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Is Your Network Ready for IP Telephony?

Straight facts about IP telephony planning and deployment

ShoreTel

Enterprises are rapidly adopting IP telephony for cost savings, productivity gains and business innovation, but delivering a high quality voice service takes more than just buying the latest IP telephony equipment.

Successfully deploying IP telephony to your enterprise also means understanding the requirements for delivering toll-quality voice over your company's network infrastructure, and then appropriately planning for, choosing and deploying the right IP telephony solution. Making sure the network is ready for IP telephony is a critical success factor.

ShoreTel, the innovation leader in enterprise IP telephony, has set new standards for usability and manageability while reducing telecom costs. To ensure an optimal IP telephony experience, ShoreTel requires customers to have a thorough assessment of their networks. ShoreTel's solutions providers can assist with this assessment.

Here are the straight facts about planning for and deploying IP telephony in your enterprise.

IP Telephony's Business Benefits

Cost reduction, improved productivity, increased innovation and better collaboration are the measurable business benefits of IP telephony.

IP telephony offers significant cost savings by providing an alternative to high cost toll services. Organizations can reduce recurring voice toll charges and slash the high cost of supporting remote offices. IP telephony also enables enterprises to simplify their data and voice networks into a single converged infrastructure, thus improving operational efficiency.

Savings from IP telephony can be substantial. Savings can range from $9,600 to $28,000 per site annually for large organizations and between $4,800 and $9,600 for mid-sized organizations, according to Nemertes Research's study "Convergence: Reality at Last," published in 2004.

IP telephony increases organizations' ability to innovate. It increases worker productivity and efficiency by integrating call handling with Microsoft® Outlook® for efficient call management. IP telephony also enables collaborative and productivity operations such as unified communications, distributed contact centers, multimedia training and spreadsheet and data file sharing. Business, within and across the company, gets done more efficiently.

Architectural Requirements

IP telephony creates an opportunity for IT organizations to deliver superior service than with legacy PBXs, but delivering an enterprise-quality service means IT managers must pay close attention to IP telephony's architectural requirements.

- **Reliability and scalability:** Because of a distributed architecture, IP PBXs can deliver a more reliable, scalable voice service. For example, ShoreTel's systems are extremely reliable with no single point of failure. Organizations can easily scale their phone system as their needs grow. Expansion is flexible and seamless.

What's the Difference: Voice over IP and IP Telephony

VoIP is a generic term for using IP data networks like the public Internet to transmit voice traffic. VoIP has promised consumers savings by transmitting calls over the Internet, bypassing traditional phone companies.

After years of resistance, many service providers are now offering VoIP services. Phone companies and long distance providers have long been concerned about cannibalizing their traditional revenues by offering lower cost VoIP services.

The U.S. Federal Communications Commission considers VoIP services as an information service which requires regulation on a case by case basis. Neither federal nor state governance currently regulates VoIP services.

IP telephony uses a private IP network for voice calls, not the public Internet. IP telephony provides organizations with the ability to leverage their existing private IP data networks to transport voice traffic.

IP telephony is a cost-effective way of migrating an organization's intra- and inter-site voice calls away from traditional analog circuitry and PBX tie trunks (typically dedicated T-1s) and onto a company's dedicated data network, which are typically T-1 or T-3 on the WAN side and 100Mbps Ethernet or Gigabit Ethernet on the LAN side.

- **Ease of use:** Users can maximize their productivity with unified communications, converged conferencing, contact center and softphones. Integration with Microsoft Outlook makes it a snap to know who's calling as the phone rings. With find-me or presence applications, employees no longer miss calls when they are not sitting at their desks. Mobile workers can easily relocate their extensions to any other handset themselves on a temporary basis, allowing them to freely move from one campus to another while keeping their extensions.

- **Simple management:** IP PBXs simplify management, enabling management of a global IP telephony system from any location. For IT administrators, deploying IP PBXs frees them from the proprietary hold of the legacy PBX manufacturer. Adds, moves and changes don't require the assistance of costly outside systems integrators, as they can be done with a few mouse clicks.

Network Requirements for Toll-Quality Voice

The fundamental requirement to achieve toll-quality voice is to deploy an IP PBX over a properly architected network infrastructure. The LAN/WAN infrastructure must deliver sufficient throughput and meet latency, jitter and packet loss requirements.

