



Is Your Network Ready for VoIP?



Tough Questions, Honest Answers

For many years, voice over IP (VoIP) has held the promise of enabling the next generation of voice communications within the enterprise. Unfortunately, its adoption has been slowed; sometimes by the reality of poor performing IP-PBX systems, in other cases by perceptions that VoIP technology is not quite ready for prime time. These problems — both real and imagined — have forced IT organizations and VoIP vendors to take a closer look at the technology and ensure that IP-PBX solutions deliver the features, quality, and reliability that customers demand.

With a new generation of mature, reliable solutions, VoIP is once again building market momentum. As vendors and customers implement strategies to move from a traditional TDM-based circuit-switched PBX to a converged IP voice and data network, an increasing number of enterprise organizations are trying to understand how their existing network can handle the convergence of voice and data traffic without compromising performance and reliability. This paper summarizes in question and answer format the many critical factors pertaining to VoIP's viability, regardless of which vendor's IP-PBX system is deployed across the distributed enterprise.

I am considering migrating our circuit-switched PBX to an IP-PBX system, then running my company's voice communications across our existing data network. How can I be assured of voice quality and reliability for all users?

There are many variables that can impact VoIP quality and reliability, including (but not limited to) the IP-PBX system used for voice communication, the networking equipment used to carry voice and data traffic, the amount of bandwidth available to all sites and users, and the total amount of traffic moving across the network. VoIP quality is not typically an issue within a corporate LAN because bandwidth tends to be plentiful. The most significant point of congestion and potential compromise to voice quality is at the LAN/WAN boundary. Here voice (and data) traffic must be carefully controlled as it leaves the LAN and transitions to a WAN link that is far more bandwidth constrained. Blue Coat is focused on this congestion point with application delivery solutions that enable the distributed enterprise to ensure optimal performance of voice and data applications across the WAN.



If my carrier offers an MPLS service across its backbone network, won't that eliminate the traffic congestion problem?

It will certainly help, but the benefits of MPLS do not always extend to the LAN/WAN boundary at the edge of the enterprise. Carriers try to manage to the edge of the network using conventional routers, but they are limited in their ability to monitor application traffic and apply appropriate traffic controls within the local network. As a result, this is a major congestion point for critical voice and data traffic, which must be shaped to assure optimal end to end call performance. Fortunately, Blue Coat is able to prepare traffic for MPLS even at the network edge by applying the appropriate DiffServ markings to voice and data application traffic, either on its own or in conjunction with edge routers. This capability is an important feature of Blue Coat's application delivery technology, which combines visibility, acceleration, and security to minimize voice and data traffic congestion, maintain business productivity, and the satisfaction of service level (SLA) objectives.

How much bandwidth is required for optimal voice quality across the WAN?

First, you must determine how many VoIP users you will be supporting, as well as their average call length and call volume. Your VoIP system vendor can help you make this determination. In addition, the codec you select (discussed later in this paper) will influence bandwidth requirements. You can use Blue Coat's application discovery and monitoring technologies to gain visibility into how bandwidth is being used by your data applications, so you can segregate essential from non-essential application traffic. Once you have detailed information on current bandwidth usage and projected voice traffic, you will be better prepared to conduct accurate capacity planning.

If it looks as though my WAN links will be too small to support the addition of VoIP traffic, should I immediately upgrade my links to prevent performance problems?

It is possible that you may need to increase the size of your WAN links to support VoIP traffic, but it is not advisable to increase bandwidth without first adopting a sound application delivery strategy. Why? Because IP allows users to consume all available bandwidth regardless of the link speed — much like traffic tends to rapidly fill new lanes on a freeway. Simply adding more bandwidth without proper bandwidth management and control may only result in adding more traffic — rather than improving the quality of VoIP service.



So how do I begin to determine if my network is ready for VoIP?

As suggested, before planning a VoIP deployment or invest in more bandwidth, you must have adequate visibility and control to make sure you are generating maximum performance and value from your existing WAN links. The best place to start is by understanding exactly how your WAN bandwidth is being used. Most network managers do not have full visibility into their WAN traffic, and without application-layer visibility you will be ineffective at preparing your network for voice traffic — and that's when the problems begin.

