SIP-O-Nomics

Saving Money and Simplifying Architecture with the Session Initiation Protocol

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Executive Summary

SIP, the Session Initiation Protocol, offers the potential to reduce telecom operational cost and complexity, take advantage of new hosted services, and integrate disparate applications via unified communications to improve collaboration. The introduction of SIP session management offers the potential to simplify communications system and policy management by fundamentally rethinking the way organizations deploy and integrate disparate communications applications.

But implementing SIP is not without challenges. IT architects must leverage solid ROI case studies to build tangible business cases to justify investment. They must also address training and interoperability concerns to ensure a successful deployment. Those organizations that meet these challenges stand to reap the benefits of SIP via delivery of new services and/or reduced operating costs.

The Issue

The emergence of SIP offers companies the opportunity to reduce telecommunications spending, streamline and optimize telecommunications architectures without the need for large-scale rip and replace of existing systems, and take advantage of unified communications to improve collaboration. SIP provides clear opportunities to:

- Leverage SIP Session Management to integrate mixed-vendor PBX environments to reduce operational complexity and management costs;
- Use SIP trunking to reduce PSTN access costs;
- Improve collaboration and productivity via unified communications to integrate voice, video, messaging, and presence with business applications.

In mid-2009 Nemertes Research and Avaya convened a "Leaders of SIP" meeting to bring together IT executives from a variety of industries, each with real world experience in benefiting from SIP-based solutions. This issue paper provides insight into the key opportunities, and challenges discussed at this meeting, as well as data and insights gathered from hundreds of IT leaders who regularly participate in Nemertes' research benchmarks.

SIP Session Management

SIP session management fundamentally changes the architecture for telecom system integration, creating a new layer to enable easy integration of disparate systems and applications. Several "Leaders of SIP" participants noted that an architecture based on SIP session management is leading to tangible benefits, in one case eliminating 90% of the cost of audio conferencing by interconnecting IP-PBXs directly with an internal conferencing platform, thus eliminating reliance on hosted conferencing services.

> *SIP Success Story:*
> Large global financial services firm
> - Used SIP to implement in-house audio conferencing solution
> - $98 million savings over seven years compared to hosted audio conferencing bridges by eliminating hosted service fees and long- distance charges for remote workers.

SIP session management addresses the challenge of operating a multi-vendor, multi-protocol telecommunications environment. Twenty-two percent of organizations (mostly larger companies) say they operate multivendor voice infrastructures.

In fact, multivendor integration is a top concern for organizations seeking to implement unified communications (Please see Figure 1: What is Lacking in the UC Market, Page 3.)