Deliver sufficient throughput: The amount of bandwidth required for voice depends on the number of simultaneous calls, the voice encoding scheme used in the IP handset or softphone and the signaling overhead.

The International Telecommunications Union (ITU) G.711 codec is commonly used in LAN deployments, where LAN bandwidth is plentiful. With G.711 and RTP header compression, each call requires 82Kbps.

ITU G.729 is commonly used in a WAN environment because it uses substantially less bandwidth. With G.729 and no header compression, each call requires 26Kbps. With ADPCM and no RTP header compression, each call requires 52Kbps.

Meet latency and jitter requirements: Latency is the time from mouth to ear. It is the time it takes for a person's voice to be sampled, packetized, sent over the IP network, de-packetized and replayed to the other person.

Distance alone on the WAN circuit can cause delay, as can lower-speed WAN circuits. If latency is too high, it interrupts the natural conversation flow, causing the two parties to confuse latency for pauses in speech. Latency must not exceed 100 milliseconds (ms) one way for toll-quality voice and must not exceed 150 ms one way for acceptable quality voice. At 150 ms, delays are noticeable, but callers can still carry on a conversation.

Users hear jitter as degraded voice quality. Jitter is variation in latency over the LAN and WAN, as the IP telephony packets arrive in uneven patterns at their destination. Jitter has many sources: network congestion, queuing methods used in routers and switches, or routing options such as MPLS or frame relay used by carriers.

To compensate for jitter, ShoreTel's ShoreGear voice switches continually measure the jitter in the system and dynamically change the size of jitter receive buffers in 5 ms increments to optimize voice quality.

Packet loss requirements: Packet loss results in a metallic sound or dropouts in the conversation. Packet loss is caused by congestion, poor line quality and geographical distance. Since IP telephony is a real-time audio service that uses the Real Time Protocol (RTP) running over User Datagram Protocol (UDP), there's no way to recover lost packets. If even one or two percent of IP telephony packets drop, voice quality degrades.

ShoreGear's lost packet concealment capability reduces the impact of packet loss. When there's no voice sample to be played, the last sample is replayed to a receiving party at a reduced level. This is repeated until a nominal level is reached, effectively reducing the clicking and popping associated with low levels of packet loss.

Network Requirements for Toll-Quality Voice

Parameter	Requirement
Bandwidth	• With ADPCM and no RTP header compression: 52Kbps per call • With G.729a and no RTP header compression: 26Kbps per call • With G.711 and no RTP header compression: 82Kbps per call
Latency and jitter for toll-quality	• <100 ms total • 100 ms less 42 ms allocated for the ShoreTel5 system yields a 58 ms budget for the network • When G.729a encoding is used, 100 ms less 62 ms allocation for the ShoreTel5 system yields a 38 ms budget for the network
Latency and jitter for acceptable quality	• <150 ms total • 150 ms less 42 ms allocated for the ShoreTel5 system yields a 108 ms budget for the network • When G.729a encoding is used, 150 ms less 62 ms allocation for the ShoreTel5 system yields a 88 ms budget for the network
Packet loss	• <1 percent for voice calls and no packet loss for fax and modem calls

Assessment/Deployment Checklist

Planning, network assessment, systems integration, deployment and remote monitoring are key steps for a successful IP telephony implementation.

1. Start with the business requirements. How will the IP telephony system be used? What is the frequency and quantity of calls over the network? How many sites will be supported? The required bandwidth will depend on the call volume, applications used and even the codecs in the IP phones. For instance, supporting 10 simultaneous calls using G.711 requires 820Kbps of bandwidth.

What applications, such as video, voice, Web-based applications, enterprise applications, e-mail, backups and Web browsing, are used—and how much bandwidth does each consume? Gaining a good understanding of the application load on the network will help you prepare to meet the real-time demands of IP telephony.

Plan for growth. When designing the LAN/WAN infrastructure for IP telephony, consider your organization's needs in two years. Today's requirement may be for ten calls per minute, but in a year, it could grow to 30 calls per minute.

2. Assess your local network. Know what equipment exists in your network, and have an accurate architectural diagram. Make sure your network equipment is current, and use virtual LANs (VLANs) for voice traffic.

