

WHITE PAPER

Is Your Network Ready For IP Telephony?

Straight facts about IP telephony planning and deployment

Introduction

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. Ensuring the network is ready for IP telephony is a critical success factor.

To ensure an optimal IP telephony experience, ShoreTel requires that customers undergo a thorough assessment of their networks. ShoreTel solutions providers are specially trained to assist with this assessment. As a result, ShoreTel customers enjoy high standards of usability and manageability, while reducing their communications costs.

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

The Business Case for IP Telephony

The main measurable business benefits of IP telephony are cost savings, improved productivity, increased innovation and effective collaboration.

Cost savings: IP telephony offers significant cost savings by providing an alternative to high cost toll services. Organizations can reduce recurring voice toll charges and reduce the high cost of supporting remote offices.

Productivity: Worker productivity and efficiency are increased with advanced, easy-to-use features, such as integrated call handling with Microsoft Outlook for efficient call management.

Innovation: IP telephony increases an organization's ability to innovate by converging data and voice networks into a single infrastructure, thus improving operational efficiency.

Collaboration: IP telephony also provides a flexible platform for collaborative unified communications (UC) capabilities, such as distributed contact centers, multimedia training and file sharing. As a result, business both within and across the organization is conducted more efficiently and productively.

WHAT'S THE DIFFERENCE BETWEEN 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 typically delivers huge cost savings by transmitting calls over the Internet, bypassing traditional phone companies.

As VoIP technology emerged, phone companies and long distance providers were initially concerned about cannibalizing their traditional revenues by offering lower cost VoIP services. After years of resistance, however, many service providers are now offering VoIP services.

Unlike VoIP, IP telephony uses a private IP network for voice calls, rather than the public Internet. Enterprise-class IP telephony enables organizations 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 100 Mbps Ethernet or Gigabit Ethernet on the LAN side.

Architectural Requirements

IP telephony creates an opportunity for IT groups to deliver superior service compared to legacy PBXs. But delivering an enterprise-quality service means IT managers must pay close attention to IP telephony's architectural requirements.

- **Reliability and scalability:** With a distributed architecture, IP telephony delivers a highly reliable and scalable voice service. ShoreTel uses a distributed architecture to help ensure exceptional reliability with no single point of failure. Also, system expansion is flexible and seamless, so organizations can easily scale their phone systems to meet changing business needs.
- **Ease of use:** UC features such as conferencing, contact center and softphones enable users to maximize their productivity. For example, integration with Microsoft Outlook shows who is calling as the phone rings. With find-me and presence applications, employees no longer miss calls when they are not sitting at their desks. Mobile workers can easily relocate their extensions themselves to any other handset on a temporary basis, allowing them to freely move from one office to another while keeping their extensions.
- **Simple management:** IP telephony designed as a single image system saves organizations valuable time and money by simplifying and centralizing management. For IT administrators, deploying IP telephony frees them from the proprietary hold of the legacy PBX manufacturer. Adds, moves and changes don't require the assistance of costly outside system integrators, as they can be done with a few mouse clicks—reducing the wait time from hours down to just a few minutes.

Network Requirements for Toll-Quality Voice

The fundamental requirement for achieving toll-quality voice is to deploy IP telephony 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 82 Kbps.

TU 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 26 Kbps. With ADPCM and no RTP header compression, each call requires 52 Kbps.

- **Meet latency and jitter requirements:** Latency 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 on the WAN circuit, by itself, can cause delay, as can lower-speed WAN circuits. If latency is too high, it interrupts the natural conversation flow and can cause 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, including network congestion, queuing methods used in routers and switches, and routing options such as MPLS or frame relay used by carriers.

To compensate for jitter, ShoreTel has designed its voice switches to continually measure the jitter in the system and dynamically change the size of the 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 the User Datagram Protocol (UDP), there's no way to recover lost packets. If even 1 or 2 percent of IP telephony packets drop, voice quality degrades.

ShoreTel Voice Switches include a lost packet concealment capability that reduces the impact of packet loss. When there is 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.

Table 1:
Network Requirements
for Toll Quality Voice

Parameter	Requirement
Bandwidth	<ul style="list-style-type: none"> • With ADPCM and no RTP header compression: 52 Kbps per call • With G.729a and no RTP header compression: 26 Kbps per call • With G.711 and no RTP header compression: 82 Kbps per call
Latency and jitter for toll-quality	<ul style="list-style-type: none"> • <100 ms total • 100 ms less 42 ms allocated for the ShoreTel system yields a 58 ms budget for the network • When G.729a encoding is used, 100 ms less 62 ms allocation for the ShoreTel system yields a 38 ms budget for the network
Latency and jitter for acceptable quality	<ul style="list-style-type: none"> • <150 ms total • 150 ms less 42 ms allocated for the ShoreTel system yields a 108 ms budget for the network • When G.729a encoding is used, 150 ms less 62 ms allocation for the ShoreTel system yields a 88 ms budget for the network
Packet loss	<ul style="list-style-type: none"> • <1 percent for voice calls and no packet loss for fax and modem calls

Assessment/Deployment Checklist

The essential steps in a successful IP telephony deployment are planning, network assessment, systems integration, deployment and remote monitoring.