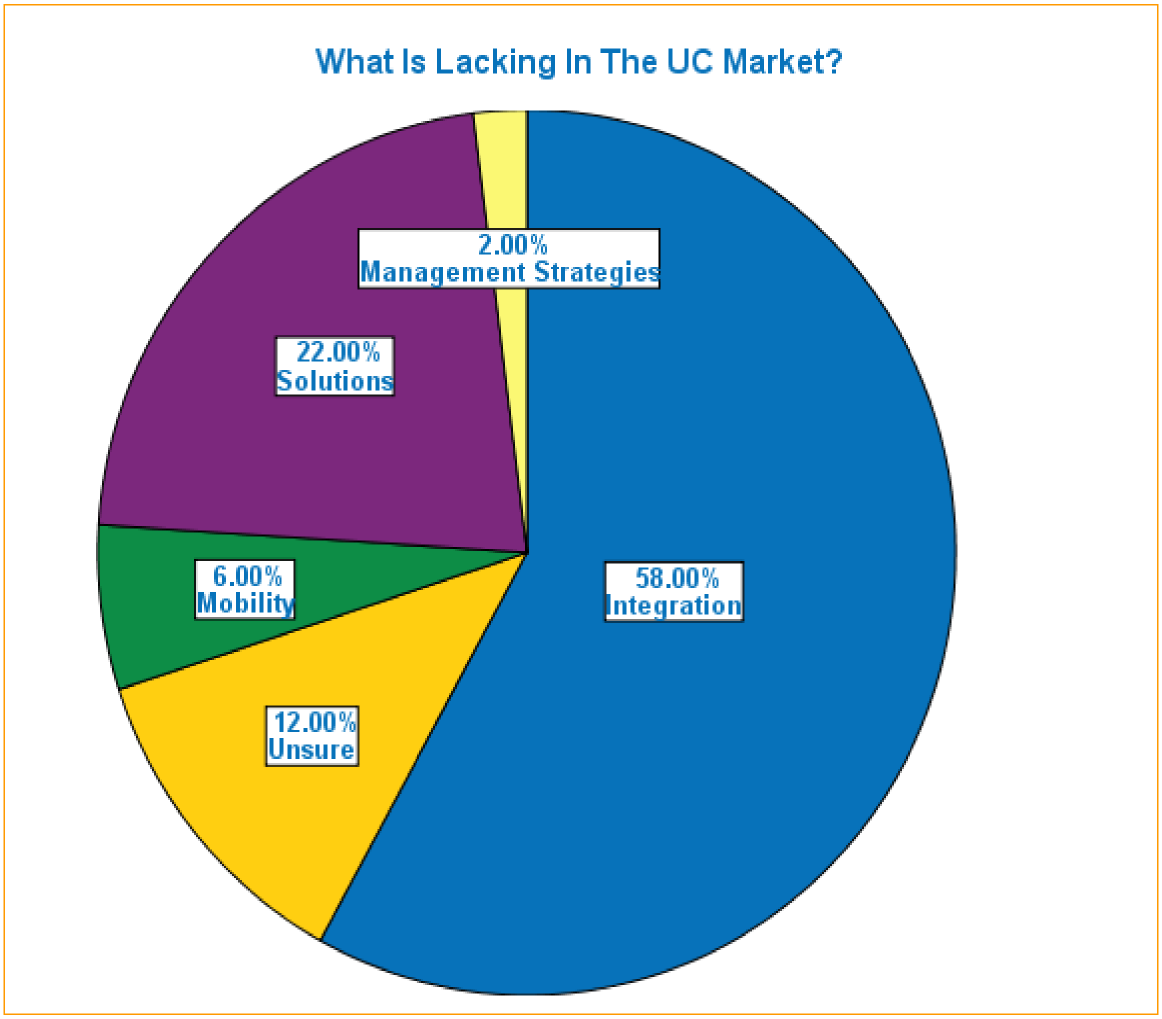

Figure 1: What is Lacking in the UC Market

SIP session management addresses these concerns by enabling easy interconnectivity of SIP-based applications, often mediating between various proprietary implementations of SIP or SIP extensions.

An architecture based on SIP session management enables easier implementation of SIP trunking by allowing IT architects to centralize SIP trunks into the data center, using the session management layer to interconnect distributed PBXs with the SIP session manager, reducing cost and complexity for SIP trunking, and more easily enabling the ability to implement policy management for inbound and outbound calls (e.g. based on load, time of day, source of call, etc.)

The Rise of SIP Trunking

SIP trunking has significant potential to reduce telecom costs and improve service flexibility. Several participants in the Leaders of SIP forum noted that they had achieved significant cost savings by using SIP trunking to replace TDM access to the PSTN. Nemertes research participants note on average, 20-60% annual savings in PSTN access costs via use of SIP trunking.

SIP trunking provides a number of architectural and cost benefits. These include the ability to load balance PSTN access across different links to improve resiliency or performance, trunk sharing to support bursting across multiple trunks during peak call times to reduce trunk costs, and integration of SIP trunking with disaster recovery plans to enable quick fail-over of PSTN access to DR locations. IT architects are also taking advantage of extranet connectivity services that deliver managed SIP interconnections for voice, video, and presence between enterprises and their partners.

> *SIP Success Story:*
> Global manufacturing and professional services firm
> - Implementing SIP trunking to eliminate PRIs for PSTN access and for trunking to mobile phones
> - 10x savings compared to legacy approach
> - Simplified management of access links.

Enterprise IT leaders are getting the message about SIP trunking. Fifty-three percent of Nemertes research participants are either using SIP trunking today, planning to use SIP trunking services in the next two years, or evaluating SIP trunking for eventual deployment (Please see Figure 2: SIP Trunking Plans, Page 5.) Larger organizations more often centralize PSTN access at the data center, often in conjunction with a VOIP roll-out, or via adoption of an architecture enabling SIP-based session management for PSTN access. Small, mid-size, and distributed organizations are more likely to implement SIP trunking either via dedicated connections to sites, or by bundling SIP trunking with other services such as Internet access.

