XO Communications

White Paper

Why SIP Makes Sense?

Enabling the Evolution to Unified Communications

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Commissioned by
XO Communications

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Why SIP Makes Sense:

Enabling the Evolution to Unified Communications

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Abstract

Written specifically for IT decision makers in medium to large-sized businesses, this white paper presents an overview of the Session Initiation Protocol (SIP) standard, its drivers, benefits, and barriers to implementation. The paper presents several business applications for SIP trunking and discusses the advantages of SIP as an enabler for Unified Communications. Because SIP has become the de facto industry standard for IP telephony development, the paper concludes that businesses and enterprises of all sizes will be well served if they consider SIP as part of their evolving network strategy.

Introduction: The Evolution to Unified Communications

Setting the Stage

Remember the first mobile phones? You could always tell when someone had one: they walked hunched over like Quasimodo because of the heavy battery inside. We all coveted them — everybody wanted one. They were so desirous that in the mid-80s the Sharper Image Company actually sold a fake cell phone that had a suction cup on the end of the handset cord that could be affixed under the dashboard, allowing the owner to drive down the street looking *cool*.

Remember the first laptops? They were equally cool; we all wanted one of those as well. But I'd be willing to bet that very few people reading this paper today carry a laptop because it's cool. My suspicion is that they carry a laptop because it has their *stuff* on it. All the Microsoft Word documents, spreadsheets, presentations, images, PDF documents, sound files, music, movie clips, and other digital paraphernalia that our work and personal lives require are typically resident on our computers and must, therefore, accompany us wherever we go.

However, what if the need to have the physical computer were to go away? What if there was a way to have access to work-related content at *anytime, anywhere* in the world, on any network, using any access device over any access technology? Would that not simplify life dramatically and make use of the network more efficient and relevant? Well, that is more than a possibility — it is becoming a reality *today*.

"The growing proliferation of Unified Communications (UC) technology in the workplace is enabling anytime, anywhere communications."

Enabling the Evolution

Three phenomena are making this evolution possible:

- *Inexorable, steady migration of content* from the hard drive on the user's device (PC or laptop) to storage arrays located within the network — and managed by the network provider or hosted by the enterprise.
- *The growing proliferation of Unified Communications* (UC) technology in the workplace, which is enabling anytime, anywhere communications. UC applications include IP telephony, unified messaging (i.e., voicemail, email, fax), instant messaging, presence, web/rich media conferencing, document sharing, and the support for Internet Protocol (IP) applications supported across different device types, including

mobile phones, PDAs, and softphones, such as laptops. Driving the adoption of UC is the growth of IP telephony, which has already overtaken TDM in the enterprise. According to industry analysts, 70% of telephony ports shipped worldwide in 2006 were for IP, expected to reach 87% in 2011.[1]

"Driving the adoption of UC is the growth of IP telephony, which has already overtaken TDM in the enterprise. According to industry analysts, 70% of telephony ports shipped worldwide in 2006 were for IP, expected to reach 87% in 2011."

- *The extension of the Internet Protocol* as an application layer signaling protocol in the telecommunications network infrastructure, for both wireline and wireless networks. SIP enables the suite of IP-based Unified Communications applications to be extended from PBX systems and/or an IP network across different network transport services and end user devices to support remote sites and mobile workers. In essence, it allows any user on any network to have access to any content on-demand, regardless of their physical location, the device they're using, or the access modality they're employing. How important is SIP in enabling Unified Communications? According to one industry analyst, "It's very important; I don't really see how we can do without SIP in the long term."[2]

This three-part migration facilitates a number of customer advantages, including increased enterprise productivity through improved support for distributed workforces, timelier and higher quality responsiveness to customers, and reduced capital expenses (CAPEX) and operating expenses (OPEX) for telecommunications infrastructures. On the small business and enterprise side of the house, it gets even better. Using SIP helps reduce the cost of networking, plus SIP trunks free the customer from being locked into a specific system vendor or network provider — now and in the future.

Signaling the Future

Signaling — the process of establishing a call, invoking enhanced services required for the call (many based on the Caller ID function), maintaining the call, and tearing it down at the end — is performed by protocols that are part of the Signaling System 7 (SS7) network, a functional adjunct to the Public Switched Telephone Network (PSTN). However, as the migration to Internet Protocol (IP) continues, SS7 protocols are being replaced with IP protocols, most notably the *Session Initiation Protocol*, or SIP. Instead of signaling being done in the SS7-based core of the network, it is being done at the edge, and specifically by a small SIP client that resides on the customer's access device. In essence, this represents an inversion of network functionality: storage is leaving the device and going into the network; call setup is leaving the network and going to the customer's device.

Concurrent with this trend, service providers are working feverishly to upgrade their network platforms to provide value-added features and applications that differentiate their offerings and complement the SIP standard. One such feature is virtual numbers, a significant benefit for companies with a distributed workforce and customer base. Business continuity and network disaster recovery can also be elements in a SIP environment, affecting not only customer service, but also providing network managers with additional ways to improve network availability.

This paper presents an overview of the ongoing SIP migration, including an explanation of what it is, how it works, and why it is important to businesses today.