Planning

Identify 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?
- What applications—such as video, voice, Web-based applications, enterprise applications, email, backups and Web browsing—are used?
- How much bandwidth does each application consume?

The required bandwidth depends on the volume of calls, applications used, and even the codecs in the IP phones. For instance, supporting 10 simultaneous calls using G.711 requires 820 Kbps of bandwidth. 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 that could grow to 30 calls per minute.

Assess Your Local Network

Know what equipment exists in your network and keep an updated 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 10 Mbps, 100 Mbps or Gigabit Ethernet. You may need to upgrade older routers, switches and servers. Also, limit or eliminate broadcast or chatty protocols such as IPX, which add considerable unnecessary traffic.

VLANs 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 from the IP phones all the way through the switched network. Setting up voice traffic in separate VLANs also improves security and protects 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 the switch and router settings. Since the backbone has such a huge impact on performance, setting backbone connections to full duplex is particularly important.

Plan for Multisite Connectivity

Assess how much WAN bandwidth exists today between sites, and how much is required 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 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 a private backbone rather than 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.

Use of Quality of Service on the Network

Your quality of service (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 that can then have QoS policies applied. For example, Web access needs to be reasonably responsive, but email response time can range from seconds to minutes. IP telephony and IP videoconferencing need a high level of QoS for enterprise quality.

The type of end-to-end QoS policy implemented depends 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 Multiprotocol Label Switching (MPLS) for WAN links. MPLS explicitly reserves the combination of paths and QoS ahead of the arrival of any packets, and helps service providers optimize the design of their network core and deliver reliable services.

IS YOUR NETWORK READY FOR IP TELEPHONY?

When deploying IP telephony over your organization's network, preparation counts. Sound preparation helps eliminate 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 voice calls do you plan to carry 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?*

Establish a Service Level Agreement

Negotiate a service level agreement (SLA) with your WAN service provider to provide guarantees of throughput, availability, latency, jitter and packet loss. An SLA for voice quality might also include: call completion rate, 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 these partners also support the SLAs. In some instances, a major service provider may partner with a local provider to provide last-mile services using DSL or wireless.

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.

Choosing the Right IP Telephony Partner

While the eventual costs savings can be substantial, the start-up costs of deploying an IP telephony solution depend on a number of variables, including the size of the enterprise and the choice of vendor.

To help organizations understand the total cost of ownership (TCO) of an IP telephony deployment, ShoreTel has developed the ShoreTel TCO Tool. This proprietary analytical assessment helps organizations calculate and compare the TCO of alternative systems over multiple years—an invaluable guide for decision-making.

Additional key benefits of the ShoreTel IP telephony system include:

- **Distributed reliability.** ShoreTel 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 (five nines) reliability.
 - **Best-in-class management.** Ideal for multisite companies, a single-view interface enables a global IP network to be managed from a Web-based browser 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 UC desktop applications, including converged conferencing, contact center and softphone.
 - **Exceptional clarity.** ShoreTel leverages IP to deliver superior system and IP phone sound quality—often surpassing the quality of traditional landlines.
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- Interoperability, scalability and legacy integration. ShoreTel systems fully interoperate with leading switches and routers. They scale modularly for rapid or gradual system expansion, and easily integrate with existing legacy phone equipment, such as PBXs and voicemail.

Start Talking Communications Improvements

With an optimized configuration, IP telephony delivers superior voice quality at a significantly lower total cost of ownership. Delivering a toll-quality voice service means choosing the right solution, working with an experienced systems integrator, and thoroughly preparing your network infrastructure for the demands of voice.



advanced
network
systems

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ABOUT SHORETEL

ShoreTel is the provider of brilliantly simple Unified Communication (UC) solutions based on its award-winning IP business phone system. We offer organizations of all sizes integrated, voice, video, data, and mobile communications on an open, distributed IP architecture that helps significantly reduce the complexity and costs typically associated with other solutions. The feature-rich ShoreTel UC system offers the lowest total cost of ownership (TCO) and the highest customer satisfaction in the industry, in part because it is easy to deploy, manage, scale and use. Increasingly, companies around the world are finding a competitive edge by replacing business-as-usual with new thinking, and choosing ShoreTel to handle their integrated business communication. ShoreTel is based in Sunnyvale, California, and has regional offices and partners worldwide.