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Sonus Special Edition

SIP Trunking for Dummies®

Learn to:

- Understand the basics of SIP trunking
- Realize huge cost savings in your enterprise with SIP trunking
- Appreciate the important role of an SBC in SIP trunking
- Choose a SIP trunking solution that's future proof

Pat Hurley

SIP Trunking FOR DUMMIES®

SONUS SPECIAL EDITION

by Pat Hurley

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Contents at a Glance

Table of Contents

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Introduction

Most enterprises are familiar with VoIP (Voice over IP) and probably have begun at least thinking about implementing Unified Communications (UC) — a suite of integrated voice, video, data, and text communications delivered via the VoIP protocol known as Session Initiation Protocol (SIP). In fact, many enterprises have begun to deploy these services but are still connecting their primary voice communications systems called Private Branch Exchanges (PBX) by using legacy Time Division Multiplexing (TDM) T1 lines. This approach adds cost and complexity and makes your network less ready for taking the next step in UC.

A better way to connect PBXs is to move past those expensive T1s and on to an integrated Internet Protocol (IP) network using SIP trunking. *SIP Trunking For Dummies*, Sonus Special Edition, is here to help you understand this transition with some timely information about SIP trunking, the service providers and equipment that enable it, and the benefits you gain when you make your move to SIP trunking.

About This Book

SIP Trunking For Dummies, Sonus Special Edition, is *not* written as a technical manual for telecom engineers looking for deep techie knowledge or tips for planning SIP trunking deployment — there are plenty of 600-page technical books for those purposes. Instead, you'll find that this book is designed for the non-technical folks — marketers, sales professionals, finance wizards, and so on — in mid- and large-sized enterprises who want to save money and enable new UC services by adopting a SIP trunking solution. If you know a little bit about SIP trunking today, I think you'll know more when you've finished reading, and if you know nothing, don't worry; I walk you through all you need to know before you take that sales call or put out your Request for Proposal (RFP).

How This Book Is Organized

This book is organized into five chapters. As is the case in any *For Dummies* book, each of these chapters is self-contained, so you don't need to read the entire book from start to finish. If you see a chapter that you already know everything about, feel free to skip it. Whenever a complex topic from another chapter is raised, you see a reference to that chapter — no need to spend your time digging around in the table of contents.

Chapter 1: Getting Acquainted with SIP Trunking

Chapter 1 introduces you to SIP trunking and trunking in general. In this chapter, you discover enterprise PBX trunking and how SIP trunking fits in with traditional methods like TDM trunking over T1s. Then you learn how SIP trunking actually works and what components you need to implement it.

Chapter 2: Seeing the Benefits of SIP Trunking

Chapter 2 is where you find the (many) benefits of adopting SIP trunking, with a focus on operational rather than monetary benefits. SIP trunking can simplify the management of your network and enable the deployment of new UC services. SIP trunking can also help you adopt cloud-based services.

Chapter 3: Reaping the ROI Benefits of SIP Trunking

SIP trunking is a huge money saver for many enterprises — saving many as much as 50 percent on their recurring telecom costs. In this chapter, you discover what these Return on Investment (ROI) benefits are all about. You'll also see some real-world case study examples from three enterprises that successfully implemented SIP trunking. If you're in finance or have a budget to worry about, you'll enjoy this chapter.

Chapter 4: Taking a Peek at SIP Trunking Considerations

In Chapter 4, you find out the questions to ask and the things to consider before you commit to a SIP trunking deployment and when you choose an Internet Telephony Service Provider (ITSP) to provide your SIP trunking service. You also see the important considerations to keep in mind when selecting a Session Border Controller (SBC).

Chapter 5: Ten Reasons to Choose Sonus SBCs for SIP Trunking

If you read Chapters 1 through 4, I hope you're sold on SIP trunking and also have a good idea about what makes a good SIP trunking solution. But even if you haven't, Chapter 5 is where you look at some of the attributes of the SBCs offered by Sonus and understand how Sonus SBCs may match up well with your SIP trunking needs.

Icons Used in This Book

This book calls out important bits of information with icons on the left margins of the page. You'll find three such icons in the book.

The Tip icon points out bits of information that aid in your understanding of SIP trunking or provides a trick of the trade to help save you time, money, and much more.

The Remember icon points out textual tidbits that you should lock away in your memory for future use.

I try to keep the technical jargon and discussion to a minimum. You won't need to know these factoids to get the most out of the book, but you may find them interesting.

Chapter 1

Getting Acquainted with SIP Trunking

In This Chapter

- Saying hello to PBX trunking
- Digging deeper into enterprise PBX trunks
- Discovering how SIP trunking works
- Learning the components of SIP trunking

Enterprises of all kinds are dealing with two big trends when it comes to their real-time communications needs. First, they're trying to save money in an era when just about every business is tightening its belt. Second, they're trying to figure out how to support a host of new unified communications applications in an era where new devices, mobile connections, and more sophisticated users are demanding change *right now*.

In this chapter, you learn about a technology being considered by many enterprises that meets both of those goals: Session Initiation Protocol (SIP) Trunking. First, I discuss enterprise Private Branch Exchange (PBX) Trunking in general and then the difference between circuit-switched and SIP trunking. I follow that by then digging deeper into SIP trunking and looking at the components required. By the end of this chapter, you should have a pretty good idea what SIP trunking is all about.

Introducing Enterprise PBX Trunking

Many enterprises — in fact, I'd say most medium and large enterprises — use a PBX telephone system to provide voice and other communications services to their employees. You know what I'm talking about if you dial "9" first to get an outside line or if you can dial a coworker with just a few of the digits of her phone number.