[1] "Ports shipped" numbers include both IP PBX and IP modules for TDM equipment. Frost & Sullivan, *Worldwide Enterprise IP Telephony Markets*, June 2007.

[2] Zeus Kerravala, Senior VP of Enterprise Research, Yankee Group, Interview with XO Communications, 27 December 2007.

Overview of the SIP Signaling Protocol

SIP is an application layer protocol that is used to establish, maintain, modify, and end communications sessions between two or more parties. As such, it can establish and manage:

- Two-party, multiparty, or multicast sessions
- Internet telephony
- Distribution of multimedia content
- Management of multimedia conferences

SIP is designed to be completely independent of the transport layer and can operate over the Transmission Control Protocol (TCP), the User Datagram Protocol (UDP), or the Stream Control Transmission Protocol (SCTP). SIP-based clients (devices) use TCP or UDP protocols to connect to SIP servers.

The primary driving force behind SIP's development and deployment was the perceived need to develop a signaling protocol for IP-based networks that can support the standard call processing functions found in the PSTN. The SIP protocol itself does not define these features, but enables their creation in network elements, such as proxy servers and user agents, which are discussed later (see **Figure 4**). These features include digit collection, call ring, ring back, busy tone, and fast busy or reorder. And while the manner in which these functions are delivered in an IP-based SIP environment is somewhat different from the PSTN, the overall result is identical.

SIP Advantages and Applications

When deployed in a network, SIP offers a collection of distinct advantages that can be used to develop powerful and compelling end-user applications. For example:

1: Intelligence at the Edge

SIP-enabled telephony systems offer most of the call processing and feature invocation procedures offered through SS7 — but in a different way. SS7 is a hierarchical, centralized, core-based protocol designed around the limited requirements of telephone sets — which have no innate intelligence. SIP, on the other hand, is a *peer-to-peer protocol*, which requires a different type of network core infrastructure (preferably MPLS built on an IP platform) with a blend of intelligence located at the edge (i.e., software or end-user device hardware or PBX), complemented by more scalable, granular, and rapidly deployed services offered by network operators.

2: SIP Is Part of the Overall IP Suite

As part of the overall IP suite, SIP is *flexible* and extraordinarily *dynamic*. Its functionality can be extended to any number of applications, including enhanced signaling for value-added services, VoIP, and XML-tagged applications. Because XML is used to structure, store, and send information across the network, it works well with SIP in environments where data needs to be retrieved and used, as in a call center environment where customer records must be accessed, or in a healthcare environment where access to customer data is critical.

"SIP is flexible and extraordinarily dynamic. Its functionality can be extended to any number of applications, including enhanced signaling for value-added services, VoIP, and XML-tagged applications."

As a "lightweight, text-based protocol," SIP relies on a text-based command structure that uses the now universally familiar HTTP syntax and URL addressing, both ideal for delivering telephony over an IP network where the logical integration of applications (e.g., voice, messaging, conferencing, and Web access) can create an enhanced customer experience.

3: Supports Any Network Transport Medium

Because SIP is an *application layer protocol*, it can ride seamlessly across any transport scheme and be transported across any access modality — cable, DSL, private line, Ethernet, and wireless. Thus, SIP can enable a broad range of applications and remote session capabilities (such as mobile application delivery and supply chain management) without the need to provision additional transport services unnecessarily. From an enterprise point-of-view, this is critical because SIP offers seamless connectivity options for service delivery for branch locations, remote workers, or trading partners. Since branch offices account for anywhere from 30% to 90% of the enterprise's employees globally, depending on the industry,[3] this will be a significant benefit for the enterprise.

4: Mobility and Presence Support

"Because SIP unshackles users' physical location from their logical address, they can have fully integrated corporate communications regardless of location. They can also integrate instant messaging and desktop collaboration applications."

SIP is now incorporated into a range of user devices, including mobile wireless devices and desktop clients. Using SIP, session establishment requests are not sent to a device; they are sent to the network, which locates the user's "presence" and establishes a session based on the user's current location and usage profile. Because SIP unshackles users' physical location from their logical address, they can have fully integrated corporate communications — regardless of location. They can also integrate instant messaging and desktop collaboration applications.

Presence is a relatively new concept in the networking arena. The user's client publishes his or her availability within the presence application, and all users then have access to that person's availability via all services — office wireline voice, mobile, e-mail, chat, etc. Users can customize their availability profile and publish it for the world to see, thus making communications much more efficient. In short, SIP is the protocol that supports the universal availability of presence information.

Example: A call center might use presence to ensure that a customer has the ability to get back in touch with a call center agent who was helping them with a technical support problem. With presence, the agent would not have to be in the call center, but could be located by the network and have the call routed to them, regardless of where they actually are, thus making it possible to fulfill and exceed customer service requirements without reliance on a physical call center presence (see **Figure 5**).

5: Virtual Numbers

A fifth advantage is the ability to utilize *virtual numbers*, an assignable telephone number that has no physical phone line associated with it. In most cases, virtual numbers are forwarded to either a VoIP account or to an alternate fixed or mobile number. For example, virtual numbers are perfect for sales forces, business travelers, small businesses, and field service personnel. With virtual numbers, businesses can also create a local identity in markets that the company serves.

