

SIP Trunking: The Next-Generation IP Communications Option

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Introduction

There was a time when a company seeking multiple voice lines and high-speed data transfer for its Private Branch Exchange (PBX) was limited to a few local service providers. The services offered were typically a T-1 circuit with communications routed through a central office trunk or Primary Rate Interface (PRI) circuits.

But with the latest advancements in Voice over IP technology, there are now real alternatives that combine managed IP-based multiline voice services with high-speed data transfer, easily outpacing the functionality of the traditional model. At the forefront of this technology is Session Initiation Protocol, or SIP Trunking, offering tailored, cost-efficient voice and data services with all of the traditional trunk features. SIP Trunking is spreading fast and already services thousands of enterprise clients.

SIP Trunking surpasses the traditional trunk in many ways, including scalable features like "Bursting" and end-user mobility options that enhance and integrate corporate communications.

What is SIP Trunking?

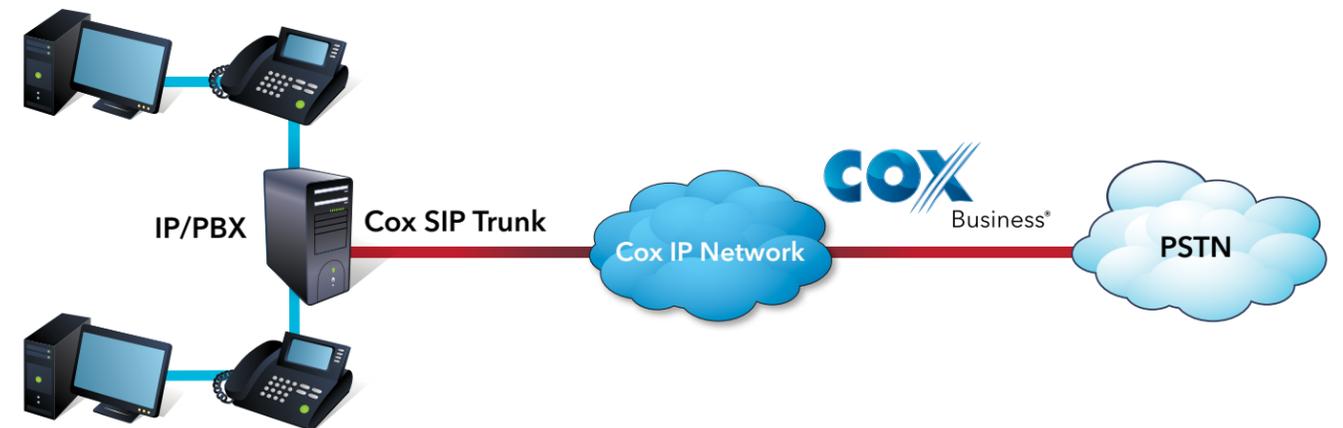
SIP = Session Initiation Protocol

SIP Trunking allows a customer's IP PBX to communicate with the Public Switched Telephone Network (PSTN), which is the traditional communication network that enables calls between users everywhere.

At the heart of SIP Trunking is Session Initiation Protocol, a signaling protocol that manages the initiation, modification and termination of IP communications sessions. A session can include communications like telephony, data transfer or multimedia.

The result is an IP version of Primary Rate Interface (PRI) trunks, the familiar Time Division Multiplexing (TDM) circuit bundles used to feed multiline PBX systems. But instead of using physical trunks with copper wires, SIP Trunking establishes virtual IP channels for real-time communications over an Ethernet connection, be it voice, data, video or messaging.

A SIP trunk is provisioned between the service provider's network and an IP PBX much in the way a Primary Rate Interface (PRI) trunk is established to feed a traditional PBX. As with PRI service, SIP Trunking can also support direct inward dialing (DID) and call features such as Caller Name and Number. Additionally, a variety of devices can be used with a SIP trunk, ranging from a desktop computer to a traditional voice phone.



Market Growth

Boosted by a larger, ongoing transition toward IP-based services, SIP Trunking is gaining momentum among enterprise customers. A recent study by Infonetics Research projected that SIP trunking would rise to second place behind T-1 lines as the most common trunking service. The research firm's survey of 92 companies with more than 100 employees found that 39% of those surveyed are deploying SIP trunks. Based on that research, Infonetics also projects that SIP Trunking revenue will increase at a compound annual growth rate of 89% through 2013. Similarly, research firm Frost and Sullivan estimated that by the end of 2009 the SIP Trunking user base in North America had risen 40.1% year-over-year to 3.8 million users, and projected the user base will grow at a 42.8% compound annual growth rate to reach 46 million users by 2016.

These studies clearly indicate that the migration to SIP Trunking is well under way. Most voice equipment manufacturers have retooled their operations to focus solely on building SIP-based units that more easily integrate voice and data services. Manufacturing of traditional communication equipment is expected to decline, and businesses may find it more costly to maintain and operate these systems as vendors migrate to SIP-based products.

Benefits to the Subscriber

Customized Configuration

For businesses who are considering a move to SIP Trunking versus traditional PRI service, some of the advantages to SIP Trunking are the flexibility and advanced features uniquely associated with this service.

While a standard switched PRI is usually configured for 23 channels and can mix data and voice in a converged delivery system, there can be restrictions to the number of channels it can offer and the services those channels can support.