The telecommunications network connections that connect the PBX to a telephone service provider are known as *trunks*. The name implies a tree analogy that explains what a trunk is: all of the individual lines (phones on peoples' desks) are the branches of the tree, connected to a bigger line (the trunk), which connects back to the network. Trunking is a big deal because it lets an enterprise pay for fewer phone "lines" than it has users with phones — the PBX lets a larger number of users and stations effectively share a smaller number of connections to the outside world — it's sort of the time share of the phone world, but you get to use it more than two weeks a year.

The thing to remember about enterprise voice trunking is that it's an efficient and cost-effective alternative to buying an individual phone line for every person in the enterprise. Imagine both the cost and the management hassle of that approach.

Understanding the Types of Enterprise PBX Trunking

For many decades, PBX trunks were traditional *analog* circuit-switched voice lines just like the ones you may buy from your local phone company for your home. For example, a small business may purchase three lines from the phone company, connect them to a small business phone system (essentially a mini-PBX, sometimes called a *key system*) and then share those lines among the six or seven people (extensions) in the office.

Going Digital with PRI

Another option for connecting a PBX to a telephone carrier is known as a Primary Rate Interface (PRI) line. PRI is a Time Division Multiplexing (TDM) interface, provided over standard telecommunications copper or fiber lines. PRI uses a T1 interface in North America or an E1 interface in other parts of the world.

T1 and E1 lines are carved into 23 or 30 individual voice lines, respectively. Essentially, a T1 or E1 is a souped-up phone line that can carry multiple voice lines over a single pair (or pair of pairs, in some cases) of copper phone lines, by using some digital trickery to divide the voice signals up by time slots. All the extensions in the office connect to the PBX, which then connects to one or more PRIs and on to the phone company network.

Entering the SIP world

The PRI system worked well for a long time, but a more efficient (in terms of both bandwidth utilization and cost) and modern system has begun to replace circuit switched voice: *SIP trunks,* which utilize Voice over IP (VoIP) and the SIP.

SIP trunking uses a data connection — the same connection that a business uses for Internet access, for connecting to cloud services, or for hosting its e-commerce site — to carry voice signals as VoIP to a service provider who can handle that kind of voice signal.

A carrier who can offer VoIP services is called an Internet Telephony Service Provider (ITSP). An ITSP could do just that, or it could be your familiar local and long distance carrier who offers VoIP services in addition to more traditional circuit-switched voice services.

Understanding How SIP Trunking Works

SIP trunking relies on Internet protocols and Internet services instead of old-fashioned circuit-switched voice protocols

and services. With SIP trunking, voice communications can be merged with the data services an enterprise uses. In fact, many smaller enterprises merge their voice and data onto a single "Internet" data connection while larger enterprises may keep them separate but save significant amounts of money by using cheaper data services for voice instead of TDM trunking. In other words, voice (and related communications services such as video, which I discuss in Chapter 2, when I talk about Unified Communications, or UC) is just another data service being delivered over an enterprise's data network.

A good way to understand this concept is to think about the phone functionality on your Android, Windows, or iPhone smartphone. It's just another one of the apps on the phone, instead of a dedicated hardware device. When you want to check Facebook comments, you click the Facebook app, and when you want to make a call, you click the Phone app (or the Skype app, for that matter).

The key protocol that supports VoIP connections is SIP and is basically the controller of calls (and other UC sessions). Here's a way of understanding SIP: Think of your voice as a data connection, and your language (like English) is the protocol that lets other people understand your voice. SIP isn't the only language (there are others such as H.323), but it's the most common. SIP relies on clients and servers in the network to exchange information about the following:

- Who's connected to the network?
- Where are they located?
- What resources are available to them (in other words, what applications can they access and use, and what kind of phone or device are they using)?
- Who's inviting people to begin a session, if they accept, and when are they done?

A SIP trunk is a virtual connection between an Internet Protocol Private Branch Exchange (IP PBX) and a telephone service provider providing SIP-based voice and UC services, connected over an enterprise's data network connections.

Introducing the Components of SIP Trunking

Some of the components needed to implement SIP trunking are hardware devices that you need installed on your network, while another component is the service itself, provided by a third-party service provider. I cover those components in this section, but Figure 1-1 shows some of the necessary components in a typical deployment.

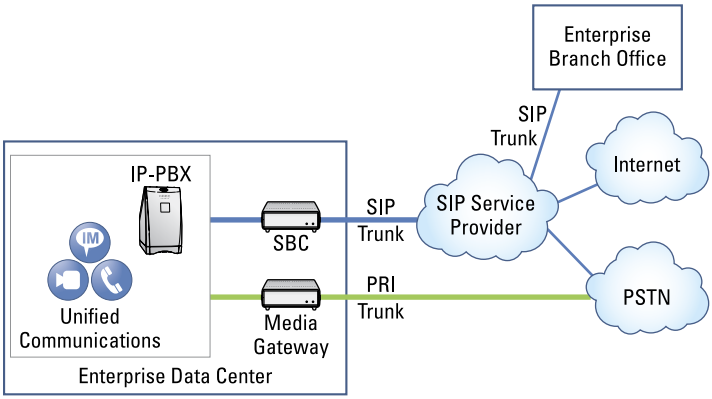

Figure 1-1: SIP trunking components.

IP PBX

The key element to SIP trunking is a phone system that can convert voice calls into VoIP calls for transmission across the SIP trunk. While there are other devices out there that can do this job — various VoIP gateway type devices, for example — the most common and cost-effective mechanism is an IP PBX.

