SIP Trunking Gets Down to Business

Debunking five common misconceptions about extending business IP telephony

SIP trunks are setting the stage for an all-IP environment where business communications can be dynamic, blended, multimedia, mobile, efficient and intuitive.

Forward-thinking businesses have been quick to take advantage. Others are held back by concerns that have been mitigated or eliminated by technology advances and changing market conditions.

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Debunking five common misconceptions about extending business IP telephony outside the company's network

Voice over IP (VoIP) has come a long way since the first rudimentary services enabled primitive, free calls over the open Internet. Times have changed. The maturity of VoIP standards and quality of service (QoS) on IP networks has made VoIP a compelling choice for premium business communications, and not just to avoid toll charges.

Running business applications on a converged voice-and-data network streamlines the architecture, optimizes use of available bandwidth, reduces costs and enables powerful new services, such as "call follow-me," unified communications and number portability—blending the many ways people communicate in a dynamic workplace.

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In spite of the burgeoning popularity of VoIP for internal business communications, few companies are fully exploiting the potential for extending VoIP outside company walls, across service provider networks as well. Consider some possibilities:

- A call to your office number could ring at your desk phone, then your cell phone, then remote office, etc., until it finds you wherever you are, at home or on the road.
- From an airport, a hotel room, a Wi-Fi hot spot or anywhere, you could change these forwarding instructions as you change locations or form temporary project teams.
- You could collaborate with distant colleagues, customers or suppliers using shared Web browsing, desktop collaboration and videoconferencing, all linked to email and voice mail.
- You could log in to use your personal communications features, message stores, contact lists, preferences, etc. from any desktop or mobile IP phone, which doesn't even have to be a phone.
- A company could publish local telephone numbers for the various geographic locations it serves, yet handle all those calls in one location.

VoIP has the potential to make business communications more powerful and productive than ever, even as business is conducted far outside the reach of the enterprise network—and even when the "office" is an airplane seat today, a hotel room tomorrow, and a WiFi hot spot the next day.

SIP trunks offer ISDN-like features—and many new capabilities that ISDN cannot support—and delivers them with greater flexibility, more efficient use of bandwidth and lower cost.

The key ingredient that makes this possible—that enables VoIP applications to extend outside the company's internal network—is Session Initiation Protocol (SIP). SIP is an application-layer protocol that establishes and tears down sessions in an IP network, from simple telephone calls to collaborative, multimedia conference sessions.

SIP trunking is a service offered by an ITSP (Internet Telephony Service Provider) that connects a company's PBX to the ordinary telephone system (public switched telephone network, PSTN) via the Internet using the SIP VoIP standard and the same connection the company uses for Internet access, usually one or more T1 circuits.

SIP trunking offers many advantages over other types of trunk services. SIP offers ISDN-like features—plus a host of new capabilities that ISDN cannot support—and delivers them with greater flexibility, more efficient use of bandwidth, and significantly less network complexity and cost.

What Does SIP Do?

When a user places a call from an internal phone to an external number, the PBX sends the necessary information to the SIP trunk provider, which sets up and manages the call.

- If the number being called is a traditional telephone, the service provider routes the call to the PSTN gateway closest to the called number, to minimize long-distance charges.
- If the number being called is also on a SIP trunk, the call could travel on IP networks for the whole journey (for very low or no per-minute charges), since ITSPs often have agreements to carry calls for each other.
- If the called and calling numbers are both on SIP trunks, but there is no continuous IP path between those points, the call could be handed off to the PSTN and then back into the IP realm via gateways.

Organizations purchase as many SIP trunks as they need to support their specific needs, typically one trunk for three to five employees. Some providers allow several calls to be carried over a single trunk. For these VoIP calls, SIP manages the following primary functions:

- Translates name and user location—Ensures the call reaches the called party, wherever he/she is located, and supports the details of the call (session)
- Negotiates features—Enables all participating devices in the call to agree on the features supported, since not all parties will have the same features
- Manages call participants—Transfers or places a caller on hold, or calls or cancels connections to other users during the call
- Changes call features—Changes call characteristics during the course of a call, such as from a voice-only call to a voice-and-video session as needed

SIP is the key technology behind IP Multimedia Subsystem (IMS) services, such as texting, instant messaging, online gaming, and sending photographs and video from mobile devices.