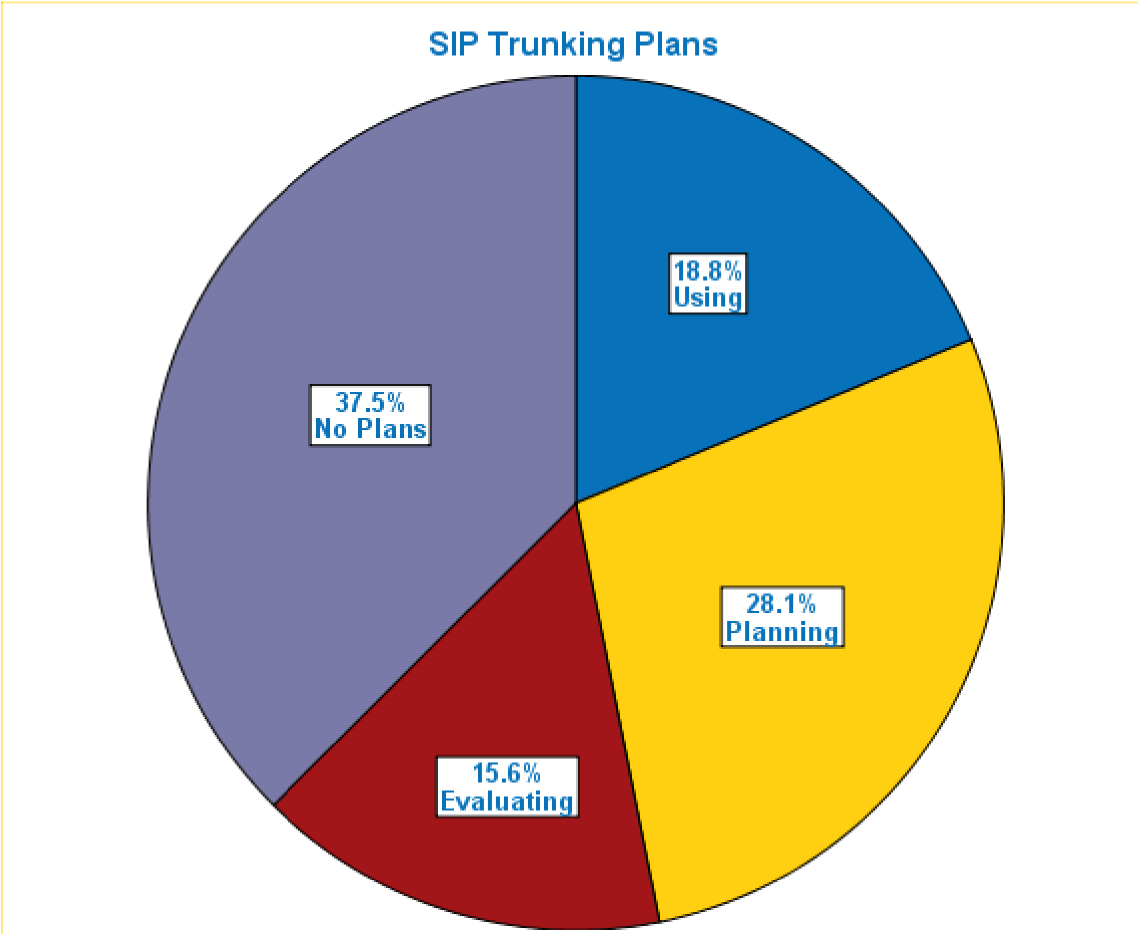

Figure 2: SIP Trunking Plans

SIP and Unified Communications

UC deployments are rapidly growing with 60% of IT practitioners saying they are implementing unified communications. (Please see Figure 4: UC Implementation Plans, Page 7.) Those deploying are seeing tangible, quantifiable benefits such as increased sales, reduced travel, more efficient use of field support personnel, and greater contact center optimization, in addition to gains in productivity.