Toll-quality voice requires a switched Ethernet network, whether 10Mbps, 100Mbps or Gigabit Ethernet. You may need to upgrade older routers, switches or servers. Limit or eliminate broadcast or chatty protocols such as IPX, which add considerable unnecessary traffic.

VLANs will improve voice quality on the LAN. By setting up voice traffic to run in separate VLANs, IT managers can separate delay-sensitive voice traffic from data traffic right from the IP phones all the way through the switched network. Setting up voice traffic in a separate VLANs will also improve security and protect the conversation content.

Check for duplex mismatches. Duplex mismatches – full duplex on one end of an Ethernet connection and half duplex on the other end – are a major cause of IP telephony performance problems. Be sure to check the duplex settings of your connections and as well as the switch and router settings. Because the backbone has such a huge impact on performance, setting backbone connections to full duplex is particularly important.

3. Plan for multi-site connectivity. Assess how much WAN bandwidth exists today between sites – and how much is needed to support the anticipated number of voice calls. Define the number of connections between sites and understand how much WAN throughput is necessary.

IP telephony can be deployed over shared or dedicated WAN circuits as well as over an IP managed services. When connecting small offices or home offices, DSL can be used. Dedicated WAN circuits such as T-1 and T-3 will deliver the highest quality service.

Managed IP services are becoming a popular alternative to traditional dedicated circuits. Managed service providers offer IP connectivity over their private backbone, not over the public Internet. Because Internet performance varies, you should not rely on the public Internet to deliver an enterprise-quality voice service to remote users.

4. Use quality of service (QoS) on the network. Your QoS policies should give voice traffic higher priority over other less delay-sensitive traffic, so that voice conversations aren't interrupted by large data transfers.

Layer 3 QoS, whether DiffServe or Type of Service (ToS), is a system of identifying IP packets or traffic flows to group them. Once identified, the traffic can be marked into groups so that QoS policies can be applied to them. For example, Web access needs to be reasonably responsive, but e-mail response time can range from seconds to minutes. IP telephony and IP videoconferencing need a high level of QoS for enterprise quality.

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Is Your Network Ready for IP Telephony?

When deploying IP telephony over your organization's network, preparation counts. This way, you avoid any surprises at deployment time. Ask these questions first:

- Do you have a logical LAN/WAN diagram of your network?
- Do you have an inventory of all your network equipment?
- Do you have an IP address database?
- How much bandwidth do you have between sites?
- How many IP calls do you plan to run simultaneously through your IP network?
- Is your local network switched Ethernet?
- Does your WAN have sufficient bandwidth?
- What is your plan for setting up VLANs for voice on the local network?
- On the LAN/WAN, does your Layer 3 network support QoS?
- Does your network consistently meet toll-quality network performance standards for latency, jitter and packet loss?

The type of end-to-end QoS implemented will depend on the QoS supported on your routers and the IP telephony solution. Your IP telephony equipment, including phones and switches, should support QoS.

Note that Layer 2 QoS (IEEE 802.1p) settings are lost when the router rebuilds the frame. Most routers can translate the appropriate Layer 2 QoS information into Layer 3 QoS, but check that your router can do this translation at wire speed.

Service providers are migrating to MPLS for WAN links. MPLS explicitly reserves the combination of paths and QoS ahead of the arrival of any packets, and helps service providers better design their network core and deliver reliable services.

5. Establish a service level agreement (SLA). Negotiate a SLA with your WAN service provider to provide guarantees of throughput, availability, latency, jitter and packet loss. A SLA for voice quality might also include the call completion rate; the delay from when the last digit is dialed until a user hears a ringing or busy signal; fax performance; and a voice mean opinion score to measure voice quality. Carriers are beginning to put together increasingly complex SLAs as a point of service differentiation.

When deploying IP telephony to remote offices, ask the service provider which partners they use to deliver these services and if their partners will also support the SLAs. For instance, a major service provider may partner with a local provider to provide last-mile services using DSL or wireless.

6. Perform a network assessment. Network assessment services and tools are an invaluable measure of your network's readiness to support IP telephony and other real-time applications. A network assessment provides comprehensive performance assurance and real-time verification of performance right to the users' desktops. By scouting out potential problems in advance, the success of the deployment is increased.