Session border controllers

A key element of a SIP trunking solution — and I do mean *key* — is the Session Border Controller (SBC). The SBC is the

device that sits on the border between an enterprise's private network and the public network provided by data and telephony service providers. The SBC plays a few very vital roles in managing SIP traffic for voice and other UC services and applications, including the following:

- **Security:** SBCs protect the network against threats by
  - Protecting against Denial of Service and Distributed Denial of Service "flood" attacks, which attempt to overload network components and prevent voice services
  - Preventing unwanted ingress into the network (SBCs "hide" the topology and network addresses of devices "inside" the enterprise network) — you don't want everyone coming in and making calls from your IP PBX
  - Providing encryption for sessions traveling across the network, especially if you care about people eavesdropping on you
  - Protecting against toll fraud from malicious outside users accessing the network
- **Routing:** The SBC provides inter-network connectivity to allow remote workers to make phone calls and also allow older IP devices to be used on your network.
- **Quality of Service:** The SBC keeps an eye on overall network utilization and on the policy-based priorities for different applications and users. It also determines if and when sessions should occur and what priority and amount of bandwidth they should be assigned — all on the fly.
- **SIP translation:** Different vendors have slightly different dialects of SIP — an SBC can "speak" these different dialects and allow uninterrupted connectivity between different network segments.
- **Transcoding and transrating:** The SBC can actually modify the "payload" or content of SIP sessions to change codecs (coder and decoders) and change bit rates to make sure both ends of a SIP session can communicate, regardless of what kind of client or network is in use.

Session Management System

An optional but often very useful addition to a SIP trunking implementation is a *Session Management System.* Without session management, many or even most UC applications will have their own servers, their own management systems, their own policy enforcement, and their own policy database systems — each of these elements individually controlled and managed. Session management brings all these elements under a single system's control, saving time and money.

Session management has an additional benefit: It provides intermediation and federation (in the computing sense, not the political one) between different platforms, different networks, and different geographies. So a session management system makes it easy for a UC application to be deployed one time and then provided to users (based on policies) no matter where they are, what device they are using, what kind of IP PBX is installed in their office, and so on. This ability to centrally and quickly deploy new applications is a big deal for enterprises with heterogeneous network infrastructures — like companies who've merged or with subsidiaries who "did their own thing" when they built their networks.

Session management isn't a "must have" for SIP trunking, but it's a problem solver for complex UC deployments. Session management is something you definitely should consider as you move from a simple voice-centric UC deployment and start adding new multimedia apps.

ITSP/SIP service provider

Your SIP sessions — whether they're voice calls, video conferences, multimedia sessions, or whatever (the sky's the limit) — need a public network provider to get to where they're going if they're not internal calls handled on your own enterprise network. That's the role of a service provider — just as it was with traditional circuit-switched voice.

A service provider that provides transport and termination of SIP calls is the ITSP — sometimes also called a SIP service provider. You can really think of an ITSP as just a "phone

company," with the difference being their interface with your network (through your SBC and onto your IP PBXs) is a data connection using SIP to control the flow and routing of sessions.

In fact, many traditional phone companies are beginning to offer SIP trunking services to their enterprise customers — after all, if someone is going to cannibalize your legacy circuit-switched voice services, it may as well be your own company, right?

In its doorman role, the SBC first and foremost determines which sessions should even be allowed on and off the private portion of the network. So "spam" calls or calls from blacklisted users (like telemarketers) can just be stopped at the door and not allowed in, while other sessions can always be connected (whitelisted) or connected when certain conditions are met (greylisted). Additionally, the doorman part of the SBC also determines what UC applications a session can access, based on policy, availability, and the capabilities of the clients on either end of the connection.

Seeing the role of an SBC in SIP trunking

The SBC is essentially the traffic cop and doorman of SIP trunking. In its first role (traffic cop), the SBC looks at each individual session crossing between the internal enterprise network and the external ITSP network and determines where and when that session should go — in other words, where the session should be routed and what priority the session will be assigned when the network is busy. The SBC also determines how many lanes (how much bandwidth) should be assigned to a session, based on network utilization and the policies established for the network.

Finally — and perhaps most importantly — the SBC performs SIP interworking, which allows devices that use subtly different variants of SIP to communicate with each other effectively and efficiently.

Chapter 2

Seeing the Benefits of SIP Trunking

In This Chapter

- Combining voice and data
- Deploying new SIP services faster
- Embracing the cloud
- Having a more reliable network
- Skipping local and 800 calls

Session Initiation Protocol (SIP) Trunking isn't just something you should consider implementing because it's the next big thing. Nor is it something that's just for tech-focused firms looking to be on the bleeding edge. Instead, SIP trunking provides some concrete and measurable benefits for most enterprises, by simplifying network elements, enabling new services, and reducing expenses.

In this chapter, you discover more about the benefits of SIP trunking — how SIP trunking makes it easy to integrate your data and voice networks, improve the services you offer, and save money. Check out Chapter 3 for more info about the cost savings and business case for SIP trunking.

Unifying Your Access Network

SIP trunking lets you combine your voice and data network onto a single, unified data connection. Let us repeat this one, this time with emphasis: SIP trunking lets you combine your voice and data on a single network connection. Got it? Good!

Not every enterprise chooses to combine its voice and data networks onto a single connection. Many smaller enterprises will, but larger enterprises may choose to use *separate* data connections for their voice. The overall effect, however, is still pretty much the same: using (far less expensive) data bandwidth instead of Time Division Multiplexing Primary Rate Interface (TDM PRI) facilities for handling voice. So even if you choose to provision separate data facilities for your SIP trunking network, you're still gaining all the other advantages of SIP trunking, and you're still saving a significant amount of money.