How can I gain total visibility into WAN application traffic?

Blue Coat can help you discover all application level traffic running across your WAN links, whether it be critical business applications, email, FTP file transfers, Web surfing, recreational applications, malicious traffic, and more. Once you have this level of visibility into your network you may be surprised to learn that a significant portion of your WAN resources are used for recreational activity such as peer to peer (P2P), Utube video streaming, and audio file downloads. Blue Coat gives you the ability to immediately control or block this traffic, individually or as a class, as meets the particular network use policies of your organization — an essential step toward effective bandwidth management for VoIP.

Once I've determined bandwidth requirements and gained visibility into my applications, what are some of the critical VoIP performance challenges that need to be addressed?

There are three critical performance issues that need to be considered prior to VoIP deployment: The first is latency — the end-to-end delay is delivering the voice stream from the speaker's mouth to the listener's ear; the second is jitter — the unpredictable, variable delays in the delivery of each voice packet; and the third is packet loss — the dropping of individual packets caused by network congestion. Each of these three issues can cause significant degradation in voice quality and overall system reliability.

Tell me more about latency.

Because VoIP is real-time two-way communications, it is very sensitive to delays in the network. Acceptable VoIP quality requires a latency or delay of no more than 80 ms each way for true toll quality voice. Quality degrades as latency increases, but even with a delay of 150-180 ms each way, voice



quality is still in the “acceptable” range. In addition to the voice stream itself latency must also be addressed with other VoIP protocols (SIP, H.323, MGCP, etc.) that handle the call control functions between two systems or users. In fact, these signaling protocols are often even more sensitive to delays in the network than voice packets.

How can I minimize network delay/latency?

Latency is commonly associated with network congestion and poor bandwidth management. Again, this is seldom a problem within the corporate LAN where significant bandwidth is generally available to everyone who needs it. The real problem is at the LAN/WAN boundary as traffic transitions to smaller links. Some network devices attempt to overcome this problem by employing various queuing techniques (DiffServ, 802.1 p/q, IP ToS, etc.) to ensure voice packets take priority over other traffic waiting to get on the network. This is helpful to a certain extent, but being first in line to get on a crowded freeway doesn't mean you'll get to your destination quickly. What's required is more stringent and intelligent bandwidth management/QoS that can classify all VoIP related protocols, allocate a guaranteed amount of bandwidth to each traffic type, appropriately prioritize VoIP traffic as it traverses the WAN, and provide granular assurance of VoIP on a per-call basis. These advanced capabilities are available today in Blue Coat's application delivery network solutions.

Tell me more about the problem of jitter.

Jitter causes irregularities in the flow and delivery of data. This can be disruptive to a real-time application like VoIP. The tolerance for VoIP jitter is in the range of 20-30ms. If jitter causes delays to exceed this range — especially on a consistent basis — voice quality will suffer. Some VoIP vendors have tried to solve this problem by introducing their own jitter buffers or queues to temporarily store and “smooth out” the delivery of voice packets. Likewise, routers also offer queuing mechanisms for the same purpose. Both options, however, can exacerbate the problem by actually contributing to delays.

How can I prevent jitter from disrupting my VoIP traffic?

The ability to control and minimize network jitter is key to Blue Coat's application delivery capabilities by applying two patented technologies — TCP Rate Control (for data traffic) and UDP Rate Control (for voice traffic) — to assign and maintain a guaranteed rate and quality of all voice and data traffic across



the WAN. These technologies provide a similar function as conventional queuing in that they “smooth” delivery of the traffic, but do so in context of a more intelligent and policy-based bandwidth control strategy.

Tell me more about the problem of 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 VoIP’s performance. This is usually not an issue in the corporate LAN and should not be a problem if you rely on a MPLS service across your carrier’s backbone network. But packet loss can become a serious problem at the LAN/WAN boundary where the smaller pipe results in much greater contention for bandwidth. Although a packet loss of 1 percent or less is within the bounds of toll quality voice, once packet loss reaches 3 percent or more the listener will notice the conversation breaking up. Unless this problem is controlled packet loss can ultimately lead to dropped calls and the possibility of VoIP system failure.

