WHITE PAPER
IP Communications

SIP Trunking Deployment Steps and Best Practices

A practical guide for planning, evaluating, and deploying production service in your network

Introduction

Today's market conditions are forcing businesses to deliver maximum efficiency from their IP-based infrastructure. In our white paper, The Hidden Costs of Telephony Networks, we demonstrated how SIP trunking reduces the inefficiencies of time division multiplexing (TDM) technology by enabling an organization to pool voice network capacity for dynamic utilization and improving usage visibility for greater cost control and capacity management. Given these benefits, many organizations are making their own plans to leverage SIP Trunking as part of their transition to unified communications and collaboration.

Notwithstanding the inherent opportunities in a SIP migration, this advanced technology may present challenges to the IT organization, especially those who have limited experience managing voice technologies or who have never used SIP Trunking before. Like any IT project, careful planning and knowledge of implementation best practices will generate insights that can help save time and control costs related to a SIP Trunking migration. To assist, we've compiled some of the best practices we've observed throughout the hundreds of customer implementations we've been involved with over the past nine years.

Deploying a SIP solution should start with a general roadmap to prepare your organization for an effective implementation. To develop this initial roadmap, companies need to understand how SIP changes the network and make a careful inventory of existing communication assets and services. To help you get started, this white paper presents an easy-to-use, step-by-step guide to deploying SIP trunking, along with proven best practices to follow during the implementation. More detailed information is available in the book, "*SIP Trunking*" by Cisco Press.[1]

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Key Deployment Steps that Enable SIP Success

Although different deployment steps might be necessary to meet your individual business needs and may vary from one implementation to another, most SIP trunking deployments can be addressed by a few key steps. These include:

Step 1: Plan for Migrating to SIP Trunking

Proper planning helps ensure a smooth and predictable SIP implementation that meets or exceeds expectations. During the planning step, perform a detailed analysis including an equipment inventory at each site, noting how many trunks are installed. Once completed, take an objective comparison of current PSTN access expenditures with future expected costs. Beyond cost analysis, the planning step helps uncover the various impact of SIP to the network so that you can evaluate each issue and

1 Hattingh, Christina, Darryl Sladden and ATM Zakaria Swapan. "SIP Trunking," Indianapolis: Cisco Press, 2010.

address areas of concern. For example, you should consider traffic volumes and patterns, call policies, selecting pilot and production users, support for new services, and all security implications. Due to the IP connectivity of a SIP trunk, make sure your security group is involved in ensuring the connection is adequately secured. You may have to deploy new firewalls or reuse existing ones. Be sure to work through the physical design of the SIP trunk terminations with your security group long before you get to the installation of the trunk.

Step 2: Evaluate Available SIP Trunk Offerings
SIP trunk offerings vary considerably when compared to traditional PRI services. That is why you must carefully evaluate the exact capabilities and terms of each service provider's SIP trunk. Consider the provider's experience in deploying business grade VoIP solutions, network footprint, feature innovation and leadership, design flexibility, and business continuity flexibility. You must also determine how your network will interact and interconnect with your service provider's network. During the evaluation step, perform a side-by-side comparison of your existing PRI offering to the features available on the SIP trunk. While the features will not be equal—geographic coverage, physical delivery, and network demarcation—you must decide which of these are essential to your unique business needs.

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Step 3: Conduct a Pilot Trial
For the pilot trial, it's time to take what you've learned during the planning and evaluation steps, and apply the details to a real SIP trunk installation. Here, you will install the SIP trunk, configure the hardware and software, and begin running test calls across the network to see how it operates in your environment. One of the most important things to define before starting the trial is the success criteria and the duration. Otherwise, you might find yourself in an endless project, lacking the clear objectives that can determine success or failure. Equally important, be sure to create a clear test and user training plan, gain stakeholder buy-in, and then execute it. Once completed, you can then check off the pilot trial results, and make final service migration decisions based on accurate data points.

Step 4: Begin Production Service
After completing the planning, evaluation, and pilot steps to your organization's satisfaction, it's time to move the service into production. To launch this phase, you'll need to educate the wider user base that will join SIP trunk services using the same methods you used for the pilot user community. First, be sure to maintain TDM backup access for these users until you are satisfied the SIP trunk service achieves stability and QoS requirements. In addition, your service provider (SP) must port the Direct Inward Dialing (DIDs) of the new user community to join the SIP trunk, either as a single phase, or over several phases, depending on your initial project plan.

Essential Best Practices for Implementing SIP

While no two SIP trunking deployments are the same, there are common aspects that all IT departments will encounter. The following best practices will help you get the most value from your SIP deployment. Use the information in these best practices to gain clear and practical advice on how to approach your SIP trunk deployment.

