The Essential Guide to SIP Trunking

What you need to know about SIP Trunking and how it can benefit your business.

by Cindy Waxer | VoIP News

An SIP (Session Initiation Protocol) trunk is a service that allows businesses with an installed PBX to use real-time communications including VoIP. By connecting an SIP trunk to an internal traditional PSTN (public switched telephone network) phone system, companies can communicate over IP outside the enterprise. What's more, companies can replace traditional fixed PSTN lines with an IP phone system connected externally through an SIP-trunking service, thereby creating a single conduit for multimedia components including voice, video and data. As a result, an SIP-trunking service typically delivers greater cost savings and increased reliability.

How SIP Trunking Works

SIP trunking blends connections for both data and voice into a single line. By serving as a converter between a legacy phone system and a company's Internet connection, an SIP-trunking device allows the data network to carry voice traffic. SIP-trunking features typically include local and long-distance calling, E911, directory listing and caller ID, all of which integrate into a company's existing phone system.

Here's a hypothetical example of how SIP trunking works: A San Francisco-based sales representative places a long-distance call to a client based in New York City using a dialing prefix and the local-area phone number. The call either originates as an IP call or is converted to one before it leaves the office and then travels the majority of the way over the IP network of the service provider, then drops back down to the PSTN once it reaches its termination point. Since a sizable portion of the call traveled over the IP network at no additional cost rather than on the PSTN, the service provider can (and does) charge a mere fraction of what the traditional fee would be without the IP connection.

SIP trunking consists of three primary components. These include:

IP PBX: An IP-based PBX communicates with all endpoints over an IP network. It also switches calls between VoIP users on local lines while allowing all users to share a number of external phone lines. The typical IP PBX can additionally switch calls between a VoIP user and a traditional-telephone user, or between two traditional-telephone users in the same way that a conventional PBX does.

ITSP (Internet telephony service provider): The role of an ITSP is to ensure connectivity to the PSTN from an IP network for mobile and fixed communication devices. It also transports IP communications across a private IP network or public Internet.

Border element: The border element facilitates connectivity between an enterprise IP network, the PSTN and an external IP-carrier network. The border element may be an SIP-capable firewall or a SIP-enabling edge device connected to the firewall, or it can be a switch to transfer calls into and out of the PSTN. Border elements are usually managed by the service provider.

Note that a service provider will typically include SIP trunking as an element of an IP-phone-system package; it will only be separated into hybrid solutions when there is a need to maintain some locations on a traditional TDM (time-division multiplexing) system, which will connect to the overall IP infrastructure through various border elements and an SIP-trunking service.

Benefits

An SIP-trunking service benefits companies by:

- Eliminating the need to invest in costly (and less capable) TDM-gateway equipment infrastructure or desktop equipment
- Nullifying the need to purchase equipment, such as managed-media gateway devices, to interface between IP voice and the PSTN
- Reducing monthly expenses, since only one connection for data and voice is needed
- Eliminating the need for PRI (primary rate interface)/BRI (basic rate interface) connections, lowering telephony costs
- Allowing companies to outsource their PSTN connectivity to a third party, reducing long-distance charges
- Providing points of presence in multiple U.S. cities so that companies can establish local numbers rather than rely on a 1-800 number
- Accessing the benefits of a hosted VoIP service, without discarding existing investments in a traditional phone system

Costs

Although SIP-trunking service providers offer many options at varying costs, it is not uncommon for companies to realize instant cost savings from a relatively modest investment in SIP-based technology. Certainly, SIP requires purchasing services and products including an IP PBX, IP phones, soft clients and SIP-friendly firewalls, to name a few out-of-pocket expenses. But if deployed properly, SIP trunking can produce a healthy return on investments in less than six months.

Challenges

Whenever companies attempt to mix equipment from different vendors in a unified IT environment, problems can arise. Various equipment based on the SIP protocol is certainly no exception. That's why companies should establish best practices for interfacing a PBX implementation with an ITSP to minimize incompatibilities.

SIP trunking can also give rise to security risks. Fortunately, SIP-server and -proxy technologies can manage the flow of SIP traffic, enabling an IT manager to ensure correct routing, apply verification and authentication policies and minimize threats.

As for quality, companies would be wise to establish quality-of-service agreements with their SIP-trunking service providers before signing on the dotted line. Fortunately, measures such as the proper provisioning of links and prioritizing voice traffic can help eliminate delays and allow for proper scaling. What's more, a network-monitoring solution can help find the best real-time traffic route for greater quality and reliability.