SIP provides the "glue" for application interworking as part of a UC architecture (Please see Figure 3: Unified Communications Architecture, Page 6.) In a SIP-based UC environment, a user's desktop client or in-box shows presence status information from voice or video systems and enables the user to quickly initiate a voice, video, or web conference all through a few clicks of a mouse. SIP provides the common method for sharing session and presence information between disparate systems and applications.

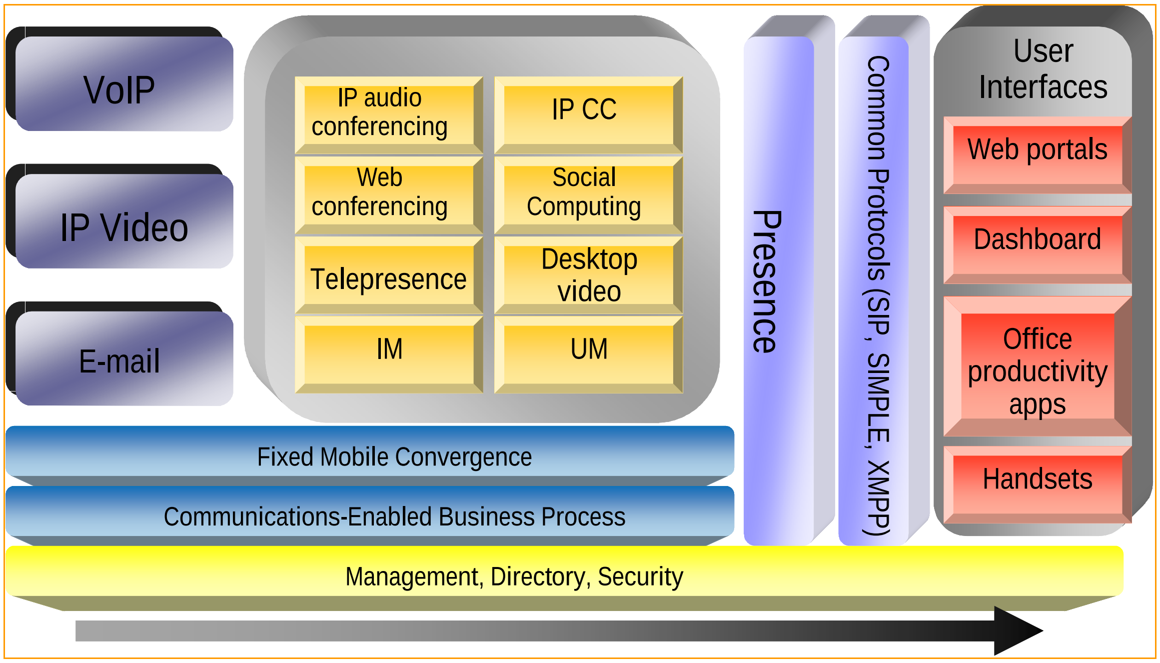

Figure 3: Unified Communications Architecture

Perhaps the holy grail of unified communications is the ability to tie communications services such as text, voice and video into business-process applications. IT architects can integrate specific business processes with their communications systems. For example, a business event such as a manufacturing alarm, inventory shortage, or medical emergency can trigger initiation of a communications session such as notification of key personnel, automatic creation of meet-me conferences, or establishment of a Web conference. In one scenario, a shortage in a warehouse could trigger an instant message to all product inventory managers, along with a proposed conference call meeting time, and additional information about the trends leading to the shortage.