Example: Imagine a business serving customers in multiple, far-flung locations, such as Dallas, Los Angeles, and New York City. The company seeks to create a local identity in these markets by publishing phone numbers with local area codes vs. toll free numbers. They would also like to route the call to specific individuals supporting each market. The company, which is headquartered in Chicago, might purchase virtual Direct Inward Dialing (DID) numbers from their service provider with area codes in Dallas, L.A., and NYC, giving customers the impression that the company has a "presence" in those locations — a major element of a customer-friendly contact strategy (see **Appendix A**).

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[3] Laura Devoto, Research Analyst/Enterprise Communications, Frost & Sullivan, *Branch Offices Reap the Benefits of IP Telephony*," 24 May 2007.

6: Business Continuity and Disaster Recovery

SIP trunks, in concert with VoIP, can play a major role in *business continuity and disaster recovery*. With a SIP supported IP PBX, many businesses are now able to design disaster recovery plans using plug and play phones, softphones, and IP PBX programmability capabilities. What's more, automatic reroute in IP environments is possible, thus reducing the headache of planning for every contingency. For businesses of any size, SIP trunks provide connections to the PSTN so that outbound calls can be rerouted and delivered over an Internet connection when the normal connection (or location of the connection) is unavailable.

Example: Workers can setup their laptop from home or a remote location as a virtual office, using the softphone capability that can support "presence" to detect online status of company employees, customers, and trading partners. Additionally, IP PBXs can be programmed to redirect calls in seconds to different phones or locations in the event of an outage at key company locations.

7: Supply Chain Support

The overall *supply chain* is enhanced when executed in a SIP environment. By converging its voice, data, and expanding video/rich media applications, businesses can enhance the end-to-end supply chain, thus measurably improving their effectiveness and efficiency.

Example: Mobile inventory tracking devices not only speed up the overall execution of supply chain management, but also reduce the errors associated with the logistics of tracking constantly changing inventory. Supply chains are also early adopters of Unified Communications applications.

8: Unified Communications and Conferencing Applications

Because as much as 87% of today's enterprise workforce works outside of a traditional office environment,[4] collaborative applications that overcome the challenges of distance are key. *Videoconferencing* has slowly become a part of this equation, but the cost and difficulties traditionally associated with setting it up and the need to go to a specific location to use it have been showstoppers. Thus, videoconferencing and SIP have combined forces. Not only do they offer anywhere, anytime video, but they also make it possible to incorporate presence, screen sharing, Web sharing, and instant messaging.

"Because as much as 87% of today's enterprise workforce works outside of a traditional office environment, collaborative applications that overcome the challenges of distance are key."

Example: Customers and suppliers located thousands of miles apart can use videoconferencing and collaboration tools to review and modify design specifications. Moreover, customer training and other training events can be held more frequently and effectively by leveraging rich media applications that can be delivered on-demand or in real-time.

Accelerating the growth of Unified Communications will be the adoption of Microsoft's Office Communications Server (OCS), launched in 2007. OCS is a Unified Communications client that helps people be more productive by enabling them to communicate easily with others, using a range of communication options. Among its key features are support for enhanced presence and enterprise voice capabilities, enabling users to place computer-to-computer calls and to place outbound calls to (and accept incoming calls from) traditional PBX / PSTN phone users.

What's the Unified Communications forecast? As illustrated in **Figure 1**, the entire Unified Communications services market will be $11.7 billion in 2008, growing to $24.2 billion by 2012.[5]

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[4] Nemertes, *Technology and Trends* 2004.

[5] In-Stat & Wainhouse Research, *Worldwide Unified Communications Services Forecast*, November 2007.

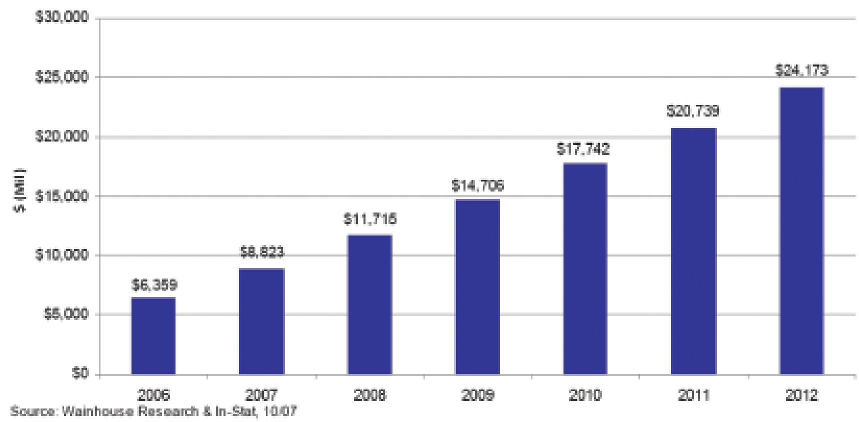

Figure 1. Unified Communications Services Forecast Summary (US$ in Millions), 2006-2012

"The entire Unified Communications services market will be $11.7 billion in 2008, growing to $24.2 billion by 2012."