The SIP protocol interworks with many other existing protocols to enable a variety of innovative IP services, such as voice-enriched e-commerce, Web page click-to-dial, Instant Messaging with buddy lists, IP centrex and IP contact centers.

So it is no surprise that SIP has been formalized as a standard (RFC 3261) by the Internet Engineering Task Force (IETF), the body responsible for administering Internet mechanisms. SIP is also the key technology behind IP Multimedia Subsystem (IMS) services, such as texting, instant messaging, online gaming, and sending photos and videos from mobile devices.

Forward-thinking businesses and institutions have been quick to take advantage of SIP for serious business communications. “SIP support is swelling,” wrote Edwin E. Mier and Martin Milner of Miercom, a New Jersey network consulting and product test center *(Business Communications Review*, June 2006). “Vendors—as well as enterprise users—are increasingly serious about putting SIP to work.” Revenue in the total VoIP industry in the US is expected to grow by 24.3 percent in 2008 to $3.19 billion, with 21 percent growth in subscribers to 16.6 million in 2008. (*VoIP Monitor*, March 19, 2008).

However, even those who love VoIP for their personal communications often hesitate to commit to it for external business communications—held back by five common myths and misconceptions about the technology. Let's take a look at these concerns and how advancements in the technology and the industry at large have mitigated or eliminated them.

Myth #1 "We wouldn't gain much by changing ISDN PRI trunks to SIP trunks."

Actually, you do, and here's why. Typically, a small to mid-sized business sends its voice and data communications to the outside world via one or more DS1/T1 lines to the service provider, each providing 1.544 Mbps of bandwidth. This T1 circuit consists of 24 eight-bit channels, each capable of transmitting 64 kilobits per second (kbps), the standard rate for carrying a voice telephone call. If the T1 is set up as an ISDN PRI (Primary Rate Interface) trunk, it has 23 channels that can carry voice and data; the 24th channel carries signaling for caller ID and automatic number identification (ANI).

In the traditional digital world, this T1 trunk can be ordered as a channelized circuit. Any number of channels can be configured for data, and other channels configured for non-data traffic, such as traditional voice calls.

However, there are several problems with this model of separate voice and data networks using ISDN trunks, most notably: complexity, wasted bandwidth and excessive cost for T1s.

Would you like to simplify the communications infrastructure?

With traditional telephony, the T1 trunk from the business to the service provider can carry both voice and data, but the company still has to maintain separate infrastructures on its end to handle voice and data.

- A time-division multiplex (TDM) network handles voice communications and legacy data services (fax, modem) over traditional telephone lines.
- An IP network handles Internet access and modern data services, such as email, IP videoconferencing and high-speed Ethernet.

With separate infrastructures, a company has two networks to buy, deploy, operate, manage, troubleshoot and upgrade—a costly proposition.

SIP trunking dramatically reduces that cost. VoIP with SIP trunking enables the business to converge voice and data on one IP network—fewer network elements, streamlined architecture, only one network to manage.

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Would you like to minimize the number of T1s to buy?

> *With traditional telephony*, a lot of costly ISDN PRI trunk bandwidth sits idle. Since channels are fixed as either voice or data channels, you cannot allocate bandwidth at will as the traffic mix changes. For instance, a company must configure enough voice channels to meet peak call volume, even though call volume would normally be far lower. This limitation easily translates into paying for T1 trunks that receive light or occasional use—another costly proposition.
>
> *SIP trunking* eliminates that inefficiency. You don't have to specify certain trunk channels as voice and others for data. To the SIP trunk, it's all data. It's either data to begin with, or voice that has been converted to data. Available bandwidth can therefore be dynamically allocated as needed among voice and data services. When an influx of phone calls comes in, the system applies more bandwidth for voice and less for email or other data services where slight delay isn't noticeable. Conversely, idle voice channels can be used for data services, such as Internet access.

On SIP trunks, available bandwidth can be dynamically allocated among voice and data services as needed.

There's no need to overprovision trunks just to meet periodic peak demand on one service or another.

Physically the ISDN PRI trunk and SIP trunk are the same. The T1 circuit itself is a chameleon, its character determined by the interface cards at either end of the connection and programming from the service provider office. If your organization already has a T1 line for Internet access (and who doesn't?), the same physical connection can very effectively be used to carry VoIP traffic onto the public network as well.