Today, most enterprises have separate (and expensive) network facilities to connect their Private Branch Exchange (PBX) to the network of their phone company for transporting inbound and outbound calls that are outside of the company network. And here's the thing about this situation: Most enterprises have Internet Protocol Private Branch Exchanges (IP PBX) that use SIP and Voice over IP (VoIP) for the portion of calls between the PBX and the actual phone, as well as using SIP and VoIP for intra-company calls (for example, calls between corporate offices, campuses, and branches). See Chapter 1 for more on SIP if you're not familiar with it.

But for calls that are coming from or going to third parties, these SIP sessions are using devices called *Public Switched Telephone Network* (PSTN) *Gateways* and are sent on old-fashioned circuit-switched phone lines.

SIP trunking lets you get rid of the expensive PRI and Basic Rate Interface (BRI) connections and PSTN Gateways used for these calls (see Chapter 1 if you need a refresher on PRI). Dumping these PRIs and moving to SIP trunking keeps your calls as SIP sessions (instead of converting them to TDM) and delivers them to a SIP service provider (or ITSP — Internet Telephony Service Provider). With SIP trunking you can even use the data connection you already have in place for Internet and related services (like your intranet, access to cloud-based services, e-mail, e-commerce, and so on) for your voice calls.

Why would you want to get rid of your BRIs and PRIs? Well, primarily, because they're expensive, but also because they mean more things to manage, more contracts to review and sign, and more "stuff" to deal with for your overworked IT staff.

Making management easier

When you eliminate PRIs and all their related equipment and standardize on data networks for voice and Unified Communications (UC), as you do with SIP trunking, you make the network easier to administer and manage. You have less equipment and network access connections to manage. But there's more to it than that.

SIP trunking, with the addition of Session Border Controllers (SBC), allows you to centralize the management of your entire network — all your locations and IP PBXs. When you centralize this control, you can do the following:

- Create centralized dialing plans that apply to every location in your enterprise (instead of creating custom dial plan for each individual PBX)
- Use a single carrier (your ITSP/SIP service provider) rather than contracting with multiple local and long distance carriers in each location
- Set your security and application policies one time — in your centralized policy management database — and apply them to users whenever needed
- Centralize your billing and cost accounting — so you know exactly what you're spending where and can charge or rebill your costs accurately and easily

Simplifying growth

SIP trunking makes it a *lot* easier to add capacity to your VoIP and UC network when needed. In the old BRI/PRI days, if you needed more "lines," you called the phone company and had them installed and integrated. With SIP trunking — because your voice and UC sessions are carried over your integrated IP data connection — you can simply allocate more or less bandwidth as needed and you're done. That's not only easier, but also it's cheaper!

REMEMBER

With PRI connections, you have to add lines in increments of 8, 12, or even 20+. So if you just need to add five more lines worth of capacity to your IP PBX, you're stuck adding (and paying for) more.

Additionally, the centralized management of your unified UC network makes it easier to quickly deploy new UC applications, without worrying about location and the different types of IP PBXs and other devices you have at each location. With SIP trunking and SBCs, you can deploy your new UC applications once and access them anywhere on your network.

Deploying SIP Services More Easily

Adopting SIP trunking and SBCs — especially when combined with a session management solution — facilitates the adoption of new SIP-based UC services and applications. That's because one of the key attributes of the SBC (and of session management systems, as well) is the ability to make issues of location, vendor interoperability, and legacy equipment obsolescence pretty much go away.

Don't "rip and replace"

The SBC plays a vital role in SIP trunking. It's the traffic cop and the doorman for a SIP-based UC network. But it also plays a role in saving some serious money as enterprises move to an all-IP and all-UC future by providing the intermediation and federation between disparate IP PBX systems and related access and network equipment.

Many enterprises have relatively non-homogeneous network infrastructures — Brand X IP PBXs in this location and Brand Y in another — simply because there wasn't a perceived need to standardize or maybe because the Brand Y infrastructure came along with a merger or acquisition — or even because the upgrades were simply too expensive. As these enterprises begin moving toward an IP and UC network, they may find that some vendors want them to rip everything out and start from scratch. There's no need for this!

The right SBC and/or session management solutions — ones that are broadly interoperable with legacy equipment — make it easier and cheaper to move to SIP trunking and unified communications, and let your old equipment keep up with current standards without an upgrade.

The SBC (and the session manager, when used) is designed to intermediate between different and disparate network locations and equipment choices — translating protocols, codecs, and the like when needed. The result is a network that no longer looks (from an application server perspective) like ten different and disconnected networks but instead as one big network (substitute your enterprise's number of locations for ten to personalize this).

Applications can be deployed one time, in one place, and made accessible to everyone — even to remote users like teleworkers or your sales team on the road in Paris pursuing that big deal. You no longer need to deploy a SIP application in your San Francisco office for their Avaya PBX and then do it all over again in Chicago for their Cisco PBX. You do it once and let your SBCs, your session management solution, and your SIP trunks connect them all together, seamlessly.

What kind of services can be enabled through SIP trunking and unified communications? Well, the sky is pretty much the limit, but here are a few:

- Audio and video conferencing
- Audio recording
- Presence (determining the availability and communications preferences of a user on the network)
- Instant messaging (IM)
- Online collaboration (whiteboarding)

Enabling Cloud-Based Services

Beyond enabling SIP services that are deployed on your own servers within your own network, SIP trunking makes it easier to connect to services that are hosted in the cloud. Because your voice and UC communications are entirely IP and SIP-based — all the way from the desk or mobile phone and on into the network of the ITSP, there's no longer a need to have applications delivered from a server connected directly to your IP PBX.