9: Expanded Directory Services

Because of the ongoing effort to reconcile the various addressing schemes used to locate and identify network users, ENUM (a contraction for Tel**E**phone **NU**mber **M**apping) has come about. ENUM, supported within the SIP structure, is a collection of capabilities designed to bridge the gap between traditional E.164 telephone numbers and SIP addresses, eliminating the gap between the two.

SIP Trunking and Cost Benefits

With these advantages in mind, let's now take a closer look at SIP trunking and the cost benefits that SIP trunks make possible.

SIP Trunks

SIP trunks are one of the more remarkable offshoots of the SIP family of capabilities and one of the more important enablers of SIP-dependent applications. SIP trunks are nothing more than virtual circuits configured and delivered over an Internet connection, typically via the IP backbone of a VoIP-enabled carrier, as shown in **Figure 2**.

SIP trunks are often used in conjunction with an IP PBX as replacements or "evolutionary next stages" from traditional ISDN PRI or analog circuits. In fact, many analysts believe that SIP trunks will ultimately replace E1 and T1 facilities in business networks. SIP trunks not only make network deployment more flexible, but also make possible the seamless assurance of operational continuity in the event of a network failure. Their popularity, which is growing rapidly, is largely due to a collection of factors, including *cost savings and overall reliability*. Some of the more relevant cost benefits are as follows:

Reduced Network Service Costs

Convergence implies that a single connection can serve multiple access requirements. With SIP trunks:

- Voice and data applications ride over one IP connection, instead of separate voice and data services. As a result, businesses can save at least 25% on traditional T1 costs and Internet bandwidth costs.[6]

[6] Brian Partridge, Program Manager / Enabling Technologies Service Provider, Infrastructure Solutions, Yankee Group, *Anywhere Enterprise Market Should Provide Fertile Ground for SIP-Enabled Operator Services*, 22 August, 2007, p. 1.

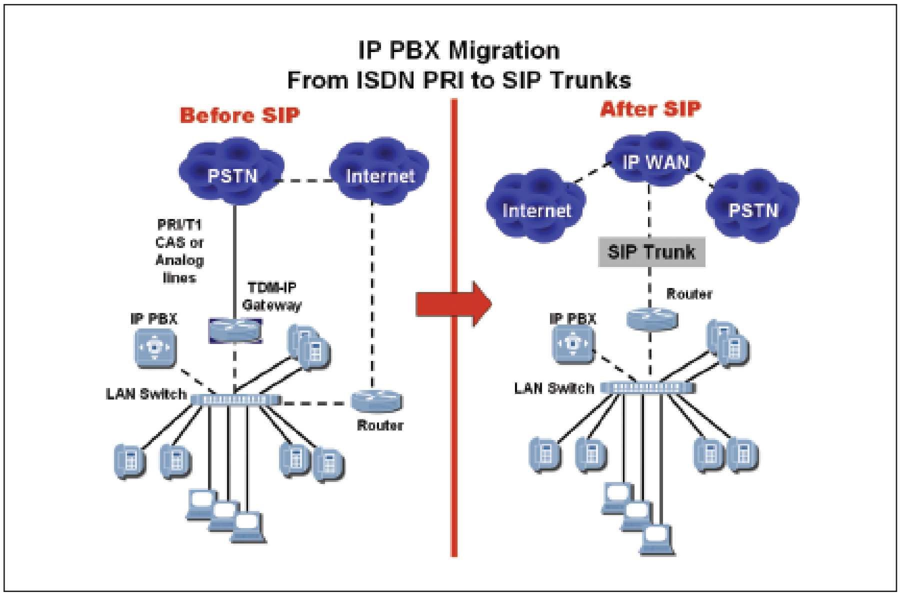

Figure 2. SIP trunks enable convergence to one IP connection over a standards-based connection, eliminating the need for TDM-IP gateways.

- The connection is highly efficient because unallocated SIP bandwidth is automatically and dynamically made available for other uses and applications as required (see **Figure 3**). Added voice compression is available from some service providers, such as XO Communications, enabling higher throughput and efficiency as well.
- On-net, 4-digit dialing plans can be established amongst a company's locations and connected via SIP trunks, resulting in lower toll costs as well (see **Appendix B**).

"With SIP trunks, bandwidth is automatically and dynamically made available for other uses and applications as required."