This convergence enables the new productivity and convenience services described earlier, such as call follow-me and multimedia conferencing. Even without the productivity gains, the business case is compelling. The price of a SIP T1 trunk tends to lower than an ISDN PRI trunk, and the organization requires fewer T1s overall. Long distance termination charges associated with SIP trunks are also lower than traditional analog or TDM rates.

Myth #2 "Call quality won't be good enough to reflect our company image."

When people think of VoIP, they often think of Yahoo Messenger or Skype, applications that offer free or low-cost voice calls over the Web, with variable quality. The trouble is, the open Internet is not necessarily engineered to meet the stringent requirements of business-grade voice traffic, in terms of latency, jitter and packet loss. Those Voice over Internet services can be great for avoiding toll charges for personal or intra-company calls, but an organization could rightfully expect better sound quality and reliability for its communications with clients, customers, the media and other external audiences.

With engineering and QoS measures in place, users should not experience a difference in voice quality between circuit-switched (conventional) and packet-switched (IP) telephony.

VoIP over SIP trunks is quite different from Voice over Internet. VoIP with SIP trunking travels over privately managed IP networks or controlled Internet backbones that have been engineered for the bandwidth and quality-of-service (QoS) attributes needed for voice.

For example, AT&T manages VoIP services over a global IP backbone network that has enough fiber-route miles to circle the equator more than 20 times. Service level agreements assure business-grade levels of packet delivery, latency and "five nines" (99.999 percent) availability of the IP network from end to end.

Granted, business-grade VoIP SIP trunks are more expensive than your typical $39.95 Voice over Internet service, but the quality is far superior. In fact, with engineering and QoS measures in place, users should not experience a difference in voice quality between circuit-switched (conventional) and packet-switched (IP) telephony.

Of course, the quality of the service provider's network is only part of the equation. A company will need voice-grade data services for the last leg of the call, inside the company, from the SIP trunk connection to users. This is not a technically challenging requirement. Most modern business networking systems—even the consumer-grade switches generally available at retail stores— support QoS features and operate at the 100 Mbps or 1000Mbps throughput speeds needed to support business-grade VoIP.

Myth #3 "It is too complicated and costly to deploy a VoIP infrastructure."

Actually not. VoIP with SIP trunking streamlines a company's overall communications architecture. You can converge services from two separate networks into one network— one that has relatively simple requirements. Here are the typical elements:

- A PBX with a SIP-enabled trunk side. Toshiba PBX systems equipped with a VoIP interface card—from the Strata CIX40 for small businesses to the Strata CIX1200 for large enterprises—natively support SIP trunking using an interface card. [Deleted statement about "older PBXs…" and changed order of bullets.]
- IP telephones or PCs. Full-featured IP phones that look and feel like traditional telephones are widely available. For example, Toshiba's IP5000 telephones include all the features and functionality of our digital telephones, enabling users to have the best of traditional telephone features in an IP telephone. An ordinary PC or laptop can be equipped with Toshiba SoftIPT® software to serve as an IP "softphone."

These devices connect directly into a standard computer network port. There's no need for the old phone jack and separate telephone wiring. Since the IP phones identify themselves to the network, adding or moving extensions is as simple as unplugging from one port and plugging into another.

The user's endpoint doesn't have to be an IP device. You can have a traditional telephone connect to an analog station card on the PBX that connects to a SIP trunk. This flexibility makes it easy to upgrade to VoIP in stages. However, converting voice from analog to IP and vice versa will add some delay that could erode voice quality and cause impedance mismatch, where you hear yourself talk. All-IP from end to end is the best bet for highest quality.

- An Internet telephony service provider (ITSP) or SIP trunking provider. With the rapid growth in VoIP for external business communications, these service providers are easy to find, especially in metropolitan markets. The ITSP provides any integrated access device, such as a Cisco router, that may be necessary to connect the trunk to the PBX.
- A local area network (LAN) that supports quality voice. If the company's internal network is engineered for the appropriate levels of delay, jitter and data loss, it will deliver expected levels of speech quality without compromising performance for other critical business applications.

If the company's internal network is engineered for the appropriate levels of delay, jitter and packet loss, it will deliver VoIP speech quality at least on a par with traditional telephony.