How can I prevent packet loss from affecting my VoIP traffic?

The key to achieve optimal performance for business-critical applications like VoIP is to apply intelligent controls to the IP network to convert it from “best effort” to predictable. Blue Coat’s application delivery solution offers unmatched traffic control and performance optimization at the LAN/WAN boundary, by employing QoS and bandwidth management technologies capable of minimizing IP congestion and unpredictability and maximizing application performance over the existing WAN links.

Can compression help me improve VoIP performance across my existing WAN links?

Yes. Up to this point, we have highlighted the importance of gaining visibility into all of your WAN application traffic, then applying intelligent QoS and bandwidth management controls. These are critical steps in an application delivery strategy. However, the essential third step is to apply compression to specific traffic types. VoIP is compressed by the codec (analog-to-digital voice coder) used to deliver voice packets across the WAN. Many deployments use the G.711 codec for the LAN to package voice into 64 Kbps packets, and the G.729 codec for the WAN to compress voice traffic to 8 Kbps, while maintaining an acceptable level of voice quality. It is important



to note, however, that 8 Kbps does not include IP overhead, which increases the total payload to 24 Kbps or more. Additional compression of voice traffic is not advisable because it will compromise voice quality.

However, there will also be opportunities to apply compression to various data applications, such as email, ERP, and various types of Web traffic. Compression of these applications typically reduces data bandwidth requirements by one half or even one third, which can further improve VoIP performance. Blue Coat's application delivery solutions provide the ability to take full advantage of compression benefits.

Will my choice of VoIP signaling and control protocols impact network and application performance?

Early generations of VoIP technology relied on H.323 as the primary protocol suite for media signaling and control. But now SIP, MGCP, Megaco, and Cisco Skinny have all emerged as viable alternatives. Each of these protocols behaves differently on the network, but all can be effectively controlled by Blue Coat's application delivery solutions.

What impact will voicemail have on my WAN performance?

Voicemail messages often convey business-critical information, so the same basic issues apply relative to VoIP quality. The real impact of voicemail traffic on the WAN depends on the architecture of the VoIP system. Some vendors offer only centralized voice mail systems, which means that every remote user dials into a central server to retrieve voicemail. In such cases, there may be spikes of heavy VoIP traffic when users arrive at work in the morning. Other vendors may offer distributed voicemail, where the servers are dispersed across multiple sites, thus minimizing voicemail traffic across the WAN.

As you consider a VoIP roll out, you may find the need to consider a converged voice/data application — such as a CRM or call center application.

How can I make sure these new types of applications perform well?

All of the performance issues discussed so far — as well Blue Coat's ability to address them — apply in this scenario. Because Blue Coat provides the same visibility, control, and acceleration capabilities to virtually all voice and data application traffic types, the system can also help businesses align application and network resources to address emerging business requirements for virtual call centers and voice-enabled CRM.



Summarize Blue Coat's value in helping me prepare my network for reliable, high-quality VoIP communications?

Blue Coat is the global leader in application delivery solutions to help companies align their voice and data applications, and network resources, with the priorities of their business, while generating measurable cost savings in the process. This value is delivered through a family of intelligent appliances built with patented software technology that provides unmatched visibility, acceleration, and security capabilities. Specifically for VoIP, Blue Coat addresses chronic congestion, and related latency, jitter, and packet loss, that afflict voice quality at the LAN/WAN boundary. Blue Coat effectively manages all critical VoIP protocols and ensures the highest quality end-to-end communication on an individual, per-call basis.

About Blue Coat: Blue Coat Application Delivery Network solutions provide intelligent control with visibility, acceleration and security technologies that optimize application and network performance for any user, anywhere, across a distributed enterprise. Blue Coat's application performance management, WAN optimization and Secure Web Gateway solutions enable IT to provide greater business efficiency, effectiveness and competitiveness. Additional information is available at www.bluecoat.com.



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