SIP Trunking: A new voice in communications service

WHITE PAPER

With the rise of next-generation technology, business customers have more options than ever from providers offering communications services that better fit not only their day-to-day telephone needs but also their pocket books.

Nowhere is that more evident than in SIP Trunking, the newest option for high-capacity trunk services providing voice communications. With growing acceptance among corporate clients, ability to support traditional PBX (private branch exchange telephone system) voice features, simplified equipment requirements and compatibility with a wider range of business applications, SIP Trunking is rapidly becoming the standard among businesses of all sizes.

What SIP Trunking is

At its core, SIP Trunking uses an Internet Protocol (IP) alternative to traditional T1 or PRI trunk phone lines, removing the limitation of expensive hardware for every 24 channels or call paths. All new IP telephone systems have an Ethernet connection built in, so connecting over 80 SIP channels (more than 3 T1s) can be accomplished at no additional cost.

The technology behind SIP Trunking is Session Initiation Protocol, an IP signaling architecture that governs the communications exchange between two devices to set up and then terminate calls. Developed in 1996, SIP is now a ratified standard overseen by the Internet Engineering Task Force (IETF) and supported by an array of equipment vendors and network operators worldwide. In addition, the SIPconnect Forum was established to standardize how SIP is used for telephone services, and its Technical Recommendation 1.0 standard was approved in January of 2011. Several vendors have now been certified as compliant to that standard (link to the SIPconnect forum website: http://www.sipforum.com/sipconnect).

The SIP protocol was designed originally to handle individual IP communications sessions. Enter SIP Trunking, which extends the technology to govern multiple simultaneous communications sessions, much in the way older Primary Rate Interface (PRI) trunks manage bundles of switched voice circuits. Instead of using antiquated technology (Time Division Multiplexing or TDM), SIP Trunking sets up virtual IP voice channels that flow over an IP connection which can maximize network efficiency.

To set up service, a technician at the provider's network operations center remotely configures the customer's new SIP Trunk group, essentially providing a means for the service provider's network to communicate with the customer's IP PBX, just as a PRI trunk connects to a traditional PBX. The physical link between the customer and the service provider is an Ethernet cable in most cases, but can be fiber, copper or coax.

Once configured, the SIP Trunk supports all of the PBX call functions. It allows voice traffic to flow over a dedicated private network with quality of service or QOS (prioritizing SIP trunks over Internet traffic), such as Charter's fiber or coaxial cable (hybrid fiber/coax) infrastructure, or possibly an Internet connection which some providers use (no QOS). Interconnections between the service provider's network and the larger nationwide Public Switched Telephone Network (PSTN) allow call traffic between an employee and a caller using a traditional line or mobile phone to flow back and forth across the SIP Trunk.

The SIP Trunk can be closely configured to support the number of voice call paths the customer needs. For example, Charter Business customers can purchase call paths in blocks of four as opposed to blocks of 23 voice channels as required by older switched PRI trunk services.

Bandwidth also is needed to support SIP Trunking, and the amount needed is based on the total number of call paths. In general, each call path requires about 115 kilobits per second of bandwidth, so a business requiring 20 lines would need 2.3 Mbps of bandwidth available to support the resulting call traffic. Some service providers include the necessary bandwidth as part of their SIP Trunking service while others require their customers to purchase bandwidth independent of the SIP Trunking service.

A growing market

SIP Trunking comes at a time when telecommunications networks are making a major transition from older switched technology to newer, more flexible IP-based voice and data services. That, and the fact that SIP technology often can cost less than comparable PRI trunk services, is combining to create a boom market for SIP Trunking.

In 2010, the SIP Trunking market in North America increased 65.6% to total 7.2 million users, according to analyst firm Frost & Sullivan. The research firm estimates that the SIP Trunking market will continue to expand at an annual compound growth rate of about 35% and eventually connect some 59.1 million users by 2017.

Fellow research firm Infonetics Research, meanwhile, predicts a similar high growth rate, estimating that worldwide revenue from SIP Trunking will grow at a 52% compound annual growth rate through 2015.