7. Beware of virtual private networks (VPNs). Many enterprises use VPNs for secure remote access; however, the encryption adds overhead to the user sessions. Most VPN appliances do not increase latency, but software VPNs will introduce latency and can be problematic.

Performing a Network Assessment

ShoreTel's IP Telephony Network Assessment is a complete service to help you plan, design and implement an IP telephony solution to meet your organization's specific needs and ensure that IP telephony will run smoothly. The assessment is provided by ShoreTel's solutions partners or directly from ShoreTel. It is required prior to deployment.

ShoreTel's IP Telephony Network Assessment combines real-time and simulated testing which results in the pre-emptive discovery of network faults and potential performance problems. ShoreTel uses a network performance management tool from Viola Networks called NetAlly. NetAlly uses active application traffic to monitor and test actual applications and servers. It also collects passive performance information from IP PBXs, gateways and other network components. This provides repeatable and real-world tests for the most comprehensive performance assessment.

The NetAlly tests simulate IP telephony and the ability of the network to handle latency, jitter and packet loss. NetAlly also reports on voice quality in the form of a mean opinion score (MOS), which is a five-point scale established by the ITU in which 1 represents the poorest voice quality and 5 represents perfect voice quality.
NetAlly's agents send each other a variety of network traffic packets – using different application protocols, packet size, packet spacing and QoS levels. In addition to measuring peer-to-peer traffic, NetAlly's agents can also generate actual client transactions against production servers, including communicating with IP PBX servers.

By performing a pre-deployment network assessment, organizations gain an end-user perspective of network behavior. NetAlly's Web-based agents allow almost immediate performance testing to the user's desktop without the time, expense and security concerns of deploying physical resources or installing client software.

The initial results of a network assessment are delivered in minutes, although an assessment tests typically run for several days. It is vital that assessments be performed during peak operation hours to ensure an accurate picture of the network traffic. If trouble spots should arise either pre-deployment or during ongoing operations, these tools enable your solution provider or your IT team to rapidly isolate the source of the problem.

Network Readiness Deliverables: The Network Assessment service provides a detailed network readiness report of an organization's network environment and outlines the requirements for a successful IP telephony implementation. The Network Assessment has three components:

- Detailed assessment of the current environment, including the logical LAN/WAN architecture, an inventory of network components and IP addresses.
- A comprehensive IP telephony readiness report that analyzes network performance, utilization and capacity to support IP telephony.
- Recommended corrective actions to get your network ready. Organizations are provided with a known-issues report as well as short- and long-term recommendations for their network infrastructure.

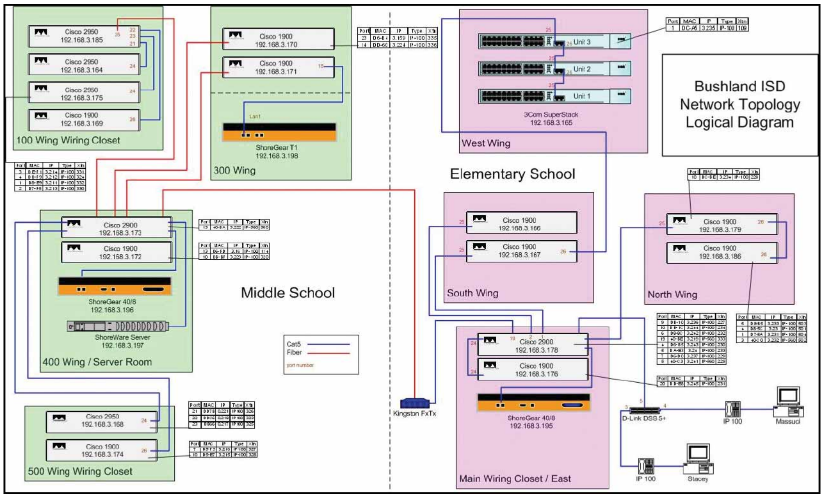

Figure 1: The IP Telephony Network Readiness report includes a detailed LAN/WAN network diagram.