1. Choose the Right SIP Provider
Who you partner with as your SIP provider will ultimately define the overall service level of your implementation. Be sure to scrutinize the provider's capability to meet the performance, reliability and scalability needs of your organization. To gain all the features that are important to you, always evaluate the SIP trunk features against your current TDM services.

- Does the provider own their VoIP infrastructure or are they reselling elements from another carrier?
- How expansive and robust is their network—how many gateways and session border controllers comprise their VoIP service?
- What level of experience does the provider have in delivering business grade VoIP service?
- What are the carrier's business continuity options?
- Can the carrier support your global locations/operations?
- What types of reporting and administrative tools will be available?
- Does the provider conduct rigorous testing on equipment?
- Do they have experience working with your preferred vendor and voice platform release?

2. Plan Your Deployment

The actual deployment can present a wide range of challenges if the project plan and requirements are not well defined. Like any large, complex project, plan carefully and don't rush into production services until you're certain of the results. Define the user communities or sites where you expect to deploy the different rollout phases. For all but the smallest of networks composed of no more than a single or handful of sites, SIP trunking requires a multi-phased rollout plan. Conduct an audit of all your existing lines and trunks to best understand what you have and how much you need. Audit your telephone numbers so you can determine which numbers need to be ported from other providers to the new SIP Trunking service provider. Determine the call flows included in each phase; for example, inbound, outbound, long distance, contact center, or general business calls. And finally, allow enough time for the actual installation.

3. Select a Robust Network Design

The SIP trunking design process takes more than just connecting a virtual topology per requirements. Carefully consider the benefits and challenges of centralized or distributed SIP trunk designs, even if the choice seems obvious. Centralized designs tend to look more attractive because of their cost savings opportunities. However, centralized designs often create network performance and connectivity implications that can increase the overall cost. In these situations, a distributed design may provide a better solution, especially if your environment already uses a distributed multi-protocol label switching (MPLS) network for inbound and outbound data.

4. Align Flexible Protocol and Codec Support with Business Goals

The protocols and codecs that you chose for a SIP implementation will define the type of services your network can support; so the decisions you make here must align with your business needs. Carefully consider the following issues for determining protocol and codec support:

- **End-to-end SIP.** For best performance, use SIP end-to-end throughout your network. Doing so creates a more flexible and streamlined interoperability solution. Otherwise, the network must translate data into other non-SIP protocols, and back again. If meeting this best practice is not possible, use a Session Border Controller (SBC) to interoperate H.323 with your SIP trunk.
- **DTMF.** Always use RFC-2833[2] dual-tone multi-frequency (DTMF) relay throughout your network. If this is not possible, make sure you understand the exact place where network translations occur between out-of-band signaling methods (such as traditional H.323 methods) and RFC-2833, which travels within the media stream. The device, such as a call agent on a SBC that performs this conversion must gain access to both the signaling and the media streams to run the conversion.
- **Payload Type.** Investigate RFC-2833[3] DTMF payload-type value assignments on your own call agents, endpoints, applications, and the values used by the service provider. In some cases, conversion or interworking between these values might require configuration on the border element to ensure proper call interoperation.
- **T.38.** Using T.38 fax relay for fax over IP transmission is a more robust and efficient fax transport method than other methods. However, T.38 is not available on all networks and all endpoints. In these cases, fax pass-through (or fax via G.711) remains in widespread use since it offers better interoperability over more endpoints. If the provider offers T.38, investigate if the provider offers failover to G.711 fax as a secondary call negotiation service. G.711 serves as backup allowing calls to connect to destinations that might not yet support T.38. Consider keeping fax on TDM trunks for a while longer if this is a critical part of your business.
- **Maintain Legacy Applications.** Similar to the previous best practice, keep modem, point-of-sale (POS), and telecommunications devices for the deaf (TDD) traffic on TDM trunks for the near term. SIP trunk technology is currently not ready to carry these traffic types in a reliable and predictable manner.
- **G.711.** Consider adding G.711 service on your SIP trunk. Although it uses more bandwidth than G.729, it doesn't compromise voice quality. The promise of a SIP trunk is to improve PSTN services, not degrade them. G.711 also eases fax issues, obviates the need for transcoding at the SBC, and better positions the network for new SIP trunk services. Doing so helps to prepare your SIP implementation for the future, enabling support for bandwidth intensive applications such as high-fidelity wideband codecs and video.

Like any large, complex project, plan carefully and don't rush into production services until you're certain of the results.

2 "RFC-2833: RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals," retrieved from www.ietf.org/rfc/rfc2833.txt on March 23, 2011.

3 Ibid.

- **Optimization Tools.** Although SIP is a standard protocol, it actually consists of a large number of individual standards such as those from the Internet Engineering Task Force (IETF), Request for Comments (RFC), along with optional components to these RFCs and alternative ways to implement the same call flows. SIP interoperability is not mature enough for all applications to predictably interwork with other SIP applications, even though they are standards-compliant. This means you might need tools to normalize and optimize SIP messages as they flow through network from the applications across the SIP trunk to the PSTN.

