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SIP Trunking: Direct Connection To Everything IP

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Real-time person-to-person communication is fast becoming a critical communications tool for enterprises of all sizes. SIP, or Session Initiation Protocol, has recently emerged as the global standard for using the Internet for real-time person-to-person communication applications such as Voice-over-IP (VoIP) telephony, instant messaging, video, and presence. SIP trunking – despite the technically bewildering name – can be a valuable communications service alternative for many enterprises. To help Canadian organizations consider SIP trunks as a new alternative in their communication services IDC analyst Lawrence Surtees addressed important questions on the topic.

Q. What is SIP?

A. SIP, or Session Initiation Protocol, is a standard for using the Internet for real-time person-to-person communications. Originally conceived as simply a way to start sessions between users on the Internet, it has since become the basis for a wide variety of communications applications on the Web, including: basic voice telephony; presence; instant messaging; multimedia and video conferencing; and data collaboration.

Technically, the SIP protocol is an application layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. It is defined by the Internet Engineering Task Force as part of the RFC 3261 standard (at **www.ietf.org**). The global standardization of SIP by the IETF and its widespread embrace has made SIP an integral component of unified communications offerings.

Q. What is SIP trunking and are there different ways of using it?

A. SIP trunking is a term applied to the services offered by any communications service provider to terminate Voice over IP (VoIP) calls to the Public Switched Telephone Network (PSTN). SIP trunking allows large enterprises and small businesses alike to eliminate a PSTN gateway at their site and outsource that function to a carrier.

SIP trunks displace circuit-based connections and replace more costly leased lines connecting distributed telephone systems within an enterprise. SIP trunks typically are adopted as a lower-cost alternative to Primary Rate Interfaces (PRIs) because SIP trunks can be purchased in single-trunk increments (as compared to 23 channel increments for a PRI).

SIP trunking services can be delivered and grouped in three categories: SIP trunks via dedicated lines, SIP trunks in conjunction with hosted services, and pure SIP trunking providers.

1. *Dedicated Lines:* Any SIP service over the public Internet requiring a circuit such as a PRI, T1 or DSL connection. It allows any enterprise, anywhere, to adopt SIP Trunking and assign some, possibly unused, bandwidth to voice at no extra charge for the connection, and providing the highest ROI.

2. *Hosted (Managed) Services*: Carriers supply a dedicated, fully managed connection from their local Points of Presence (POP) from which "last mile" access is provided to the business customer's site or sites. Because the service provider has end-to-end control of the SIP trunk, this service offers quality of service (QoS) guarantees, but is somewhat more expensive.

3. *Pure SIP Trunk providers:* Service provider that has created an underlying network architecture focused on the delivery of SIP trunking, typically Multi-Protocol Label Switching (MPLS)-based to insure the highest voice quality and reliability. These organizations do not compete with premise equipment providers, do not require a dedicated circuit, and generally cover a wide range of locations.

Q. Does SIP trunking differ from Voice-over-IP telephony?

IP telephony – where ordinary telephony is emulated over IP – represents only a fraction of the capabilities of the type of person-to-person communication for which SIP was created. Many IT vendors are developing and selling products and software that take advantage of the SIP standard: SIP telephones, PC clients (such as Windows Messenger), SIP servers, and routers and firewalls that handle SIP.

Managed SIP trunking services also enable enterprise-grade QoS guarantees to be provided unlike some plain vanilla VoIP services.

These are just the beginning of what we can expect from companies eager to help bring SIP to the enterprise.

Q. Are Canadian firms considering SIP trunking? What are the benefits of SIP for business customers?

A. SIP trunking continues to gain momentum throughout North America. There is no question that SIP trunking offers compelling advantages for businesses large and small. Typical savings of a SIP trunk over PRIs range from between 40-60 per cent with the payback period for the equipment required, which may include an upgrade to the IP-PBX, shown to range from four to 12 months.

The term may be a confusing mouthful but SIP trunks provide an elegant architecture for businesses with other associated technical benefits:

- A single network can be maintained within the organization, rather than having both a voice and data network.

- Internet bandwidth can be used more efficiently. When a SIP trunk is not being used, the bandwidth otherwise allocated to a SIP trunk is freed up for use in less intensive applications, such as e-mail and general web use. This dynamic allocation of bandwidth is another key attribute that distinguishes SIP trunks from analog circuits or PRIs.

- Moves, Adds and Changes (MACs) can be completed without major wiring upgrades.
- SIP addressing is simplified with the use of email addresses as the SIP address. This is unlike traditional IP telephony systems, where SIP addresses are mapped to a telephone number.

The geographic placement of enterprise network resources also becomes irrelevant with SIP. Moving voice to IP networks with SIP trunks allows the deployment of a distributed enterprise PBX that frees a PBX from the geographic constraint of being tied to a specific local DID numbering plan. With VoIP, there are no restrictions on the inbound DID calls that a specific VoIP signaling device can receive. This distributed architecture provides greater convenience than full Centrex at a fraction of the cost, and with none of the geographical limitations.