Instead, that server could be in a different one of your offices, or it could be located somewhere in the datacenter of a cloud

service provider. Cloud services typically cost less (in terms of capital equipment because those costs are borne by the cloud service provider) and are quicker to deploy. They're quicker because they're effectively already deployed — all you're doing is signing a contract and setting the policies that let your users access them.

Making Your Network More Reliable and Resilient

Because it isn't tied to a fixed set of PRI or BRI connections to a traditional phone service provider, a SIP trunking deployment is more resilient and reliable than a traditional PBX trunking solution. A SIP trunking solution puts much of the heavy lifting of routing and completing calls and sessions into hardened, redundant ITSP data centers off your site — instead of distributing it to a number of local phone exchanges and on local circuit-switched BRI/PRI connections.

The result is that when something bad happens — a power outage, a natural disaster, a cut line, whatever — it's extremely easy to reroute sessions to other locations with just the proverbial flip of a (software) switch. And SIP trunking also makes it easy for those businesses who require this level of redundancy to work with more than one SIP trunking ITSP so in the very worst case (and unlikely) scenario where the entire ITSP goes down, it's easy to automatically route traffic through the other ITSP.

This SIP trunking resilience isn't just useful for those rare occasions when disaster strikes. It also makes the distribution of the flow of sessions easier when the network gets congested (load balancing) or to redirect sessions on whatever ad hoc basis you can think of. For example, when the Los Angeles office is all out for a retirement lunch, you can automatically transfer their calls down to San Diego. You can even create time-of-day routing that moves your incoming calls to various offices (moving ever-westward) as the work day ends in other locations.

You benefit from the cost savings element here, too. By using policies and your centralized dial plan, you can implement least cost routing — routing sessions using the most inexpensive method at the time that session is created.

Going Local and Eliminating 800

Because SIP trunking is based on the Internet, it essentially takes location out of the equation. So with the help of your SIP trunking ITSP, you can establish Points of Presence (POPs) in local areas where you do business without having to have a physical office presence there.

When a customer or partner in that local area wants to call you, all she needs to do is dial a local number and her call is routed via the ITSP to one of your locations. No need for long distance on her end, no 800 numbers to deal with, no added grand expense on your end (you may find it's cheaper to add these local POPs than to deal with 800 toll free charges).

As a side benefit, your business, no matter how large and national or international it may be, "looks" more local for your customers when you have a local number.

Chapter 3

Reaping the ROI Benefits of SIP Trunking

In This Chapter

- Comparing TDM and SIP trunking costs
- Finding other cost savings
- Looking at case studies

If you ask enterprises that have switched to Session Initiation Protocol (SIP) Trunking for their Voice over IP (VoIP) and Unified Communications (UC) needs why they did so, you'll get one overwhelmingly predominant answer: to save money. And that makes a lot of sense in an economic and business climate where saving money is at the top of every CEO, CIO, and IT manager's to-do list.

In this chapter, you find out more about how SIP trunking saves enterprises money and (more importantly for most folks) how much it saves. Then you see some real-world case study examples of enterprises who've implemented SIP trunking — just so you don't think this is all theoretical savings.

There are non-monetary reasons to choose SIP trunking as well, and you can read all about them in Chapter 2.

Making the Case for Migrating to SIP Trunking

Traditional circuit-switched Time Division Multiplexing (TDM) trunking, delivered to your Internet Protocol Private Branch

Exchange (IP PBX) over Primary Rate Interface (PRI) lines is just plain expensive. SIP saves enterprises significant amounts of money — on a recurring basis, every month — by moving this traffic to the Internet Protocol (IP) data network and off the old TDM infrastructure. This is a pretty big deal for many enterprises — savings of 50 percent or more are common in areas such as long distance charges and monthly access line fees.

Aggregation of trunks

In the TDM world, enterprises pay for two separate aspects of the voice lines that connect their PBX to the telephone service provider. These two things include

- The access facility (like a PRI/T1) that physically connects the enterprise facility and PBX to the telephone company local exchange
- The trunks themselves — the individual "phone lines" carried over the PRI or other access line

  In the U.S., a T1 PRI carries 23 of these voice trunks

So an enterprise pays a monthly fee for both the access line(s) and for the trunks carried over the access line(s). Each category of trunks has different pricing. The categories are as follows:

- **Local trunks:** These trunks terminate local calls (in the telco world these are called intra-LATA calls). Local trunks usually cost about $35 per month, with unlimited calls (for example, no additional charge per call).
- **Long distance trunks:** These trunks are used for calls leaving the local area and heading out on a phone company's long distance network. These trunks are usually free of monthly recurring fees, but they incur per-minute charges for calls (varying, but typically in the range of .5 to 2 cents per minute, depending on the byzantine phone company pricing plans or tariffs).
- **Toll free trunks:** These trunks carry your enterprise's inbound 800 number traffic. Pricing here is similar to long distance (see the preceding bullet), with no recurring base fee but per-minute charges of about 2 cents.

Additional fees may also apply for things like transferring of calls, directory assistance, and directory listings.

In a SIP trunking deployment, the cost of the trunks themselves is reduced due to several factors:

- Fewer trunks are required than in a traditional TDM trunking scenario — often 35 to 50 percent fewer — because trunks are centralized into a few data centers and can be shared among multiple offices, due to the following:
    - Time zone considerations (different busy hours)
    - Better statistical usage of the trunks — when the number of trunks and users grows, the required ratio of trunks to users decreases
    - Different trunk categories can be shared — no more requirement to dedicate trunks to local, long distance, or toll-free
- SIP trunks are usually just cheaper in terms of monthly recurring and per-minute charges. Additionally, features that you pay more for with TDM trunking (like call transfers) are often cheaper or just included in the base fee for a SIP trunking service.

