PACKETEER®

Making the Right Call: Ensuring a Successful IP Telephony Implementation

IP Telephony

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Introduction

The benefits of a converged network seem obvious: not having to create, staff and run two different networks for voice and data, running just one cable to desktops instead of two and a reduction in phone tariffs are all good for business. But more interestingly, the converged network introduces a new paradigm in inter-office communications.

Real-time communications, such as escalating an Instant Message (IM) to a voice call mid-message or integrating presence data with multimedia and “traditional” communications like e-mail are rapidly being recognized and embraced as a true competitive advantage. Even simple things like being able to transfer your own extension number while visiting a branch office or just being able to have the same phone extension at multiple locations can really add up in terms of cost savings and convenience.

The world seems willing to invest for returns such as these. After several years in decline, the U.S. telecom business is expected to see double-digit growth again in 2006, hitting $1.2 trillion. Many analysts agree that companies moving to converged technologies will account for much of that growth. So if significant savings and flexible new ways to work lay ahead once your services and infrastructure converge, what’s the shortest, most painless route to taking advantage of these benefits?

A good convergence strategy involves finding and investing wisely in IP telephony applications that will save money and increase productivity. However, there are significant barriers to entry that must be recognized and addressed in order to make a clean transition to an operable — and effective — converged network. From unexpected details, such as the sheer number of required system components and expanded power requirements to the necessity of involving teams from outside the telecom department, a successful implementation requires extensive planning and preparation.

A large part of this preparation involves addressing the technical barriers that stand between you and your ability to fulfill the promise of IP telephony. The following six questions are essential towards ensuring a successful IP telephony implementation.

1. Is your network ready for IP telephony?
2. Can your data traffic and new IP telephony traffic co-exist on all spans of your network?
3. Do you have sufficient visibility into the network in order to initiate planning for and ensuring performance of voice and video calls?
4. Is there a way to increase capacity in order to accommodate the additional network traffic without buying additional bandwidth?
5. Can you differentiate between sanctioned and unsanctioned voice/video traffic?
6. Can you minimize the jitter, delay and packet loss that is associated with the Internet?

PacketShaper®: The Right Call for IP Telephony

PacketShaper, from Packeteer, makes convergence work by providing the tools required to ensure best technical practices are followed. With modules and software for unrivalled application monitoring, shaping, compression, acceleration and centralized management, PacketShaper enables you to:

- Assess the network for VoIP and video readiness (Monitoring Module)
- Monitor voice and video quality in real time (Monitoring Module)
- Alarm & diagnose issues (Monitoring Module and Management software)
- Differentiate consumer from business VoIP (Monitoring/Shaping Modules)
- Provision required bandwidth (Shaping Module)
- Increase capacity for converged traffic (Compression Module)

• Measure the impact of VoIP and video (Monitoring Module)

• Plan for your next application

Now, let's look again at those six essential questions and see how PacketShaper helps to address them.

Question 1: Is your network ready for IP telephony?

Assessing the readiness of your network is key to a successful VoIP deployment. You need to ask yourself if the same WAN bandwidth today is able to support your VoIP efforts tomorrow. If not, can you borrow bandwidth from less business- or time-critical applications or will you have to add capacity? And what will be the impact to your existing applications? How well will VoIP perform on your WAN?

You need to ensure your existing networks and the service levels established for data traffic are capable of supporting the additional demands placed on them by IP telephony. PacketShaper with the Monitoring Module gives you Layer 7 Plus visibility into the applications on your network and the bandwidth they're consuming. By discovering applications, tracking network utilization and monitoring performance, you can scope for bandwidth allocation. This will help you determine if you need to add additional bandwidth.

Question 2: Can your data traffic and new IP telephony traffic co-exist on all spans of your network?

If left unchecked, the additional bandwidth required for voice and video traffic on a converged network will undoubtedly impact the performance of business applications whose data shares that bandwidth. With PacketShaper, however, you can shape traffic to wring the maximum value from existing bandwidth without harming the performance of the applications most critical to your business.

In order to do so, you must first determine what applications are running on your network with PacketShaper's Layer 7 Plus technologies. You can then group applications together in a way that makes sense for converged networks by isolating:

• Bandwidth-hungry applications (sometimes called "bursty") like e-mail, file transfers, backup, database sync and distributed storage

• Applications that are time critical like some business applications and voice and video

• Recreational or malicious traffic

Before deciding that new voice and video traffic automatically merits the purchase of additional bandwidth, it makes sense to prioritize your applications. Use your list and have PacketShaper shape traffic and create room for your latency-sensitive business applications — like SAP or Oracle — as well as the new voice and video traffic.

Three easy steps to taking advantage of your existing bandwidth:

• Use the Shaping Module's partitions and policies to protect critical applications and contain or block recreational and malicious traffic.