Frost & Sullivan also found that about 85% of SIP Trunking sales in North America are among companies with 500 or fewer employees, an indication that it is by no means a technology confined to the Fortune 500 set.
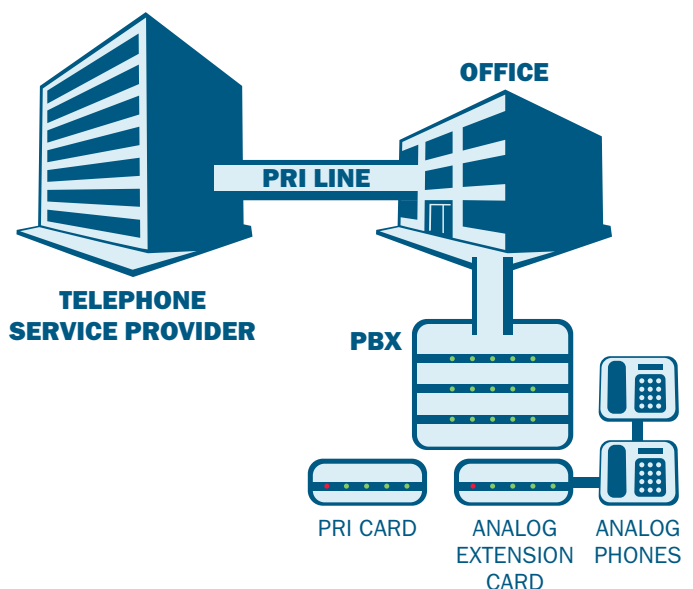
Advantages

So, what is attracting so many businesses to SIP Trunking for voice services? First and foremost, the SIP alternative offers better flexibility compared to older trunk line service.

Traditional PRI trunks lock customers into blocks of 23 switched Time Division Multiplexing (TDM) voice lines plus one signaling channel for network control purposes. But this static 24-channel setup can be problematic.
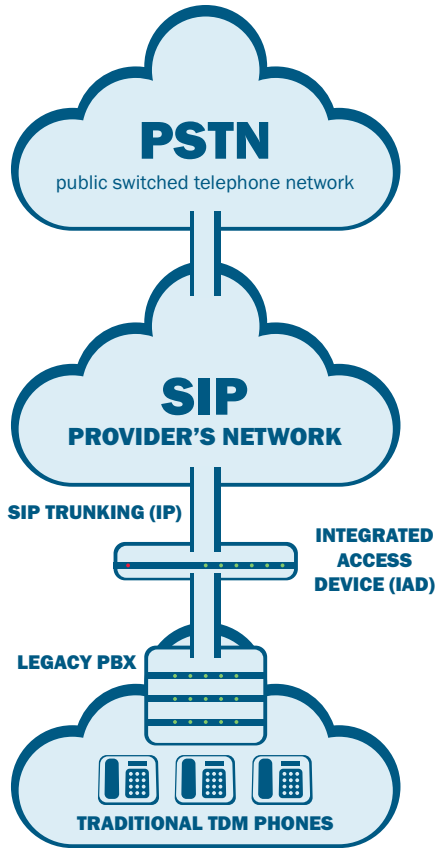
For example, if a business requires 28 voice lines, it would have to purchase two PRI PBX cards and bear the cost for the unused lines. The provider will also typically have to pay for a new line card for customer premise equipment, which increases the cost of providing the service.

If the company needs to expand its voice line capacity down the road, that also may require an added PRI trunk and possibly a PBX software upgrade, which can be very expensive.
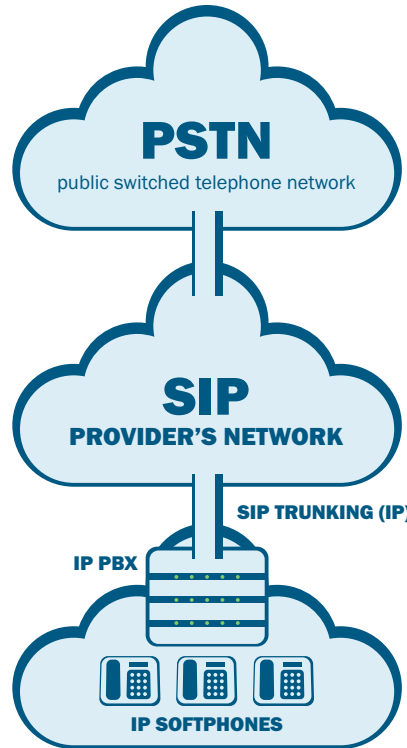
Traditional Switched Trunk Configuration