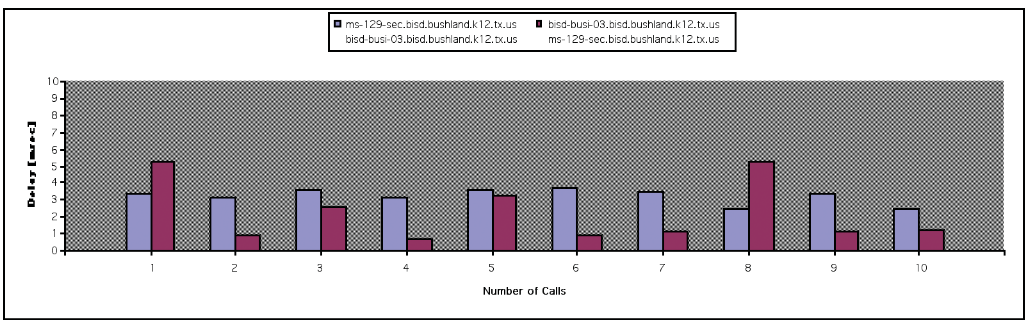

Figure 2: This snapshot shows network latency by number of IP telephony calls.

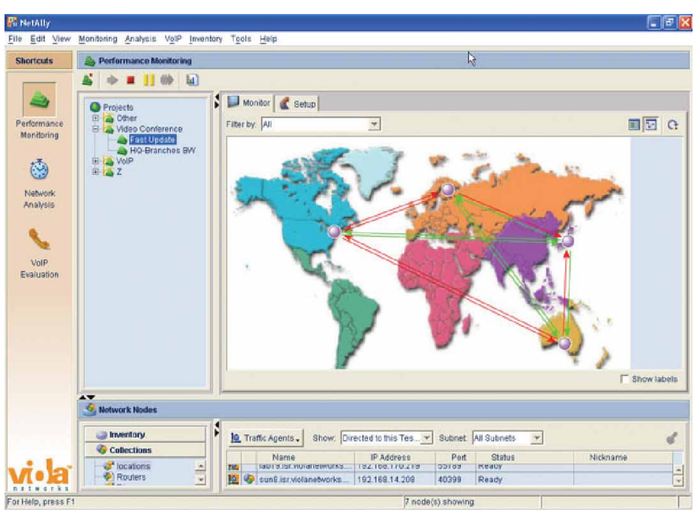

Figure 3: Monitoring of IP telephony performance across the distributed network.

Choosing the Right IP Telephony Partner

ShoreTel delivers an unrivaled, feature-rich, cost-effective solution that is easy to install, manage and maintain. Implementing IP telephony costs an organization between $525 and $1,512 per user, according to Nemertes Research's "Convergence: Reality at Last" study. While the eventual savings can be substantial, the startup costs depend on a number of variables, including the size of the enterprise and which vendor companies chose to supply the IP telephony solution. At $18 per user, ShoreTel offers the least-expensive start-up costs, according to Nemertes Research, while Cisco, Nortel and Avaya at $73, $50 and $31 respectively, are higher.

Key benefits of the ShoreTel IP telephony system include:

- **Distributed reliability.** ShoreTel IP phone systems are built on a distributed, embedded hardware platform with no single point of failure. IP phone and PSTN failover further ensure 99.999 percent reliability.

- **Best-in-class management.** Ideal for multi-site companies, a single-view interface enables a global IP network to be managed from anywhere with very little effort. Moves, adds and changes can be implemented in just a few keystrokes.

- **Unmatched productivity and ease of use.** ShoreTel has the most intuitive call management interface in the industry. Users can choose and customize more than 400 features, maximizing their productivity through powerful desktop applications, including unified communications, converged conferencing, contact center and softphone.

- **Phenomenal clarity.** ShoreTel leverages IP to deliver superior system and IP phone sound quality – often better than is possible over traditional landlines.

- **Interoperability, scalability and legacy integration.** ShoreTel systems fully interoperate with leading switches and routers. They scale gracefully for rapid or gradual system expansion and easily integrate with existing legacy phone equipment, such as PBXs and voicemail.

Start Talking

With proper configuration, IP telephony delivers superior voice quality at a significantly lower cost of ownership. Delivering a toll-quality voice service means choosing the right IP PBX solution, an experienced systems integrator and thoroughly preparing your network infrastructure for the demands of voice. Make sure your LAN and WAN have sufficient capacity. Use VLANs and QoS as appropriate. A network assessment will ensure that your phone calls come through loud and clear.

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