And if a business requires more than one but less than two PRIs, e.g., 27 voice channels, they will likely have to buy two fully provisioned PRIs and bear the added cost of additional connectivity equipment and support, even though they don't need all 46 channels. In contrast, a business capable of leveraging SIP Trunking can set up only as many voice and data channels as needed, avoiding the excess expenses.

Lower Capital Costs

As with many other IP-based Ethernet services, SIP Trunking is more cost-effective than traditional services. Unlike a standard switched PRI, there is no need for an IP gateway to be installed between a service provider's network and a subscriber's IP PBX. Once programmed by the network provider,

the SIP Trunking service can be delivered via a standard Ethernet connection directly into a business's IP PBX. Gateways to provide the protocol conversion for two traditional TDM PRIs can range from \$3,000 to \$5,000 – a cost that is avoided by having SIP Trunks provisioned directly into your IP PBX.

Reliability

Reliability is also an important factor. In contrast to early Internet VoIP services aimed at consumers, SIP Trunking services today are supported by privately managed IP networks rather than public Internet sites. As such, the provider can offer reliable support and service that is comparable in stability to a traditional PRI service.

Pricing

Arriving at an apples-to-apples comparison of pricing for SIP Trunking versus traditional TDM/PRI is critical in cost analysis. One consideration is that SIP Trunking generally does not include underlying costs for Ethernet ports needed to transport the SIP voice packets, whereas PRI purchases often include the cost of an expensive T-1 circuit. This is one reason why many quoted per-channel cost comparisons between the two technologies are lopsided, with SIP Trunking appearing to be far more cost-effective. Because an Ethernet connection is typically less expensive than a comparable switched T-1 connection, the combined SIP Trunking service and connection price often results in a lower service bill.

Pathway to Future IP Applications

SIP Trunking services bridge various existing Internet-based communications with an IP PBX and VoIP telephony, thereby interconnecting "islands" of Unified Communications (UC) functionality. Services like multilocation VoIP, instant messaging, World Wide Web access and videoconferencing are pooled together in a SIP trunk. Imagine the convenience of having a single phone number that will ring your work, mobile and then home phone in succession, giving you the flexibility to be productive almost anywhere.

Compared to legacy carrier lines and costly partitioned services, SIP trunks offer an unparalleled integration of high-speed communication services to boost your productivity and efficiency. They provide businesses with a next-generation alternative to switched services that is scalable and reaches beyond the office into the world of mobile devices, keeping your employees connected when they need to be, and how they need to be.

SIP Trunking has emerged as a solid choice for improved voice and data communications systems for many business customers.

Advanced Features

Following you will find five features that you may want to contemplate as you take a deeper dive into some of the benefits that you should expect to receive from your SIP service provider.

Bursting

In addition to providing standard trunk services, SIP Trunking can offer an option to “burst” the voice trunk group. “Bursting” enables a business to temporarily expand their local voice channel capacity and bandwidth in a SIP trunk group to accommodate increased communication traffic. A spike in traffic could be triggered by a special sales promotion, or a sudden increase in call volume due to an unforeseen event.

For example, an online retailer is provisioned with 30 SIP trunks plus the ability to burst to five additional channels. During the month, the retailer may hold a special two-day product launch which results in additional traffic to its customer call center.

If the retailer’s normal SIP trunk capacity of 30 is exceeded at any point during the two-day launch, the additional voice lines are ready on a real-time basis to handle the extra demand. An expanded volume of calls could be received or made to avoid losing potential sales.

Business Continuity

Call Forwarding Not Reachable: SIP Trunking can also be configured to provide an automatic call forwarding function in the event the office PBX is unreachable due to a network service outage or power failure. This “Call Forwarding Not Reachable” feature allows businesses to preprogram an alternate telephone number into their SIP platform, such as another branch office.

During a service interruption, whether the interruption is intermittent or lasting, affected calls will automatically forward to the customer-provided number. Once service is restored, calls will automatically reroute back to the primary location. This service function is monitored by the service provider’s platform and requires no intervention by the customer once the feature is activated.

Call Forwarding Always: This feature enables users to manually forward all of their calls from their PBX, even if it is still functioning. For example, if a business had to close its offices for a day due to a blizzard, the IT administrator could forward all of the calls to a secondary location via a customer web portal (MyAccount). Once the offices could reopen, the IT administrator would go back to the web portal and reset the call traffic to funnel back to the main PBX.

Advanced Features

(Continued)

Personal Mobility

SIP Trunking also can open the door to a range of Personal Mobility features, allowing PBX station users the ability to work remotely while still reaping the advantages of being at the office. Accessing a web portal set up by their provider, the designated users can have their incoming office calls routed to any alternate location. The system can also be programmed to ring all office, wireless or home numbers simultaneously, or ring each number in succession. In addition, calls from an employee’s cell phone or home phone will appear on the recipient’s caller ID as the employee’s office number.

Multi-Site Shared Trunking

This service enables enterprises with more than one location in a specific market to more efficiently allocate trunks among the locations. All voice traffic will be routed to specific locations based on existing numbering plans, but the SIP trunk group will be engineered to introduce economies of scale. This benefit will be created by provisioning the maximum number of trunks required to support the enterprise’s peak demand instead of provisioning a peak demand configuration for each location. This concept essentially spreads the voice resource requirements among all locations, likely resulting in the need for fewer total trunks. Of course, this service can be augmented with the Bursting function to assure that none of your customers ever gets a busy signal.

