SIP Trunking: A Single Pipe to the Cloud

A number of years ago, industry pundits and network planners began pontificating that enterprises would soon be able to use a single pipe to the carrier cloud for all of their business traffic. Voice, video, and data would traverse a QoS-enabled IP connection from the enterprise to the carrier network and beyond, unifying communications, integrating applications, and lowering costs.

by Christian Stegh

In the meantime, enterprises have been utilizing VoIP trunking to interconnect privately owned systems for years, resulting in lower toll costs between sites. Yet these VoIP islands within separate enterprise intranets had no IP connection to the outside world; enterprises have still maintained PSTN circuits for connection to the public network.

Today, the single-pipe vision is becoming a reality, due to several factors. First, VoIP has reached critical mass in the enterprise. According to recent studies, almost 75 percent of enterprises have IP telephony deployed in their network. Service providers see this as a significant opportunity for IP trunking services. Looking for new revenue streams and competitive differentiators, carriers are adapting their mature wholesale IP peering services into retail trunking services. But the most important enabler is the availability of an open, mature protocol: SIP (Session Initiation Protocol).

This article outlines the activities, progress, and next steps in standardizing the SIP interface between service providers and enterprises.

Just What is SIP Trunking?

SIP trunks are VoIP trunks from service providers that use SIP for call control and routing, enabling enterprises to create a single, pure IP connection to carrier clouds. An enterprise SIP proxy peers with a carrier SIP proxy, with the appropriate federations and security protections between them. Voice is simply layered on top of the network as another IP application, traversing routers, switches, and border elements. SIP sets up and tears down voice calls over this logical trunk, from where the call is routed via the carrier's IP backbone to its destination. Calls are handed off to a PSTN gateway for the last mile if they are bound for E.164 numbers. While the theory is simple, it has taken time and extra effort within the standards bodies to make this happen.

Why SIP Trunking Makes Sense

SIP trunks have tangible benefits in hard cost savings and operational efficiencies for both enterprises and carriers.

For enterprises, SIP networking means reducing the monthly recurring cost of separate PSTN & data circuits to the premise. When circuits are removed, the number of TDM T1 interfaces on the IP PBX is reduced, since hundreds of VoIP calls can come from the same hardware footprint as a single T1 interface. Finally, service providers offer reduced toll charges to customers when SIP is used as the interface to the PSTN.

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For service providers, the efficiencies of converged networks are driving them to extend their QoS-enabled backbones to the enterprise premises in the form of retail SIP trunking services. They're able to reduce the cost to the customer by leveraging shared infrastructures, instead of maintaining

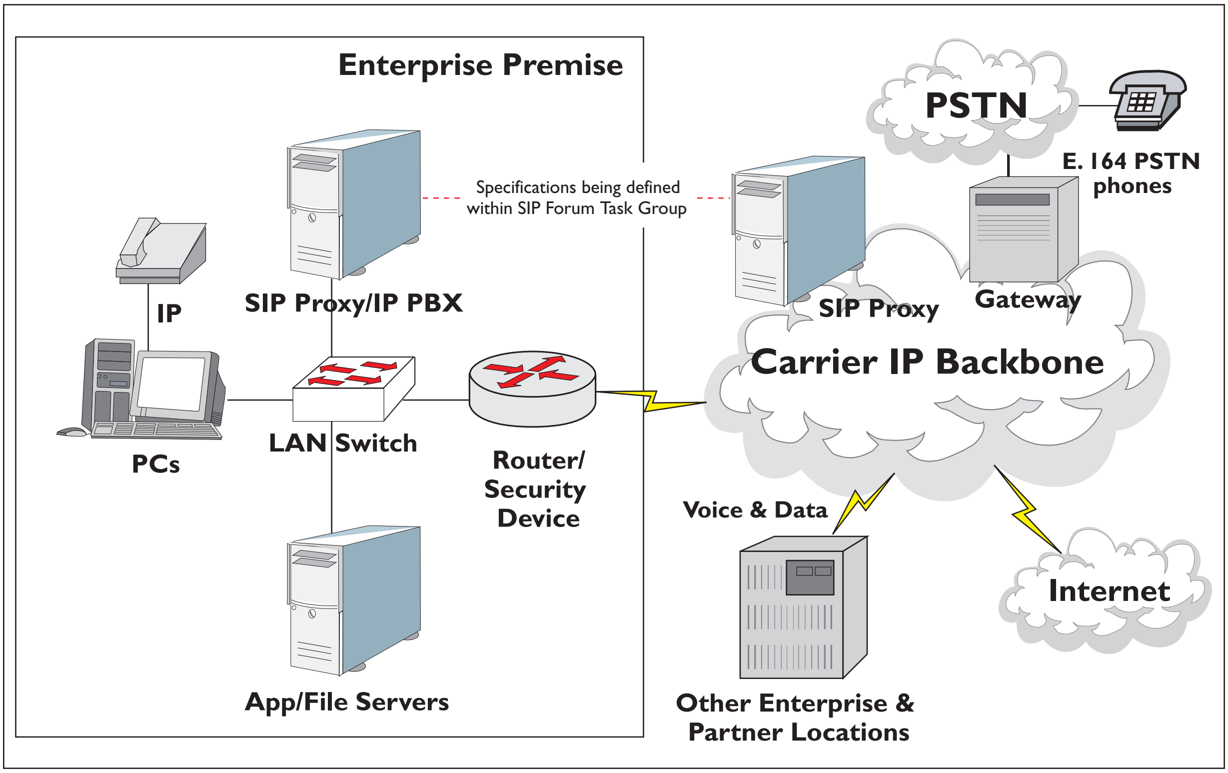

Figure One: SIP trunking enables enterprises to create a single, pure IP connection to carrier clouds

both TDM and IP equipment. Carriers leverage shared equipment to offer services to many clients, dispersing their investment and offering bundled services. A driver for many is that enterprise VoIP services allow them to enter a new market: traditional data carriers or start-ups can begin offering voice termination and origination services over their existing pipes for new revenue streams. Since SIP is an application layer protocol transparent to the underlying data link topology, trunks can ride over any suitable Layer 2 protocol, such as MPLS or Metro Ethernet.