  Don't limit yourself to a single SIP service from a single Internet Telephony Service Provider (ITSP). You can contract with more than one and use least cost routing to send calls to the appropriate ITSP on a per-call basis.
- Lower taxes and tariffs can be expected, because SIP trunking providers aren't required to follow all the rules that traditional phone companies must follow.
- SIP trunks can be added one-at-a-time instead of in increments of 23 as is the case with TDM trunking. No more rounding up to the next multiple of 23!
- SIP trunks are more flexible in terms of bandwidth, so additional trunks can be added by just decreasing bandwidth during really busy times — like the Christmas rush in a retailer's call center.

Many voice conferencing providers use 800 number access for users to "dial into" a conference call. Each of the dial-ins uses a local voice trunk. The enterprise may need to pay for many more local trunks than otherwise required for day-to-day

calling purposes. SIP trunking can obviate this requirement by providing all-IP access to the conferencing service, thereby avoiding the cost of these local trunks entirely. You still have to pay for SIP trunks (instead of PRI trunks), but what you save is the difference in cost between the two.

Aggregation of access

A large, multi-national enterprise may lease thousands of PRI lines at $450–$600 each per month. That's a lot of money. SIP trunking moves PBX trunking off these expensive lines and to (far) less expensive Ethernet and other data services (like DSL for very small remote branches and offices). This expense goes on a $ per Megabit/second basis.

For almost all enterprises, this data connectivity already exists and is being paid for and in many cases (especially for smaller enterprises) there's enough extra capacity to implement SIP trunking without adding capacity to the data network.

When an enterprise shares a single data connection for voice and video, bandwidth that's not actively being used for a call or other SIP session is available for whatever other data services it's already being used for. Also, when you use a dedicated voice SIP trunk, it frees up capacity on the data bandwidth. SIP trunks only use the bandwidth when they're using the bandwidth.

Even if a bandwidth upgrade is warranted, the cost of upgrading a business Ethernet access line is typically far less than the cost of adding the equivalent in PRI. Enterprises can save up to 80 percent, every month, on their access costs with SIP trunking.

On-Net calling

With SIP trunking, all your calls and other UC enter and leave your enterprise premises as SIP sessions on your data network connection. This means that calls that are heading between internal locations can stay on your network and stay off your trunks — in other words, they're *on-net* calls. While you're paying for the bandwidth, you aren't paying any per-minute charges or monthly trunk charges for them, so they're almost — dare I say it? — free.

Having the ability to carry significant portions of your calls and UC sessions on your own network is also a great bargaining chip when negotiating with an ITSP about your SIP trunking services. You may find that they're willing to cut a deal to get more of your traffic.

You may be able to connect to third-party services like audio or videoconferencing directly via your IP connection and also bypass using your trunks for those sessions as well.

Other cost savings

If you've been reading this entire section, you've already discovered the biggest savings in SIP trunking. But a few other areas come into play and add to the overall savings:

- **Management expenses are reduced.** By centralizing the management of your voice/UC network, you can greatly reduce the expense of managing that network. No more having dedicated engineers working on a per-site basis.
- **Changes are made with software only.** Adding or moving trunks is simply a software instruction, instead of a technician physically swapping out wires.
- **Remote workers are brought "on-net."** SIP trunking lets remote workers and folks who are on business travel use softphone (software phone) clients on their laptops or mobile devices to connect to your internal network. Instead of dialing through a (relatively) expensive mobile plan or outfitting their home offices with expensive business lines, remote workers can make and receive calls from your IP PBX and through your ITSP or internal network. This process can provide some serious cost savings and also allow workers to benefit from all the UC features that workers sitting at a desk in your office benefit from.
- **Billing is less complex and therefore less costly.** For traditional phone companies, billing systems are incredibly complex and costly — and the cost gets passed along. In fact, as much as 40 percent of your telco bill may be for the bill itself. SIP trunking greatly simplifies the rates charged and allows for simpler, cheaper billing.

Looking at Case Studies

Don't just take my word for it that SIP trunking saves money and meets real business goals for enterprises. Take a look at the real-world examples in this section. You discover three companies who've migrated — partially or in full — from TDM to SIP trunking for their voice and UC applications.

Names are withheld, but these case studies are actual companies who've adopted SIP trunking. If you figure out who they are, don't tell anyone!

Case Study #1: The global company

The company:

- Global company with dozens of international locations
- 15 to 1000 people per office

The need:

- A strategic mandate to reduce telecom costs — the average monthly international long distance billing was greater than $1.5 million per month!

The solution:

- Implement a centralized VoIP routing solution, including SIP trunking
- Install SBC and VoIP gateway devices
- Install a centralized policy server to manage call routing and service selection

The result:

- Immediate 55 to 70 percent reduction in recurring monthly telecommunications charges due to the ability of the centralized VoIP system to keep intra-corporate calls "on-net" (as described in the section titled "On-Net calling" earlier in this chapter)

- Recouped initial investment in just a handful of months and the savings rolled in after that
- No decrease in quality of service
- Easier (and less expensive) telecom management due to the centralized nature of the deployment

Case Study #2: The bank

The company:

- Global Fortune 500 financial services company
- Business divisions in both the banking sector and outside of the sector
- Maintains several call centers with thousands of agents involved in sales, financial transaction processing, and general customer support
- Geographically dispersed
- Growth — both organic and through mergers and acquisitions — lead to diverse set infrastructure vendors to support

The need:

