaling protocols are often more sensitive to delays in the network than video or voice packets.

Jitter
Jitter causes irregularities in the flow and delivery of data. This can be disruptive to a real-time application like videoconferencing. Some videoconferencing vendors have tried to solve this problem by introducing their own jitter buffers or queues to smooth out the delivery of voice packets. Routers also offer queuing mechanisms for this purpose. Both of these options, however, can exacerbate the problem by contributing to delays. Even with jitter buffering technology, the tolerance for videoconferencing is 100ms. If jitter causes delays to exceed this limit, especially on a consistent basis, conferencing quality will suffer.

Packet Loss
Because IP is a “best effort” protocol, if left unattended it will always be subject to unpredictable performance, including packet loss. Like jitter and latency, packet loss can be very disruptive to videoconferencing performance. It can become a serious problem at the LAN/WAN boundary, where the smaller pipe results in significantly greater contention for bandwidth.

Packet loss of one percent or less is within the bounds of toll quality video. When it reaches three percent, the audience will notice the conversation breaking up. Unless this problem is controlled, packet loss can lead to dropped calls and the possibility of videoconferencing system failure.

To sum up: your application control is only as good as your visibility into the applications running on your network and your ability to monitor the key
performance metrics described above. Blue Coat Layer 7+ visibility gives you the most powerful application discovery and monitoring available, and lets you control all network traffic including videoconferencing.

**Intelligent Bandwidth Control**

The ability to leverage intelligent policies to control bandwidth is the next key component of Blue Coat’s Application Performance Monitoring technology. Compared to legacy QoS solutions, intelligent policy control is:

- **Smart** – automatically discovers and distinguishes between applications
- **Granular** – controls per application, per call and per flow
- **Powerful** – controls inbound traffic to avoid congestion
- **Simple** – facilitates use with on-box policy manager and centralized management

**Per-flow dynamic control**

Routers typically have basic, static QoS features, which are unable to handle the demands of dynamic videoconferencing traffic. Most commonly, they ensure the performance of critical applications by assigning dedicated bandwidth through traffic prioritization schemes. This works for some static applications, but videoconferencing is highly dynamic and the number of concurrent media flows is uncertain. It becomes a challenge to ensure that enough bandwidth is reserved, especially if there’s a lot of high-priority traffic.

Blue Coat Application Performance Monitoring offers guaranteed videoconferencing application performance over WAN links by employing per-flow control technologies capable of minimizing IP congestion and avoiding packet loss. Our intelligent policies provide per-flow protection where bandwidth is assigned to videoconferencing according to the number of media flows, and each is guaranteed bandwidth to ensure video quality without any being wasted. This allows you to convert the IP network from “best effort” to predictable.

**Inbound rate control**

Maintaining quality for mobile videoconferencing is a challenge for most QoS solutions. All Web surfing, remote access and mobile videoconferencing traffic crowds the Internet link on the inbound connection. Traditional QoS (such as Router QoS) only enforces the point behind the Internet link, causing congestion to occur before the router can shape the traffic.
In contrast, Blue Coat uses TCP rate control to alleviate the root cause of congestion: the speed of incoming traffic. With Application Performance Monitoring, Blue Coat goes beyond existing QoS solutions to extend high-quality videoconferencing to the network edge (teleworkers).

**Adaptive Policy Capabilities**
Our adaptive policy capabilities provide the ability to adjust policy dynamically to underlying traffic conditions.

Example: Videoconferencing, SAP and non-critical traffic running on a 16Mbps WAN link; the administrator predefines multiple policies for different conditions:

- When no videoconferencing traffic is detected, assign 10Mbps bandwidth to SAP.
- When videoconferencing is running and bandwidth is less than 5Mbps, bandwidth for SAP is reduced automatically from 10Mbps to 5Mbps.
- Once videoconferencing traffic is greater than 10Mbps, all non-critical applications are blocked automatically to reserve bandwidth for SAP.

All policy adjustments are dynamic and automatic. Once policy is set, the network adapts to all traffic conditions in a pre-defined manner, eliminating the need for manual intervention.