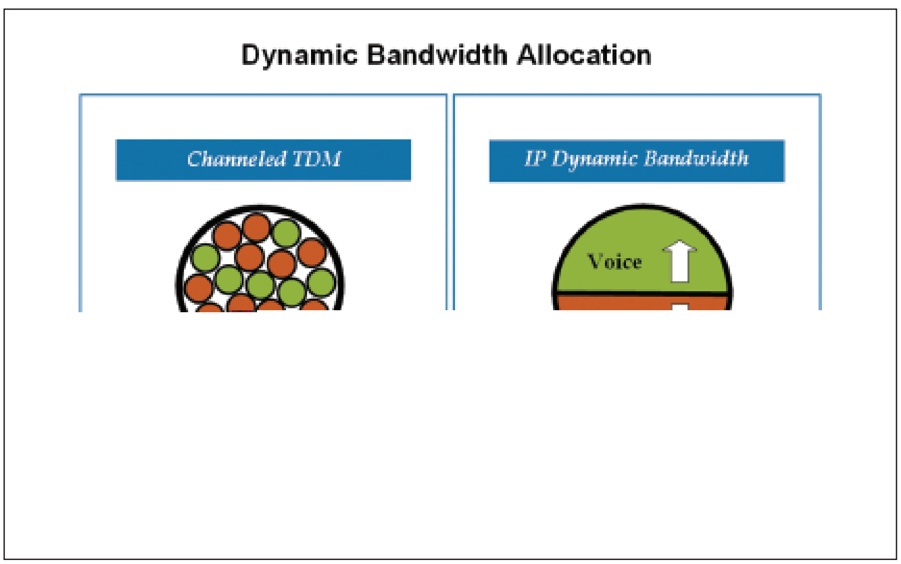

Figure 3. Real-time IP dynamic bandwidth allocation gives priority to voice traffic but makes additional data bandwidth capacity available when phone lines are not in use.

Eliminating PRI Trunks and IP Conversion Devices

SIP trunks eliminate the need for PRI PBX cards and IP conversion devices on the customer premises, typically referred to as TDM-IP *gateways*. This device supports the conversion from IP packets to PSTN traffic, which is normally transmitted over an ISDN PRI. In addition to hardware savings, better throughput is also achieved by minimizing a protocol conversion step.

Evolutionary Migration Path

Companies that are not ready to replace their legacy TDM PBXs but wish to deploy SIP trunks can deploy media gateways at branch offices to convert TDM voice communication into IP voice traffic (and vice versa), instead of installing a brand new IP system. Thus, legacy TDM PBXs placed in multiple sites are being interconnected via IP networks with the help of gateways that transform the PSTN traffic into VoIP packets.

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SIP Evolution and Standards

Enabling Wireline and Wireless Interoperability: 3GPP and IP IMS

Originally released to the world in 1996, SIP has been accepted by the Third Generation Partnership Project (3GPP) and is now a formal element in the IP Multimedia Subsystem (IMS) architecture. IMS also has been implemented by the major cellular phone providers, which is paving the way for SIP interoperability across wireline and wireless networks.

Replacing H.323 for Multimedia Traffic

H.323, an ITU standard that emerged during the rise of ISDN, is highly functional but not as popular as SIP for multimedia networks. Whereas H.323 was originally rolled out as a protocol to control the delivery of multimedia traffic on LANs, SIP was created specifically with VoIP in mind. While H.323 continues to enjoy its share of supporters in the VoIP space and is still prevalent among the installed IP PBX footprint,[7] it is slowly being edged out of the limelight as migration from ISDN to IP continues. SIP supporters claim that H.323 is far too complex and rigid to serve as a standard for basic telephony setup requirements, arguing that SIP, which is architecturally simpler and imminently extensible, is a better choice.

Enhancing SIP Trunks: SIPconnect™

Organized and maintained by the SIP Forum (www.sipforum.org), SIPconnect defines a standards-based, industry-wide solution for IP peering between SIP-enabled IP PBXs and VoIP service provider networks. It offers significant advantages and will unquestionably be a critical element in SIP's success. According to the *SIPconnect Technical Recommendation*, "SIPconnect specifies a reference architecture, specifies which VoIP protocols must be supported, provides guidance in areas where the standards leave multiple implementation options, and identifies a baseline set of features that should be supported by PBXs and service providers."[8]

Because service providers will increasingly rely on SIP trunking for their interconnect requirements, SIPconnect will play an increasingly important role because it accelerates service rollout schedules, eliminates the need for media gateways, reduces hardware and software costs, and eliminates (or greatly reduces) the complexity of the overall service rollout process.

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[7] Brian Partridge, Program Manager / Enabling Technologies Service Provider, Infrastructure Solutions, Yankee Group, *Anywhere Enterprise Market Should Provide Fertile Ground for SIP-Enabled Operator Services*, 22 August, 2007, p. 2.

[8] SIP Forum, *The SIPconnect Technical Recommendation*: *An Industry-Accepted Approach to Direct IP Peering for IP PBX and VoIP Service Provider Communications*, July 2006, p. 11.

How SIP Works

SIP is made up of a collection of functional entities that work together to provide signaling functionality across the network. These modules, shown in **Figure 4**, include the following:

- *User agent clients*, which serve as the communicating entities for the various user devices, including phones and softphones
- *Location server*, which relates a client device to a specific IP address
- *Proxy servers*, which are responsible to forward call requests from one server to another on behalf of SIP clients (hence the name proxy)
- *Redirect servers*, which transmit the called party's address back to the calling party so that the connection can be made
- *Registration servers*, which validate users on the system to ensure privacy and security

Designed with the assumption that end-user devices have a degree of intelligence greater than the core network, SIP is designed for communications among a collection of *proxy* and *location servers*. As a result, it is immensely scalable, one of the key concerns among enterprise IT personnel who see growing user populations and unbounded demand as a major challenge to their ability to satisfy QoS demands.