- Corporate mandate to move to IP-based call centers
- Desire to minimize equipment costs and to utilize legacy equipment, instead of rip and replace
- Regulation required encryption for SIP sessions
- Desire to support existing and new services while simultaneously saving space (real estate) in its data centers

The solution:

- Replace TDM gateways with IP gateways and SBCs with SIP trunking
- Keep its existing IP PBX infrastructure from multiple vendors in multiple locations
- Enable hardware-based encryption on the SBC

The result:

- A 60 percent savings in data center real estate compared to the previous TDM solution
- The SBC supported hardware-based encryption allowing the company to meet all regulatory requirements with no "hit" in performance
- The SBC provided the transcoding required allowing all the existing PBXs, regardless of vendor, to "talk" to each other and interoperate — reducing a significant expense from the previously used telco transcoding services
- A new infrastructure supporting both TDM and SIP trunking, allowing the bank to migrate fully to SIP trunking at its own pace

Case Study #3: The airline

The company:

- Global 2000 listed airline with over 50,000 employees, distributed worldwide
- Large globally distributed call center supported by a mixture of PBXs

The need:

- Desire to reduce costs while increasing customer satisfaction
- Mandated reduction in operating and provisioning expenses by shifting to a unified IP network, while improving productivity
- Desire to allow remote call center employees to access VoIP via the Internet
- Requirement to maintain security and redundancy
- Requirement to interoperate between SIP, TDM, and H.323 (an older VoIP standard) during a gradual migration to an all-SIP network
- Retain as much legacy PBX and related equipment as possible

The solution:

- VoIP gateways to support legacy TDM PBXs
- Centralized policy server
- SBCs supporting SIP trunking

The result:

- Significant cost savings based on two primary factors:
    - International intra-corporate calls were routed on-net rather than through long-distance carriers.
    - Off-net traffic utilized the optimized call routing enabled by the policy server and SBC combination, allowing least-cost routing.
- Reduced operating expenses, while maintaining network reliability and increasing security and performance through SBC-based call admissions control
- Improved productivity and customer satisfaction by the quality of service and reliability enabled by the SBC solution

Chapter 4

Taking a Peek at SIP Trunking Considerations

In This Chapter

- Looking into ITSPs
- Determining pricing and service strategies
- Evaluating SBCs

Chapters 1 through 3 cover the reasons why you should consider implementing Session Initiation Protocol (SIP) Trunking in your enterprise voice and Unified Communications (UC) network. If you skipped right here and you're not sure if SIP trunking is for you, I recommend you take a gander at those chapters.

In this chapter, you take a look at the considerations for beginning the process of migrating your network to SIP trunking.

Considering Your ITSP

A big part of any SIP trunking deployment is, of course, the company that actually provides you the service — your Internet Telephony Service Provider (ITSP). Dozens — maybe even hundreds — of companies offer SIP trunking services these days, and the number keeps growing. You may be offered SIP trunking by the following vendors:

- Your traditional Time Division Multiplexing (TDM) telephone service provider (the company you're buying your Primary Rate Interface [PRI] TDM trunks from today)

- The data service provider who offers you your Multiprotocol Label Switching (MPLS) Wide Area Networking (WAN) services
- A "pure play" SIP trunking provider, focused solely on SIP trunking

You may end up picking more than one provider. For example, you may choose one provider for your primary service and another as a backup, or one for domestic calls and another for international. You don't need to limit yourself to just one.

Pricing and pricing structure

Because most enterprises choose SIP trunking because of its cost savings benefits, figuring out the price you're going to end up paying is pretty important. Figuring out your actual pricing isn't just a matter of someone writing a number down on a piece of paper and handing it across the table to you.

Look at your usage patterns and then consider the following:

- **On-net traffic:** How does the ITSP handle your on-net traffic — the traffic that goes from one of your locations to another. Preferably you won't pay any per-minute charges for this traffic. Many distributed enterprises save a lot of money on just this factor alone.
- **Pricing structure:** For the off-net traffic you *do* have to pay for, how do you pay for it? You should pay a fixed per-minute rate for all domestic long distance calls, based on concurrent calls. Local calls probably shouldn't be charged per-minute — instead those calls should be rolled into your concurrent call fees.

  You may also find pricing structures with flat rate or bundled minutes (like a mobile phone plan). Look closely at your past usage and do some math to see if this is, indeed, a good deal.
- **International calls:** These typically are charged at different rates on a per-country basis, so look closely at your usage to see which countries you call most frequently.

- **The price itself:** Of course, after you look at how the pricing structure matches up with your usage, you can look at the actual prices charged and create your business case to estimate your SIP trunking savings.

Dealing with DID and local calls

Direct Inward Dialing (DID) is simply the series of numbers that customers and partners use to dial into your network and Private Branch Exchange (PBX). Keep in mind two things regarding DID when you evaluate an ITSP:

- **Number portability:** Simply put, can you keep your existing numbers? If you use your existing telephone provider for SIP trunking the answer is most assuredly yes. For other providers this can be an open question: so ask it!
- **Local number availability:** Beyond keeping your existing numbers, you may wish to establish local numbers in other locations, in lieu of toll-free 800 number services (particularly if you're in a consumer-facing business). Not all ITSPs are able to provide this in all the places you want to establish a local presence — so ask this ahead of time.

Dispersing aggregation points

Aggregation points are the spots where multiple SIP trunks from your offices come together and into the SIP service ITSPs network. They're a potential weak spot in a SIP trunking deployment not because there's anything inherently unreliable about them, but simply because they can be a single point of failure when the unexpected (say . . . a meteorite strike) happens.