Assess the present state of the data network. If it is slow or at times quirky, users won't complain (and might not even notice) when they are surfing the Web and sending email. VoIP demands more. If the organization uses Cat3 cabling and 10Mbps hubs—and hasn't upgraded the data network since the early 1990s— you will need to make some changes, but chances are those upgrades were overdue even before VoIP entered the picture.

Myth #4 "We'll face dead zones where SIP trunk calls can't go through."

SIP trunking and managed IP networks are more widely available than ever, covering most major business centers throughout North America, and internationally.

For example, AT&T says its global IP backbone network reaches 97 percent of the world's economies. PAETEC delivers business-class services in more than 80 of the nation's top 100 metropolitan areas. California-based American Broadband Services offers nationwide coverage in the U.S.. Cbeyond, Inc. delivers IP managed services across Atlanta, Chicago, Dallas, Denver, Houston, Los Angeles and San Diego. Even small mom-and-pop telephone companies will offer SIP trunking where demand makes it worthwhile.

Coverage is very good—and continually getting better—for VoIP calls to traverse managed IP networks from end to end.

In short, coverage is very good—and continually getting better—for VoIP calls to traverse managed IP networks from end to end. Where IP connections or reciprocity agreements don't exist, the VoIP call can still be handed off to the PSTN and brought back into another IP network again. So there should be no fears that a business that commits to VoIP will find itself out of touch with customers or suppliers they formerly could reach.

Myth #5 "SIP standards are evolving, so interoperability will be an issue."

This issue can be problematic, but it doesn't have to be. Yes, the SIP trunk needs to interoperate with the IP PBX; and no, not every SIP trunking provider interworks seamlessly with every IP PBX. Such interoperability requires complete agreement on the details of a protocol that has not been specified to the *nth* level of detail. In fact, SIP standards were intentionally designed to be flexible and adaptable.

"To assure a clean deployment, enterprises should be sure two SIP vendors have worked out interoperability issues between themselves."

***Edwin Mier** of Miercom, in Business Communications Review, June 2006.*

"SIP is not the panacea. It was never designed that way, and that's a good thing," according to the SIP Center (www.sipcenter.com). "Typically all-inclusive approaches (like H.323) have been fraught with difficulty and represent the wrong kind of thinking in today's modular network. SIP is flexible, but it sticks to doing what it does best. … SIP works with a number of other protocols to get the job done while still playing nicely with some neighboring technologies," such as H.323, MGCP, MEGACO and MIME.

The inherent flexibility of the SIP protocol puts the onus on IP PBX vendors to make it work in the field with other vendors' interpretations of it. This responsibility troubles many of them. In a vendor survey conducted by Miercom in association with *Business Communications Review*, respondents said the expense and labor of assuring interoperability was "absolutely" an issue of concern, rating it 3.7 overall on a five-point scale.

In their haste to get to market early, some IP PBX vendors delivered SIP solutions that suffered from interoperability snafus, requiring lots of troubleshooting and custom configuration—and causing lots of frustration for customers… which gets us back to the first point: interoperability can be problematic, but it doesn't have to be.

"To assure a clean deployment, enterprises should be sure two SIP vendors have worked out interoperability issues between themselves," said Mier in *Business Comunications Review* (June 2006). "Comparing SIP systems based on supported SIP features can be frustrating and misleading. Features are implemented in various ways and with varying prospects for third-party interoperability."

At Toshiba, we believe it is critical to put proven solutions in the field, even if it meant others might get into the marketplace earlier. So we have conducted rigorous interoperability testing with ITSPs, testing every type of call scenario possible, capturing the log files, and addressing any issues where SIP standards were open-ended and incompatibilities were found. As a result of this work, Toshiba has announced certification with AT&T, CBeyond, PAETEC, American Broadband Services and IPtimize. Other ITSPs will be certified in this ongoing process.

Vendor certification says to the customer, "When you buy this license, install this interface card and engage with this ITSP, it will work."

With such a testing and certification process in place, Myth #5 can become a worry of the past. Certification says to the customer, "When you buy this license, install this interface card and engage with this ITSP, you *will* get dial tone, your features *will* work, and you will not spend hours trying to figure out why it doesn't."

Closing Thoughts

SIP trunks are setting the stage for an all-IP environment where business communications can be dynamic, blended, multimedia, mobile, efficient, intuitive—reflecting the changing nature of the business stage itself.