SIP is designed to establish peer-to-peer sessions between Internet routers (peer-to-peer simply means that the relationship between the communicating devices is non-hierarchical — they behave as peers). The protocol defines a variety of server types, including *feature servers*, *registration servers*, and *redirect servers*. SIP supports fully distributed services that reside in the actual user devices, and because it is based on existing IETF protocols, it provides a seamless integration path for voice/data integration.

"SIP is immensely scalable, one of the key concerns among enterprise IT personnel who see growing user populations and unbounded demand as a major challenge to their ability to satisfy QoS demands."

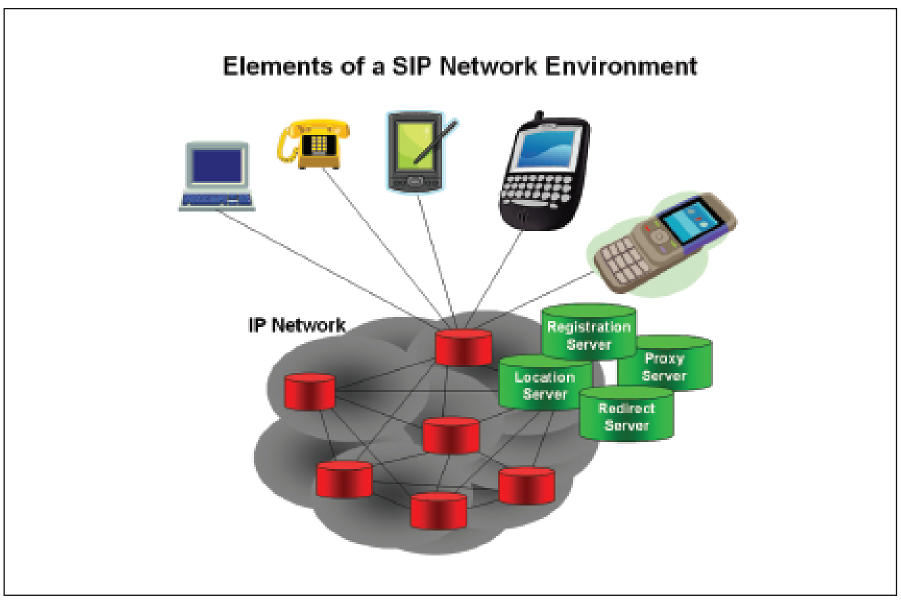


Figure 4. The SIP network environment comprises a collection of servers and agents that satisfy connectivity requirements of IP networks.

Like other VoIP-related protocols, SIP is used to set up and tear down multimedia sessions between communicating endpoints. These multimedia sessions can include multiparty conferences, telephone calls, and distribution of multimedia content. These capabilities are central to the routine care and feeding of the enterprise, not to mention the enormous boon they represent for the enterprise call center, as illustrated in **Figure 5.**

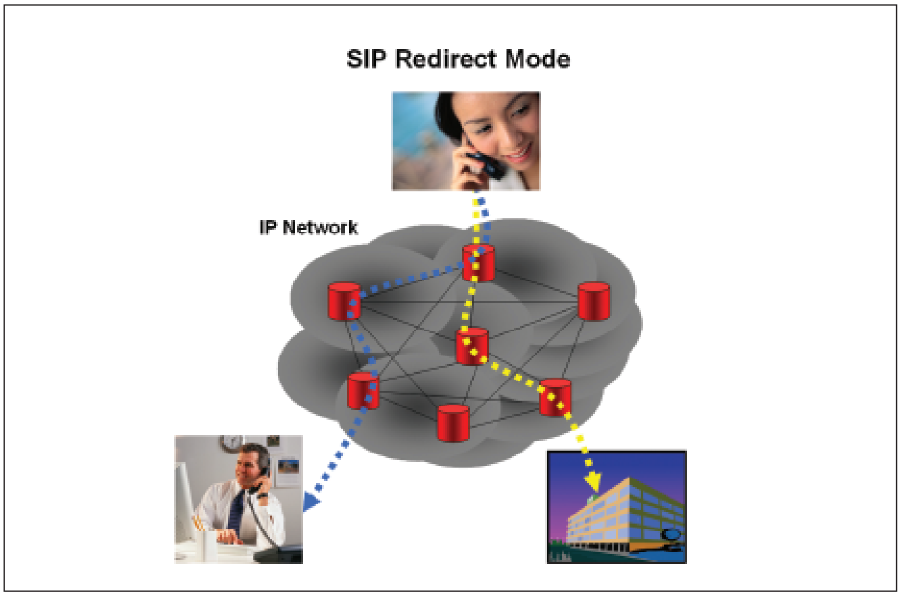


Figure 5. The caller places a call to a specific agent in call center (yellow arrow, lower right), but because the agent is out of the office, SIP routes the call to their remote location (blue arrow, lower left), thus avoiding a "service disconnect" with the customer.

Ultimately, telecommunications, like any industry, revolves around profitability. Like TCP/IP, SIP provides an open architecture that can be used by any vendor to develop products, thus ensuring multi-vendor interoperability. Because SIP has been adopted by such telecom equipment providers as Alcatel-Lucent, Cisco, Nortel, and Avaya and is designed for use in large carrier networks with potentially millions of ports, its success is reasonably assured.