**Matching WAN bandwidth to videoconference requirements**
Most videoconferencing systems include a multipoint controlling unit (MCU). The location of the MCU decides the routes of video traffic in the enterprise WAN. PacketShaper identifies all the traffic between the MCU and videoconferencing terminals, and helps IT organizations design their WAN links connecting the headquarters and branches according to the real bandwidth required for any particular videoconference.

With the focus on increasing collaboration among employees and customers, videoconferencing is often used in conjunction with interactive applications, including whiteboard, collaboration software and even file-sharing capabilities. Bandwidth for interactive applications are more random and various than video and voice traffic. PacketShaper detects T.120 and file-sharing automatically, and provides committed application performance by assigning bandwidth dynamically.
Saving bandwidth with compression and acceleration

Gaining visibility into WAN application traffic and applying intelligent policy control are two critical steps to ensuring best-in-class application delivery. A third essential step is to apply compression to specific traffic types. Videoconferencing is compressed by the codec (e.g. H.261, H.263, H.264, G.722, G.728, G.729) that delivers video and voice packets across the WAN. Although additional compression of this traffic is not advisable because it will compromise conferencing quality, there are opportunities to apply compression to data applications such as email and ERP and to various types of Web traffic to save bandwidth and reduce WAN congestion.

Acceleration reduces the data on the WAN link for Web, bulk data (file transfers, email, etc.) and video streaming applications, resulting in an improved user experience. PacketShaper provides real-time compression for a 2 to 4x gain in data application capacity, optimizing data transmission. The Application Performance Monitoring solution also supports a direct-to-net architecture that removes web surfing and recreational traffic from expensive WAN links and enables you to avoid WAN bandwidth scaling.

Ease of Ongoing Management

Real-time monitoring to detect potential problems

Performance issues can arise at the worst possible times (during a meeting with a customer, for example) and are virtually unpredictable. Even if you have people monitoring the health of the entire network 24/7, users may already be impacted by the time a problem has become critical. Blue Coat’s real-time monitoring makes it easier to detect performance issues as soon as they occur. Once the performance issue is detected, an email is sent to administrators and an SNMP trap is sent to the NMS.

Real-time monitoring works as a meter to measure the quality of the network and the videoconference application. It helps the IT department proactively find potential performance issues, contributing to reduced helpdesk calls.

Troubleshooting performance issues quickly

PacketShaper provides the IT department with over 120 measurable statistics per class along with many powerful diagnostics tools for isolating issues and recovering performance quickly. This is essential to maintaining a network that is ready for video.
A Final Word on the Inadequacy of Router QoS

We mentioned earlier that router QoS lacks Layer 7 visibility, intelligent control, inbound rate control and videoconferencing metrics, which are all provided by Blue Coat Application Performance Monitoring.

Using router QoS, your enterprise cannot identify critical applications, recreational traffic or video traffic. Routers provide only a limited means of controlling traffic types and aligning them to the enterprise business. They cannot assign bandwidth dynamically, so bandwidth is wasted. Moreover, routers cannot prevent inbound traffic congestion from impacting remote videoconferencing, making it difficult to extend videoconferencing services to small branches and remote teleworkers.

Routers also lack the capability to track user experience and help IT troubleshoot performance issues in a proactive way. Finally, routers don’t have acceleration features. This means that to deploy additional applications you must scale network bandwidth, and cause your connectivity costs to increase. Router QoS cannot deliver the low-cost, high-quality videoconferencing you need.

In Conclusion: High-Performance Video Meets Instant ROI

Blue Coat Application Performance Monitoring provides immediate ROI and ensures the enterprise will maximize its investment in videoconferencing. With the ability of PacketShaper to control all types of applications, including real-time UDP applications such as video, the enterprise is guaranteed the high performance and availability necessary for videoconferencing to be a realistic alternative to in-person meetings.

Blue Coat is the global leader in application delivery networks that enable enterprises worldwide to effectively align their video, voice and data applications and network resources with their business priorities while realizing tangible cost savings. Blue Coat’s value is delivered through a family of intelligent appliances built with patented software technology that provides unmatched visibility, acceleration, and security. Specifically for videoconferencing, Blue Coat addresses chronic congestion, jitter, and packet loss that afflict video quality at the LAN/WAN boundary. Blue Coat effectively manages all critical videoconferencing protocols and ensures the highest quality end-to-end communication.