• Use bandwidth management to smooth "bursty" traffic to reduce end-to-end congestion and improve overall network performance and efficiency.

• Take advantage of session guarantees, which will ensure the performance of applications like video and voice traffic.

Question 3: Do you have sufficient visibility into the network in order to initiate planning for and ensuring performance of voice and video calls?

Run a test session that generates voice and video traffic; PacketShaper can measure the bandwidth used. Then determine how many concurrent users you want to support. You can establish a rough estimate of your bandwidth needs by multiplying that number by the amount of bandwidth you measured.

• For voice, 10 concurrent sessions at 25 Kbps each would equal a total of about 250 Kbps in each direction

• For video, a 384 codec may require two concurrent sessions of 420 Kbps each (remember headers and control information are included)

Without visibility into voice and video performance, you're simply guessing at the requirements. PacketShaper takes the guesswork out of the equation, giving you the facts you need in order to ensure your implementation is a success.

Question 4: Can you differentiate between sanctioned and unsanctioned voice/video traffic?

Recreational voice and video traffic is becoming a greater concern as the technology is rapidly assimilated into our daily lives. Bandwidth intensive, recreational voice and video traffic can tie up even the largest of pipes. If your business relies on IP telephony as a part of your competitive edge, you need to find a way of differentiating between sanctioned and unsanctioned voice/ video traffic.

PacketShaper gives you the ability to differentiate between sanctioned and unsanctioned voice/ video traffic by looking beyond standard Quality of Service (QoS) and letting you shape network traffic to suit your needs. By monitoring the network with PacketShaper, you'll have the ability to view — and block — traffic that can potentially disrupt business- and customer-critical applications. Policies can be applied to protect critical voice and video traffic and smooth disruptive, bandwidth-intensive or bursty traffic from the other applications running across the network. Recreational traffic can be contained, and malicious traffic can be blocked.

Question 5: Can you minimize the jitter, delay and packet loss that is associated with the Internet?

Jitter, delay and packet loss are the three critical performance issues that need to be considered prior to deployment of any real-time services (such as IP telephony). Each of these three can cause significant degradation in quality and overall system reliability; as a group, they can take an entire service off-line.

Let's take a closer look at each of these issues to understand exactly why it is so essential that you minimize jitter, delay and packet loss in order to achieve a successful IP telephony rollout.

- Jitter — The unpredictable, variable delays in the delivery of each voice packet
- Delay — The end-to-end latency in delivering the voice stream from the speaker's mouth to the listener's ear
- Packet Loss — The dropping of individual packets caused by network congestion

PacketShaper has metrics designed specifically for real-time protocol traffic (RPT) like IP telephony. By monitoring the network and measuring jitter, delay and packet loss for RTP, their effect on call quality quickly becomes apparent. If the effect is substantial, controls can be enforced with the PacketShaper Shaping Module. Additionally, PacketShaper gives you the ability to alarm on those metrics so you don't just track performance to a service level agreement, you can actually automate pre-emptive steps to ensure it

Question 6: Is there a way to increase capacity in order to accommodate the additional network traffic without buying additional bandwidth?

Especially high on the list of potential problems for convergence projects is making room on the network for the increased traffic from video and voice sessions going between sites. Network convergence has many proven benefits; however, it does increase pressure on the network with additional traffic. An all-too-common response is to throw bandwidth at the problem or invest in a compression point-product. Unfortunately, this is not only an expensive reaction to the unwelcome traffic, it also tends to be ineffective. Since many recreational applications — be they voice, video or data — are opportunistic, they simply ramp to fill any pipe they encounter.

Without control in place, it's a waste of money to add bandwidth or invest in a compression product because nothing has been done to ensure the additional capacity supports customer and business-critical applications like Oracle or SAP, as well as the new video and voice business traffic. But PacketShaper integrates compression with control, giving you the best of both worlds. The PacketShaper Compression Module uses advanced compression algorithms to provide low-latency, high-impact compression to increase WAN capacity and create room for your convergence project — and the Shaping Module provides the control needed to ensure the applications most critical to your business get the bandwidth they need to perform at their peak.

RTP and Compression

Real-time protocol traffic (like the payload in voice and video traffic) is already very highly compressed by the codec. But there is scope for streamlining as they contain additional overhead: the RTP header, UDP header and IP headers. Packeteer's unique RTP compression packing technology super-compresses the headers of those traffic types and packs them together across many different calls — even with other compressed traffic — significantly saving bandwidth even as call volume increases.

What does this mean to you? Increased capacity for voice and video so you can fit more calls onto the same bandwidth, and a capacity gain that actually increases with the more calls you have on your network.

To learn more about the benefits of Packeteer, please visit: www.packeteer.com or call: 800.440.5035.

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