Barriers to Implementation

Interoperability

As IP PBXs have come booming into the market, problems of interoperability have arisen with regard to SIP. Almost all of the first-generation IP PBXs on the market were designed around proprietary IP signaling stacks because universal agreement on a single protocol had not yet been achieved. Ultimately, SIP was chosen as that universal protocol, and PBX manufacturers wrote proprietary interfaces for their legacy TDM interfaces. This created problems for developers looking to write interfaces for VoIP environments built on media server platforms, as well as complications that required system-by-system interoperability testing or, in some cases, the creation of software interfaces to perform a protocol conversion that ensures interoperability beyond very basic connect-and-disconnect capabilities.

"Leading the effort to resolve interoperability issues is The SIP Forum (www.sipforum.org), a group of companies that tests the compliance of modules, products, and systems to SIP in order to promote the interoperability of these items with each other, helping vendors ensure high quality and interoperability."

Leading the effort to resolve interoperability issues is *The SIP Forum* (www.sipforum.org), a group of companies that tests the compliance of modules, products, and systems to SIP in order to promote the interoperability of these items with each other, helping vendors ensure high quality and interoperability (see earlier discussion of SIPconnect).

Security

Because SIP is part of the IP environment, and because VoIP uses IP networks for establishing sessions, all security threats that are known to affect IP networks have the potential to affect SIP-enabled environments as well. And because the servers and gateway devices used in IP telephony environments are complex and highly utilized, there is also the risk of buffer overflow and other vulnerabilities that result from high usage levels.

Certain vulnerabilities are somewhat specific to SIP, including SIP entity impersonation, false registration, and call hijacking, tampering with message content, unexpected tear-down of an in-progress session, denial-of-service, eavesdropping, and spam. Security specialists have taken steps to reduce the impact of these efforts; their efforts involve both logical and physical security measures, including message authentication and physical security of the devices in the network.

e911 Requirements

As discussed, SIP and Unified Communications extend the mobile nature of VoIP, by enabling communications on a range of wireline and wireless devices that are not tethered to a physical location address. With regard to access to emergency services, interconnected VoIP service providers are required to comply with enhanced 911 rules adopted by the FCC that are designed to integrate nomadic interconnected VoIP services with the existing PSTN emergency 911 system.

"Interconnected VoIP service providers are required to comply with enhanced 911 rules adopted by the FCC that are designed to integrate nomadic interconnected VoIP services with the existing PSTN emergency 911 system."

The good news is that nomadic VoIP e911 solutions are available that allow online access to quickly update location information for nomadic users so that emergency calls are directed to the proper emergency call center, regardless of the location the users are calling from, based on registered address database lookup. Other solutions are on the horizon (such as those using automatic detection of phone moves with location discovery protocols) that will enable VoIP service providers to continue to support e911 requirements as Unified Communications applications expand.

Conclusion: Making the SIP Decision

Like any large-scale technology shift, the move to SIP should be undertaken only after considering all options and determining that a SIP migration strategy is the right move for the enterprise. In most cases it will be, but it only makes sense to ask the questions. Furthermore, it is *critical* to seek the advice of SIP-seasoned professionals before undertaking the migration. The move to SIP is a relatively straightforward process. However, there is a level of complexity associated with it that demands the help of a service provider already familiar with the overall process. This is what they do — use them to your advantage.

SIP is clearly a fundamentally important technology in the evolving converged network and will play an increasingly important role in enterprise and SMB networks in the very near future. The advantages listed above and the applications described in this paper are clear indicators of its relevance as a protocol and as an enabler of service provider relevance in the future. SIP, like VoIP, is rapidly becoming a *when* rather than an *if* question in today's enterprise. Its time has come, and businesses of all sizes will be well served if they consider it as part of their evolving network strategy.

"XO Communications offers a full line of IP telephony offerings, including SIP trunks, Managed IP PBX, and VoIP solutions that integrate MPLS IP-VPN and on-net dialing plans."