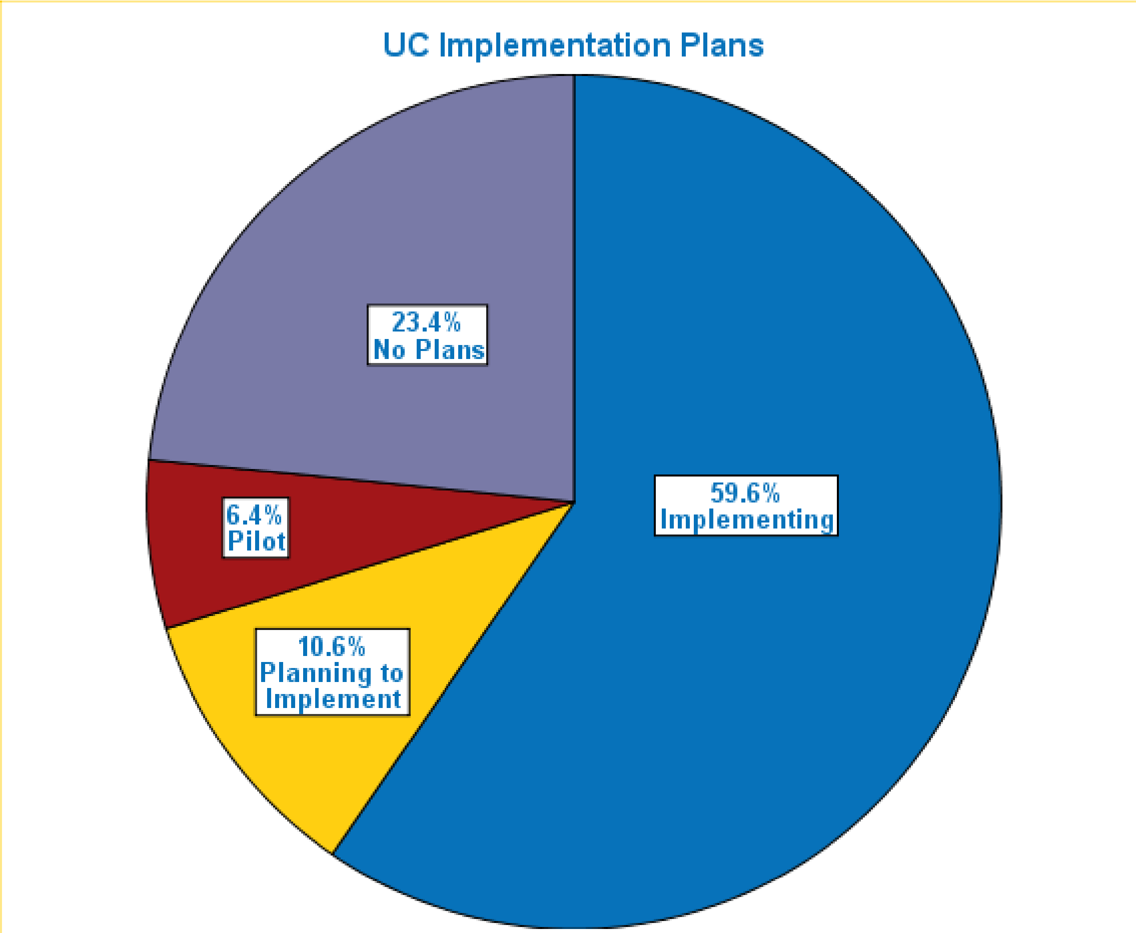

Figure 4: UC Implementation Plans

IT leaders are making short-term decisions today, with an eye toward a longer-term strategic vision for tomorrow, meaning that any tactical deployment must fit into the longer term architectural vision. One telecom manager for a large financial services firm noted that their focus is on creating an architecture, then migrating components of that architecture when and where it makes sense. For example, they may deploy unified messaging in the short term due to a need to replace aging systems or meet compliance requirements, but they must insure that whatever decision they make around UM is able to integrate with future UC plans.

Given opportunities to use UC applications such as video conferencing, web conferencing, audio conferencing, instant messaging, and contact center applications to realize tangible benefits in cost management and operational efficiencies, we expect growth of UC adoption to grow at a consistent annual rate of 10% for the next 2-3 years (Please see Figure 5: UC Adoption: 2007-2011, Page 8.)