OFFICE

PRI LINE

TELEPHONE SERVICE PROVIDER

PBX

PRI CARD   ANALOG EXTENSION CARD   ANALOG PHONES

SIP Trunk —
Traditional PBX Configuration

SIP Trunk —
IP PBX Configuration

PSTN
public switched telephone network

SIP
PROVIDER'S NETWORK

SIP TRUNKING (IP)

INTEGRATED
ACCESS
DEVICE (IAD)

LEGACY PBX

TRADITIONAL TDM PHONES

PSTN
public switched telephone network

SIP
PROVIDER'S NETWORK

SIP TRUNKING (IP)

IP PBX

IP SOFTPHONES

Less equipment

In addition to more flexible service and features, SIP Trunking also can save customers money because it requires less PBX/telephone system hardware.

Traditional PRI will require one or more expensive cards in the PBX and the software "right to use" license for each. These elements usually must be installed by a technician on site.

In contrast, SIP Trunking is normally delivered via a standard Ethernet connection directly into the customer's IP PBX. All IP PBXs include an Ethernet connection port for no additional charge. A great benefit for the customer is that additional SIP Trunks (or Packs) can be added without the need for a technician, which saves the customer and the provider money.

SIP Trunking does have some hardware and software compatibility requirements. In some cases, customers will have upgrade their corporate firewalls to be "SIP-aware." Customers using traditional switched PBX units also will have to buy a media gateway or integrated access device to provide the IP-to-switched-TDM signal translation between the SIP Trunk and the PBX.

Also, because SIP Trunking standards did not exist before the SIPconnect Forum's Technical Recommendation, many IP PBX vendors used variants in the way they connected SIP Trunks. So, interoperability between a provider's phone system platform (central office) and the customer's IP PBX model and software release is required in order to validate that all services will work as expected. Most SIP Trunk providers provide a list of IP PBX models which have been verified.

In contrast, SIP Trunking service offers customers the ability to better tailor the service to fit their voice needs. In the case of Charter Business' SIP Trunking service, customers can purchase service in blocks of four call paths at a time in a "pack," providing a far better fit for the their actual usage requirements. At the same time, the required Ethernet connection the customer purchases also can be more closely tailored to fit the total number of call paths — without the need to purchase a second connection if the bandwidth requirement exceeds a PRI.

Adding call paths also is simpler for SIP Trunking services. To expand the line count, customers need only buy additional call paths and make sure their broadband connection will support the increase. Again, because SIP Trunking allows the customer to more closely fit both the call path count and connection bandwidth to their actual needs, the result can be a much more efficient service at a lower cost.

IP applications

Another advantage SIP Trunking offers is that because the platforms that provide the technology use more up-to-date software, they are compatible with a range of IP applications business customers are increasingly adopting.

As such, business customers can more readily layer in unified communication, network-based automated attendant capability, video conferencing and collaboration — all of which can be integrated directly with the SIP Trunking service. It also opens the door for integration with mobile communications, allowing users to forward voice calls between the office, mobile phone and home and share voicemail across multiple phones.

Charter Business' customized SIP Trunking

Charter Business' SIP Trunking service builds on these advantages, adding 24 x 7 service support and extended disaster recovery features to make it a reliable alternative to traditional trunk voice service.

A key element Charter Business brings to the technology is its secure, private network shielded from the public Internet. Charter Business owns the SIP Trunk as well as the connection, and that end-to-end network control offers greater security for voice traffic as well as better overall reliability than services that rely on leased access lines and Internet routing.