5. Protect Data with Lock Down Security

In today's threat-filled cyber world, no network is without vulnerability. In fact, SIP trunking networks face the same threats as the Internet. Providing security must be a top priority for any SIP deployment. Considerations for protecting your data include the following:

- **Security Demarcation.** If deploying a single-site or a small network, a firewall or NAT often provides sufficient protection as the enterprise demarcation and security border device. For larger networks, use a Layer 7 SBC device with additional protection, configuration, and traffic control.
- **Consider an MPLS-based Private Network.** MPLS services, such as Verizon Private IP, are network-based virtual private networks (VPN) enabling users to effectively communicate over a secure network.
- **Secondary Security.** Most large enterprise use both a firewall device and a SBC at their campus sites. In these situations, place the firewall on the outside as a first line security defense for all traffic, and install a Session Border Controller behind it as a second line of defense for unified communications traffic.
- **SIP Registration.** Use SIP registration on the SIP trunk if offered by the service provider. Also, use Digest Authentication on both SIP registrations and INVITEs, if available, as part of the overall service.
- **Fraud Protection.** Hackers target SIP deployments and SIP ports more frequently than other VoIP network architectures. To help secure all SIP ports, deploy tool fraud features on the Cisco Unified Border Element (CUBE) or SBC.
- **Port Changes.** Consider changing the SIP port from the standard 5060 port to provide additional protection against Internet sweeps for open SIP ports. To interoperate successfully, both your border element and the service provider SBC must make this change.
- **Use ACL.** Always place ACLs on the CUBE to confirm that only the service provider SBC can initiate calls to it from within the PSTN side. In addition, adding ACL protection limits call initiation from the internal network to your enterprise call agents (Cisco Unified Communications Manager or IP-PBXs).

6. Provide Redundancy and Reliability

No business can afford network downtime when high-demand data users drive the network to the limit. Achieving redundancy throughout a network helps protect your business from outages that could tarnish your brand and cost market share. To maintain a high level of uptime and reliability, consider the following:

- **Centralized vs. Distributed Trunking.** Be sure to study centralized and distributed trunking network designs carefully. An advantage of distributed trunking is the same inherent redundancy as provided by TDM PSTN gateways. If one site's SIP trunk is down, the SIP trunk of a different site can temporarily route the calls until service returns. If your business would be better served under a decentralized network design, confirm the carrier can still support concurrent call sharing across the enterprise.
- **Termination.** For SIP trunks under 500 sessions, redundant termination is generally not necessary, although you can always design in extra termination. For SIP trunks of 1000 or more sessions, redundancy is required.
- **Load Balancing.** Ensure your service provider can support load balancing.
- **Inbound Traffic Routing.** If the SIP trunk is down, outbound calls from the enterprise can easily route to other trunks based on traditional TDM PSTN gateways. However, inbound traffic to your DID numbers are not easily rerouted to TDM trunks. To provide redundancy and backup connections, discuss inbound traffic alternative routing possibilities with your SIP trunk service provider. Confirm whether the carrier can re-route inbound traffic to an IP termination only, or if they also have the flexibility to route to any telephone number. Also confirm if the carrier has the flexibility to re-route traffic automatically or manually on demand.

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Conclusion

Today's large and medium-sized enterprises can leverage SIP trunking to achieve cost control and improvements in business efficiency. Like any technology investment, gaining the best value from SIP trunking requires that businesses perform the due diligence in the deployment and implementation phases. For the deployment phase, determining the actual cost benefits of SIP will help you to fully understand how the technology can influence your enterprise's bottom line.

Evaluating SIP provider offerings and undergoing a pilot trial will help you compare and contrast the features that you need, while installing a small-scale network to verify success criteria. Once you complete these steps, you'll be ready for a full-service SIP environment. Just as important, be sure to drive best practices at each SIP implementation stage—and you'll be well on your way to realizing the most value from your SIP investment.

Verizon Business and Cisco deliver a SIP trunking solution that allows you to gain high efficiency from your telecom network. Converting a TDM network to IP-based SIP trunking sets the stage for new telecom services that you can use to further increase business efficiency and productivity.

About Verizon Business

Verizon Business, a unit of Verizon Communications (NYSE: VZ), is a global leader in communications and IT solutions. We combine professional expertise with one of the world's most connected IP networks to deliver award-winning communications, IT, information security and network solutions. We securely connect today's extended enterprises of widespread and mobile customers, partners, suppliers and employees—enabling them to increase productivity and efficiency and help preserve the environment. Many of the world's largest businesses and governments—including 99 percent of the Fortune 1000 and thousands of government agencies and educational institutions—rely on our professional and managed services and network technologies to accelerate their business. Find out more at www.verizonbusiness.com.

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