Early Lessons from the Field

Early efforts to create peer relationships between SIP PBXs and carrier proxies have had mixed results. This is partly due to the very reason that SIP is ideally suited for inter-domain multi-vendor communications: its flexibility. Because of the abundance of IETF drafts from which vendors can draw from in crafting their SIP-based solutions, implementations can vary widely. Thus, while the resulting multi-vendor deployments typically enjoy a high-degree of basic interoperability, they often lack consistent means for delivering mission critical capabilities such as fax or security. In some cases, for instance, all SIP signaling was considered trusted (no TLS working across the boundary), and the identification and personalized services delivered to enterprise users was limited when compared with current services.

Industry Leaders Respond

In February 2005, a consortium of companies with a common interest in the success of SIP trunking took action. Service providers and IP PBX/carrier equipment manufacturers published the industry's first specification solely focused on the peering interface between service providers and IP-PBXs. This effort, known as SIPconnect, complements and builds upon existing IETF standards. It specifies a reference architecture, required protocols and features, and provides implementation rules.

VoIP has reached critical mass in the enterprise

Since the success of SIPconnect hinged on broad industry support, the founding members sought an established industry forum by which to further work in this area and gain consensus. The members submitted a proposal to the SIP Forum, and by July 2005, the "IP PBX and Service Provider Interoperability Task Group" was created as a SIP Forum Technical Working Group.

The Task Group is now working to define the protocol support, implementation rules, and features required to enable direct IP peering between SIP-enabled IP PBXs and SIP-enabled VoIP service providers and provide robust support for personalized end-user services, according to the group's charter.

The specific areas being worked by the group include:

- The development of a reference architecture of common network elements
- The specification of the basic protocols (and protocol extensions) that must be supported by each element of the reference architecture
- The specification of the exact standards associated with these protocols
- The specification of standard methods for negotiating protocols, protocol extensions, and exchanging capability information between endpoints
- The definition of authentication methods to ensure user security and accurate billing

The specification will also define requirements for CODEC support, the handling of fax and modem transmissions, echo cancellation, transporting DTMF tones, and the propagation of message waiting indication information.

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While the above set of specifications seems comprehensive, it remains fairly generic in that it addresses the variability among implementations that can be seen in almost any deployment scenario, not just SIP trunking. So the working group output is also expected to specify a basic set of required features, and provide guidelines for working through Network Address Translation (NAT) and/or packet filtering devices. Additionally, a security model will seek to limit fraud and ensure privacy.

Standards Progress Report

The SIP Forum has been making rapid progress; work is being done now on a third draft. Visit *www.sipforum.org* for more information. A fourth revision of the specification is expected by February 2006, which will likely be opened for last call IETF-wide.

This working group's main focus remains on peering between an enterprise IP PBX and a carrier's SIP equipment. Other business and technical models, such as hosted services, are not specifically in scope.

Typically, hosted services comprise a single vendor's solution, with call processing and/or applications managed at a service provider facility.

In such cases, the multi-vendor interoperability requirements are less complicated and the provider manages most components themselves, whereas in the peering model, the enterprise manages their IP PBX and leverages the service provider's SIP network for media transport.

Nonetheless, the security and QoS requirements in the hosted model are similar to those being addressed by the SIP Forum work group.

In the meantime, a growing number of service providers are now offering SIP networking services for enterprises, and some IP PBX manufacturers are enabling the solution with premise-based SIP equipment.

Enterprises are considering SIP trunking services as an important new option in networking capabilities that offers the following:

- New efficiencies in network design
- Cost reductions for WAN networking and long-distance calling
- Ability to enable a new wave of SIP-enabled applications

This type of offering will become more pervasive as the Task Group's work matures.

As it does, the integration challenges will diminish, as will the time and cost of implementation. At that time, businesses small and large should begin to see more options for QoS-enabled VoIP services over the same data circuit they lease from their service providers, yet still own and manage their own IP PBX.

Future Opportunities and Scope

Because SIP's design and interoperability enable it to play a role in applications and services beyond telephony, the SIP Forum is planning to broaden the scope of its efforts. Once the basics of the interface between the IP PBX and the service provider become firm, the SIP Forum will likely look to tackle the issue of how advanced SIP services, such as instant messaging and presence, are handled over the SIP trunking interface.

The vision is for the same SIP trunk interface to allow mobile user agents to connect through service provider networks back to the enterprise, in order to gain access to advanced applications and services. While these services are not excluded from the current specification, they are not the primary focus today.

Just as enterprises did with VoIP in its infancy, they can now begin to leverage SIP for cost savings and network efficiencies. Then, as standards emerge and enterprises begin to embed communications into their business processes, they can start to leverage more advanced solutions that integrate SIP and presence into business applications. The foundations being laid today will have an impact for years to come. [VM]

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