SIP trunks can offer significant cost savings for enterprises, reducing the need for costly T1 trunks and eliminating the need for local gateways to the public switched telephone network.

Geographic limitations dissolve. Teams collaborate across counties or continents. Connectivity follows people instead of tethering them. And as prices for just about everything are spiraling up, your network and communications bills can go down.

The chart shows on the next page how ISDN PRI trunks and SIP trunks compare, side by side, on six key factors that influence the cost and quality of business communications.

ISDN and SIP Trunking—How Do They Stack Up?

Issue	Traditional ISDN Trunks	New SIP Trunks
Features	Full set of ISDN features Dozens of convenience and productivity features for business telephony, such as call forwarding, call waiting, caller ID, hunt groups, audio-conferencing, and more	Full set of ISDN-like features and more Additional new IP services such as voice-enriched e-commerce, IP contact centers, call follow-me, unified communications, number portability
Multiple DIDs per Trunk	Yes. Multiple direct inward dial (DID) numbers on a single trunk (compare to analog trunks, which had to be provisioned one-to-one)	Yes. Multiple direct inward dial (DID) numbers on a single trunk, with the added benefit of more efficient utilization of each trunk
Business-Grade Speech Quality	Yes. Voice quality achieved by one-to-one provisioning; one voice call per 64-kbps channel (23 DIDs per ISDN PRI trunk) to support two-way phone conversations	Yes. Voice quality achieved by prioritization; voice call packets marked with QoS and given higher priority than data to maintain PSTN-equivalent speech quality
Network Complexity	Separate company networks for voice and data; two networks to buy, deploy, operate, manage, troubleshoot and upgrade	One streamlined network for both voice and data; fewer network elements, streamlined architecture, only one network to manage
Bandwidth Efficiency	Inflexible and inefficient. Channels configured as either voice or data channels; no flexibility to use idle voice channels for data, or vice versa	Efficient and dynamically allocated Voice and data on any channel, because voice is converted to data; all bandwidth available for data when no voice calls are active
T1s to Provision	Must over-provision for peak demand Must configure enough voice channels to meet peak demand; may end up paying for additional, under-utilized T1 trunks	Adjust policies/priorities for peak demand Bandwidth dynamically allocated among voice and data services as needed, to maximize value from every bit of available bandwidth
Service Provider Cost	High costs for trunks and toll charges $500 to $700/month for extra T1 circuits; higher per-minute toll charges for calls outside the local area; multiple bills and vendor relationships to manage	Reduced costs for trunk and toll charges Potentially fewer T1 circuits; minimal (or no) long-distance charges for calls that stay on IP networks; one bill and one point of connection for voice and broadband Internet needs

These benefits should no longer be missed because of outdated myths about the SIP protocol—or the hardware and services that support it. VoIP and SIP are definitely ready for prime time. It's time to take advantage.

About Toshiba America Information Systems Inc. (TAIS)

Headquartered in Irvine, Calif., TAIS is organized into four business units: Digital Products Division, Imaging Systems Division, Storage Device Division, and Telecommunication Systems Division. Together, these divisions provide mobile products and solutions, including industry leading portable computers; projectors; imaging products for the security, medical and manufacturing markets; storage products for automotive, computer and consumer electronics applications; and telephony equipment and associated applications.

TAIS provides sales, marketing and services for its wide range of information products in the United States and Latin America. TAIS is an independent operating company owned by Toshiba America, Inc., a subsidiary of Toshiba Corporation, which is a global leader in high technology and integrated manufacturing of electrical and electronic components, products and systems, as well as major infrastructure systems. Toshiba has more than 191,000 employees worldwide and annual sales of over US $60 billion (FY2006).

SIP Trunking from Toshiba

Toshiba IP PBX systems, from the Strata CIX40 for small businesses to the Strata CIX1200 for large enterprises, all support SIP trunking. Strata CIX systems integrate SIP trunk support right into the communications system using a multi-purpose interface card. SIP trunking is available from Toshiba's nationwide network of Authorized Toshiba Dealers.

For more information on Toshiba's leading innovations, visit the company's Web site at www.toshiba.com. For more on Toshiba telecommunications systems: http://www.toshiba.com/taistsd/index.jsp

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