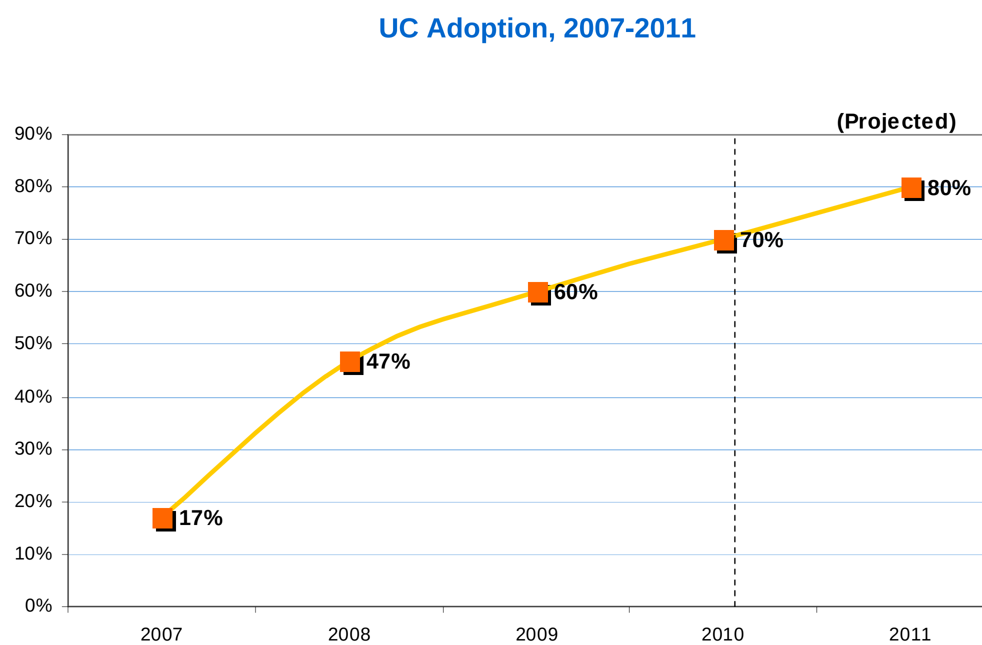

Figure 5: UC Adoption: 2007-2011

SIP Challenges and Mitigation Strategies

SIP's developers designed it as an open framework that defines the language of interconnectivity but gives developers flexibility to define the details of connectivity. This flexibility is perhaps SIP's greatest strength, but also creates significant challenges as vendors are free to implement their own primitives to define SIP-based services.

While IETF standards and the SIP Forum's SIPconnect standard offer some level of basic interoperability between compliant systems, there are still many proprietary extensions to SIP that are not yet standardized within the vendor community or associated standards bodies. The SIP Forum continues work on SIPconnect 1.1 to add additional feature standardization, but even once 1.1 is finalized, SIP's openness, and the need for vendors to deliver additional value to differentiate their products and services, means that addressing integration challenges is a key requirement in building a fully, SIP-based interconnected architecture.

"Leaders of SIP" participants, as well as Nemertes research participants, note several other challenges in implementing SIP, including:

- The need for adequate training: IT executives are experiencing challenges in finding those well trained in SIP implementation and management. Concerns extend beyond hiring their own staff to finding qualified VARs, consultants, and even vendor engineers. Successful SIP implementations require cross training between telecom and network teams to understand the inner-workings of SIP.

- Management tools: IT architects cite the need to deploy tools that allow them to manage and troubleshoot performance of SIP for both SIP trunking as well as for internetworking of SIP-based systems. In many cases engineers still rely on packet capture and manual examination of flows to determine problems, a long and arduous task that requires technicians to possesses not only a solid understanding of SIP message flows, but also vendor proprietary extensions to SIP.
- Compression: In a one-for-one replacement of TDM PSTN trunks with SIP trunks, bandwidth costs for SIP trunking can exceed TDM costs due to the additional overhead required for SIP compared with 64 Kbps channels for TDM voice. Growing use of high quality compression algorithms such as ITU G.722 and G.729 will allow SIP trunking adopters to reduce bandwidth requirements.
- Security: SIP trunking creates a new vector for attacking enterprise phone systems. In most VOIP architectures the PSTN serves as a firebreak between the enterprise phone system and the rest of the world. Risk of attack from the Internet is low as the VOIP system is physically and potentially logically isolated from the outside. Introducing SIP trunking changes this, as the enterprise phone system is now vulnerable to IP-based attacks via the SIP trunk. Session border controllers or SIP-aware firewalls can mitigate security concerns.
- Eavesdropping: VOIP traffic carried via SIP trunk across a service provider network is often not encrypted, meaning that the opportunity exists for a rogue person to listen in on private conversations via a compromised service provider network. However, this threat is no different than the risk of unauthorized interception of any unencrypted IP traffic carried across a service provider network.
- Developing a business case for SIP and UC: While determining a cost benefit for SIP trunking is fairly straightforward, only 81% of organizations have quantified the benefits of UC, even though 69% of IT executives consider a quantifiable benefit as vital or important in their UC evaluations. Those who have quantified UC benefits point to tangible returns on investment from deployments of on-premise applications, UC in the contact center, UC for field workers, reduced phone bills, travel reductions, and increases in productivity.
- Lack of usable fax services: Fax over IP has always been the thorn in the side of VOIP. While most VOIP vendors support International Telecommunications Union T.38 enable fax over IP between IP enabled fax machines and PSTN gateways, support for fax has not yet materialized in the SIP trunking market, and interoperability among T.38-based solutions is problematic. Even though fax volumes continue