Q. What do we need to proceed with a SIP trunking service?

A. There are two basic components within SIP: the SIP user agent and the SIP network server. The user agent is the end system component for the call and the SIP server is the network device that handles the signalling associated with multiple calls. SIP trunks require a SIP-ready, or IP-enabled, PBX which is essentially a network server. SIP trunks can also be made to work with traditional analog or key systems with an IAD (Integrated Access Device). The advent of RFC 3261 and SIP Connect standards have addressed interoperability issues between SIP trunk providers and underlying equipment manufacturers.

SIP-capable edge devices are an essential requirement. Something has to be built into or supported by the SIP endpoint in order for call features to work — whether the end user gets involved with it or not. An endpoint-based SIP feature is also necessary for the endpoint to interact with central call control, or with another endpoint.

SIP features may be thought of in terms of endpoint-based, PBX (or call controller)-based and trunk based. Yet this characterization is not rigid, since a feature can involve both a PBX and endpoints. Many SIP features — such as Transport Layer Security (TLS)-based encryption of SIP call control — involve both SIP call controller and SIP endpoints. The feature will not work unless both ends support it, and implement it in a consistent and interoperable manner.

Session border controllers, or SBCs, are SIP networking devices that mediate “peering” and are needed where two SIP domains meet – whether between two enterprise organizations or between an enterprise and a SIP-based carrier. Usually a service provider can take care of the SBC along with provision of a leased SIP trunk.

Q. Are there pitfalls with SIP trunking to worry about?

A. For businesses considering joining the growing SIP user community, it is important to ensure that their enterprise network is adequately prepared and safeguarded.

As with any technology implementation, careful planning and pilot testing is key to the successful adoption of SIP. For example, sharing the same bearer circuit for voice calls and data can raise its own challenges in maintaining call quality. To mitigate this, many companies split voice and data traffic into two separate Internet connections so that the resource conflict on the Internet access side is avoided. Using a managed IP service that provides multiple Quality of Service (QoS) levels on a SIP trunk also eliminates this hassle as well as allows for service level agreement (SLA) guarantees to be tied to the SIP trunk service.

And using this technology safely also means being vigilant about security.

When connecting a PC or PBX to the Internet, it is imperative to safeguard the system from hacker attacks and other unwanted access intrusions. This is especially critical if the PC is constantly connected via broadband or a fixed line. A firewall protects the PC by rejecting attacks and illegal data packets, allowing only approved traffic.

The most frequently cited concern about SIP adoption has been firewall traversal. The biggest hurdle for IT managers looking to SIP-enable their network is prepping the system to handle the traversal of this type of traffic. Many firewalls and network address translation (NAT)-routers were not designed to handle person-to-person communication on the LANs unless the enterprise firewall has specific SIP support. It is critical that IT managers evaluate their current firewall solution to ensure there is proper SIP support when new firewalls and NAT routers are installed. Use of SBCs or including a SIP proxy and SIP registrar for controlling the NAT and firewall makes it possible to handle complex SIP scenarios and even use TLS for secure and private signalling.

Another hurdle with SIP trunking arises from deployment and management of equipment from different vendors. As in any unified environment, mixing equipment based on the SIP protocol from different vendors can cause interoperability problems stemming from incompatibility. SIP Forum, an IP communications industry association, has introduced a recommended set of industry interoperability guidelines – dubbed SIPconnect - designed to facilitate the connection between SIP-enabled IP PBXs and SIP-enabled service provider networks and to establish industry-wide norms. It specifies VoIP protocols/features, a reference architecture and implementation rules. By using SIPconnect, enterprises can solve interoperability problems.

Finally, vendors tout a bewildering array of SIP-based features. Comparing SIP systems based on supported SIP features can be frustrating and misleading especially because features are implemented in various ways and with varying prospects for third-party interoperability. Think about what you really need.

SIP trunks are here to stay. For businesses who don't want a customer-owned equipment solution, SIP trunks will ultimately also be at the heart of new all-inclusive business service bundles with a new pricing paradigm that provides all local connections, unlimited long distance calling, and a set number of high-speed Internet connections for a single, flat monthly rate — all on a single bill.

ABOUT THIS ANALYST

Lawrence Surtees is Vice-President & Principal Analyst, Communications Research, at IDC Canada Ltd. Lawrence manages IDC Canada's communications research agenda and is IDC's lead analyst covering the Canadian telecom services sector, including the wireline, wireless and Internet segments. Lawrence also works on related IDC consulting projects. Widely regarded as one of Canada's foremost experts on the telecommunications sector, he has covered the telecom sector including both service providers and equipment makers, as well as related regulatory and policy issues, for 30 years. Prior to joining IDC Canada in Sept. 2000, Lawrence spent 17 years as a reporter at The Globe and Mail newspaper in Toronto where the bulk of his tenure was spent on the Report on Business covering telecommunications and

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