Also, because Charter Business Fiber customers will benefit from an Ethernet bandwidth connection separate from their Internet data service (included with the SIP Trunk service), they are assured that the voice bandwidth (using QOS) will be maintained regardless of the office's computer Internet or point to point data usage.

Charter Business Technical and service support also is a major benefit. New Charter Business SIP Trunking customers are assigned experienced sales engineers who can assess their voice and data needs and build a service that fits those requirements. They also work with the customer during the provisioning process as well as resolving migration issues from the customer's prior service to the new SIP Trunking platform.

Once it is up and running, the SIP Trunk network connection is monitored round the clock at Charter Business' network operations center. If service issues arise at any point in the chain from the customer's PBX to the network itself, these technicians can quickly diagnose and resolve them. That contrasts to other SIP Trunking services, which typically re-sell the standard telephone provider's connection service and often must rely on that third-party network provider, thereby raising the likelihood of finger pointing between the providers and longer delays to resolve service issues.

As an added feature, Charter Business offers customers the ability to divert calls in case of an office PBX failure, service failure or even a natural disaster simply by contacting the Charter Business Network Operations Center. Customers can redirect all telephone calls to a separate trunk group supporting another office location. They also can select specific direct inward dial (DID) numbers to automatically forward to any phone number.

Using the Automatic Overflow or DID Overflow features, all calls will automatically forward to the designated trunk group or phone number if service is interrupted or if all SIP trunks at one location are at capacity. If an outage is encountered, once the service is restored, calls will automatically switch back to the main SIP Trunk.

Conclusion

Given its flexibility and features, it is not surprising that SIP Trunking is on track to replace traditional switched trunking services for corporate voice and data systems over time. Its ability to provide a customized number of voice lines is attractive to many budget-minded businesses, and the fact it requires less equipment and can be more easily upgraded over time also is a plus. SIP Trunking also offers companies an easier migration path to advanced IP business applications to further drive productivity.

Charter Business' SIP Trunking service, meanwhile, adds not only secure network connections and round-the-clock monitoring but also automatic forwarding to another location or alternate numbers to assure calls are answered in the event of an emergency, network outage or a natural disaster.

For all of these reasons, SIP Trunking has evolved into a worthy communications service for today's business customers large and small.

SIP Trunking checklist: Things to consider

Customers looking into SIP Trunking as an alternative to traditional trunk line service have many factors to consider. Before deciding, customers should consider the following checklist:

How many voice lines are needed?

SIP Trunking allows for an unlimited combination of simultaneous call paths, but that doesn't necessarily have to equal the total number of extensions for a business. Not all employees are on the phone at the same time, and extensions for conference rooms, break rooms or empty offices may not justify being counted as a full line. Customer will often opt for a SIP Trunking call path total that is less than the total number of extensions. A customer's PBX vendor will typically determine how many SIP trunks are needed based on the quantity and type of individual using each telephone.

How much bandwidth is required?

The bandwidth connection itself is dependent on the number of simultaneous voice lines. In calculating this, customers should assume roughly 115 kilobits per second for each simultaneous voice line.

Is the company voice systems served by a traditional PBX or an IP PBX?

Until the SIPconnect Forum's Technical Recommendation is adopted by all vendors, interoperability confirmation between each IP PBX model and each type of provider's central office switch type is recommended. If customers have an older PBX/phone system, they may want to purchase a traditional PRI or they may consider buying a an Integrated Access Device (IAD) to translate the PRI into IP for relay through the SIP Trunk. The provider of the IAD and the PBX vendor will need to help install the SIP Trunk connection and all services must be tested.

Does the SIP Trunking provider also supply the Internet connection, or must that be purchased from a third party?

Some SIP Trunking services are offered by providers that don't actually own their own network infrastructure but rather lease access from the traditional telephone company or other network provider. Some SIP trunking services are even further removed from the underlying bandwidth and simply utilize whatever bandwidth is purchased by the customer via a completely independent broadband provider. For a customer considering such a provider, it will be important to ask if there are guaranteed service levels and response time to service outages related to this broadband connection.

Charter Business' SIP Trunking service is offered via its own private network, so this can offer an advantage in both service reliability and response time to service issues.