to decline, fax is still a key requirement for contracts. Companies often address fax over IP by deploying fax-to-email solutions for in-bound reception, or by using scanners, or fax machines connected to POTS lines for 0utbound faxing, the later resulting in additional cost and complexity.

- E-911: While some SIP trunking providers support E-911 services, passing location information on to local PSAPs (Public Safety Answering Points), IT executives tell face significant challenges in integrating SIP trunking with their E-911 architectures. Key issues include the inability for SIP trunking providers to pass location information to E-911 call routing services. Most SIP trunking providers are limited in their service areas, meaning that they can't route E-911 calls to local emergency services offices out of their operating locations. IT architects often rely on local POTS lines for 911 access, as with fax, adding additional cost and complexity.

SIP Futures

While SIP has existed as a standard since 1999 and is now widely deployed in both service provider and enterprise networks, the uses for SIP continue to grow. Evolutions in SIP include the following:

- Enhancements to SIP trunking services: SIP trunking service providers are moving beyond simple PSTN access offerings to deliver value added features such as the ability to share trunks, burst across trunks, and connect trunks to disaster recovery locations for fast failover. Many MPLS and Ethernet service providers are now bundling SIP trunking into their WAN offerings to enable greater simplification of infrastructure by bundling services across a single physical link.
- SIP-based fixed-mobile convergence: SIP trunking service providers are rolling out the capability for wireless calls to go from their network directly to their SIP trunking customers, without the need to traverse the PSTN, offering potential for significant savings in roaming, per-minute, and long-distance cellular costs.
- SIP Extranet services: SIP trunking is moving beyond simple PSTN access to enable organizations to directly connect their SIP based infrastructures with partners, suppliers, and/or customers. SIP-based video extranet services are rapidly emerging to support telepresence and room-based video conference across company boundaries. Service providers are looking to build on these offerings to deliver additional extranet services such as SIP-base conferencing, presence sharing, and support for instant messaging federation. In addition, connecting partners via SIP-based voice connections eliminates the need to traverse the PSTN for company-to-company calls, reducing PSTN toll charges.

- Hosted SIP-based application services: Software as a service providers are increasingly delivering hosted "in-the-cloud" applications accessible via SIP. These include IVR offerings, contact center applications, unified messaging, and rich-media conferencing. IT architects can integrate their on-premise systems with cloud-based applications to deliver new capabilities with minimal up-front investment.
- Telephone Number Mapping (ENUM): The idea behind ENUM is to replace the PSTN phone number database with a SIP-based alternative, also replacing the phone system for interconnectivity between disparate phone networks with an interconnection method based on IP. ENUM appends DNS records to add a SIP uniform resource indicator (URI) to translate a phone number to/from a SIP-based address for call termination. Using ENUM, if a user dialed "1-888-241-2685", their VOIP system would attempt to resolve the number via ENUM to an IP address (in this example, 5.8.6.2.1.4.2.8.8.8.1.e164.arpa) If the resolution is successful, the user's phone system would directly connect via IP to the destination phone system, completely eliminating the need for PSTN traversal. ENUM trials and deployments are underway in many areas of the world though it will be several more years before ENUM is a standard part of the global DNS.

Conclusions and Recommendations

The rise of SIP represents an inflection point in the adoption of IP-based communications as SIP enables not only internal integration of disparate systems and applications, but also external integration with emerging public connectivity services. Adoption of SIP holds promise for simplifying network architectures, improving resiliency, and reducing operational costs. IT architects should plan for a SIP-based future for unified communications and extranet connectivity while addressing challenges around management, training and staffing, security and interoperability.

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