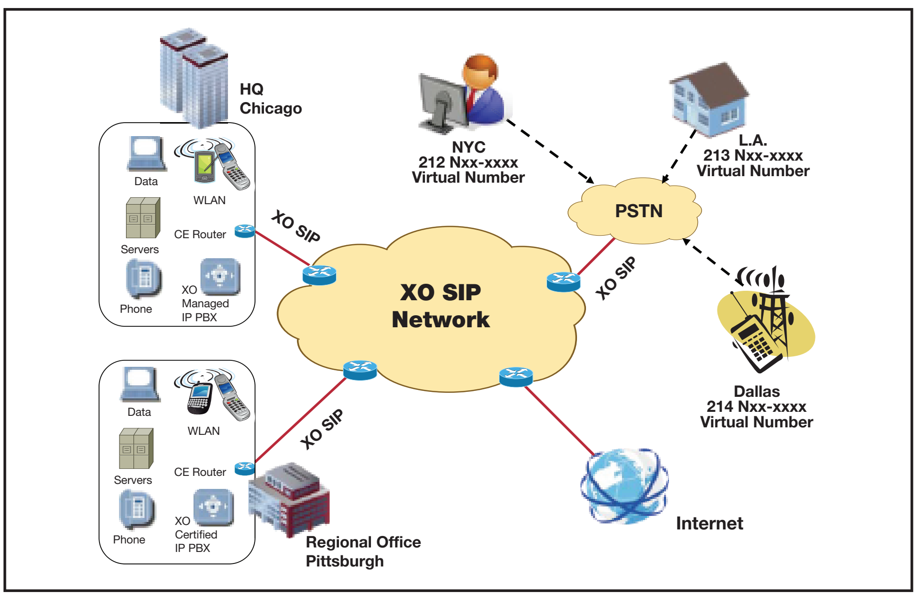
The XO Advantage

Investigate your SIP trunking options and savings potential. XO Communications offers a full line of IP telephony offerings, including SIP trunks, Managed IP PBX, and VoIP solutions that integrate MPLS IP-VPN and on-net dialing plans. The XO SIP trunking solution provides direct IP access to the XO private, OC-192 IP network, so you get the most from your IP PBX. Need to serve many markets from a single location? XO® SIP Service offers virtual inbound Direct Inward Dial numbers that allow customers outside your calling area to make local calls to reach you, giving you a local presence in markets where you want to be — one of many reasons *why SIP makes sense* in today's evolution to Unified Communications.

Appendices

Appendix A: XO IP Platform

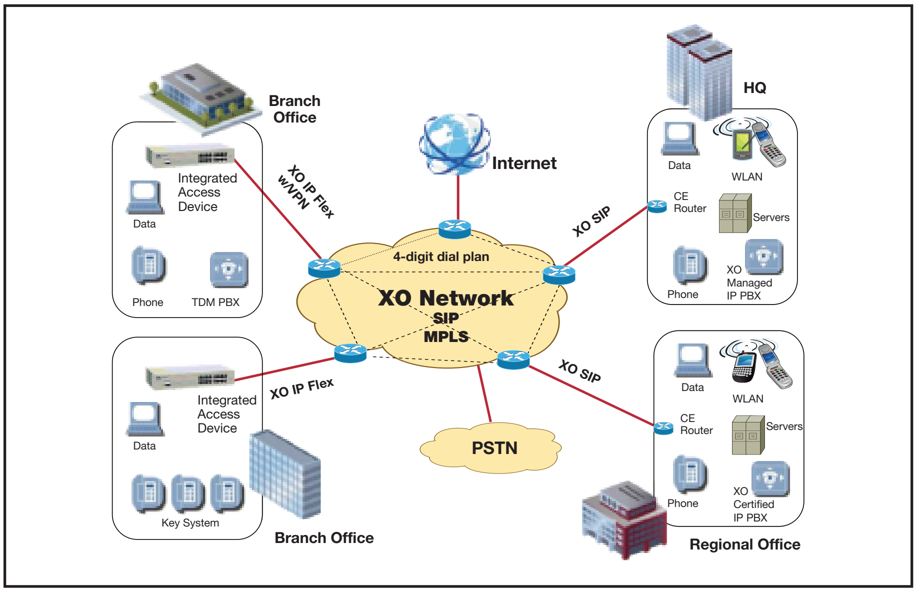
Enables Virtual Numbers with XO® SIP Service



XO® SIP Service virtual DID numbers enable a company to establish a local identity in several markets they serve. Additionally, calls can be routed to the appropriate individual assigned to each number.

Appendix B: XO IP Platform

Scalable Solutions to Support Your Enterprise Needs



From legacy key systems to IP PBX systems, XO can design a flexible, fully managed solution to meet your network migration needs, integrating a wide range of IP telephony and MPLS VPN services.

Appendix C: XO Communications Network

Serving Businesses Nationwide with a Range of Converged Services

XO operates an 18,000-route mile nationwide network that connects 75 metropolitan markets, and operates close to 900,000 miles of metro fiber.

About XO Communications

XO Communications is a leading provider of telecommunications services exclusively to businesses. XO services include local and long distance voice, dedicated Internet access, private networking, data transport, and Web hosting services, as well as bundled voice and Internet solutions. With more than a billion dollars in annualized revenue, XO is a proven provider of IP bundled services, including the award-winning Voice over Internet Protocol (VoIP) services bundle, XO® IP Flex. XO operates an 18,000-route mile nationwide network that connects 75 metropolitan markets, and operates close to 900,000 miles of metro fiber.

The XO® SIP Service is a converged solution that offers IP PBX customers direct IP access to the XO private, OC-192 IP network for data and voice communications. Plus, it eliminates the costs associated with the purchase, support, and maintenance of media gateways and reduces the recurring costs of separate PSTN and data circuits. To find out how XO can meet your specific networking requirements, visit www.xo.com or call 1.866.266.9696.

Product of the Year Award

XO® SIP Service received the 2007 Product of the Year Award from Technology Marketing Corporation's (TMC®) INTERNET TELEPHONY magazine.

About Steven Shepard

Steven Shepard is the president and founder of the Shepard Communications Group, a Vermont-based firm that provides industry analysis, technical education, management consulting, and media development to technology companies throughout the world. He can be reached at +1-802-878-0486, or via e-mail at Steve@ShepardComm.com.