You can deal with this potential in two ways:

- Ensure that the aggregation points are properly hardened. Your ITSP should be able to tell you about its redundancy and its disaster recovery plans and be able to give you a good story about how it would deal with failures.

- If your network is big enough or mission critical enough, have multiple, geographically separated SIP trunking aggregation points and configure your network to gracefully failover when the worst case scenario plays itself out. This could be through contracting with two different ITSPs or through one ITSP who can offer you this option.

TIP

Having geographically dispersed aggregation points can also be a boon to providing multiple local DID numbers, as discussed in the preceding section.

Quality of service

Your ITSP should be able to assure you that the quality of service you get from your SIP trunking service meets or exceeds the quality of service of your existing TDM trunking. After all, you *are* trying to save some money, but you're probably not trying to have missed calls, poor voice quality, and general all-around poor performance as you do so.

I give you a few things to consider in this section.

Blocking

Voice and UC networks are built around peak traffic considerations — a reasonable calculation of how many simultaneous sessions need to be supported at your busiest hour of the day. They aren't built around an assumption that everyone in your organization suddenly tries to use every UC application that they can all at the same time. It's possible that in some extreme case (maybe you sell iPhones and 5 million people are all calling in at midnight to order theirs), you'll have more sessions that your network can handle — this is called *blocking*. Your ITSP should specify in its contract with you what an acceptable amount of blocking will be and be contractually obligated to maintain this.

Codecs

Codecs (coders and decoders) are the software algorithms that digitize and compress voice and video signals for transmission across an IP network. There are multiple codecs (all standardized under International Telecommunication Union (ITU) standards that begin with "G." like G.711). The choice of codec impacts two things:

- The voice quality of the call
- The amount of bandwidth used

Ask your ITSP which codecs it supports and uses. You may, for example, be willing to trade a little bandwidth for a higher quality "wideband" codec like G.722.2, which offers more CD-quality voice. Or you may be told that a lower quality codec like G.729 is a good compromise but then find out your voice quality sounds like a cellphone circa 1983. Talk to your potential ITSPs about this upfront and test out the codecs they support.

Agreements

Look closely at the Service Level Agreement (SLA) for your SIP trunking service. Make sure there are penalties for SLA metrics that aren't met. Your SLA will cover things like

- **Uptime:** The amount of time your network is guaranteed to be working (like, 99.999%).
- **Latency and jitter:** These are measures of the timing of the signal across the network — *latency* measures the delay, while *jitter* measures the variance in that delay. Too much of either makes for lousy sounding calls and junky video.

If your ITSP also provides other services to you — like your MPLS data service — don't just assume that the SLA for those other services covers your SIP trunking service. Look for a specific SLA for SIP as well.

Considering Your SBC

In this section, I move you beyond the realm of the service provider and into the equipment you need for SIP trunking — specifically the Session Border Controller (SBC). Check out Chapter 1 for more on what an SBC does.

You also need a PBX (or PBXs), preferably Internet Protocol Private Branch Exchanges (IP PBX). For the purposes of this discussion, I assume that you already have PBXs in place, so I won't discuss them here.

The SBC plays a number of vital roles in SIP trunking, primarily by being the gatekeeper between your internal network and the external WAN that carries sessions to and from your ITSP. The SBC, as the name implies, controls the border between these networks, and determines which sessions are let in or out. The SBC also provides security for the network, deals with the routing of SIP signals, and does a whole lot of intermediation to make sure things just plain work right — like dealing with transcoding and transrating and providing SIP normalization.

For an even more in-depth discussion of SBCs, check out *Session Border Controllers For Dummies,* Sonus Special Edition.

Check out a number of important SBC buying criteria:

- **Security features:** In maintaining the security of a SIP network, SBCs play several important roles:
  - They maintain access control, determining which sessions are allowed to cross the network border, according to policy.
  - They can maintain lists of callers to determine certain sessions that are always or sometimes (conditionally) allowed.
  - They protect against Denial of Service (Dos) and Distributed Denial of Service (DDos) attacks, where malicious users attempt to flood the system with SIP requests until it's overwhelmed and no longer responsive.
- **SIP normalization/interoperability:** SIP is a standardized protocol, but each vendor tends to have slight variations in how it implements it. Make sure your SBC has a robust *SIP normalization* capability — make sure that it's been tested to work with a variety of vendors and that it can do on-the-fly translations between different SIP "dialects."
- **Transcoding/transrating support:** Not every session connecting through your SIP trunking UC network will be using the same client and on the same high-bandwidth network. For example, you may be handling sessions from fax machines that require older codecs and your network may now be using newer more bandwidth-efficient codecs. It's the job of the SBC to do real-time code conversion (*transcoding*) and bit rate adjustments

(*transrating*) to make sure that both ends of the session can actually communicate with each other effectively.

- **NAT Traversal:** Most internal/private networks use a system called Network Address Translation (NAT) to assign clients within network private IP addresses that can be reached through the one (or handful) of public IP addresses for that network. Users connecting a session from outside the network only see the public IP address. The SBC has the job of figuring out where to route that session internally without disclosing the private IP addresses.

  This same feature allows remote workers (teleworkers and road warriors alike) to use a SIP softphone client on their laptops or mobile devices to get "dialtones" from your internal network. This capability lets them make and receive calls as if they were physically located on your network, and the users benefit from the cost savings and feature enhancements.

- **Performance/scalability:** An SBC is a computationally-busy device with a lot of routing and other functionality (like SIP normalization, transcoding/rating, and media support) to perform in real-time. A key factor to think about *before* you pick an SBC is that it's highly likely that the load you put on your SBC today and even the load you *think* you'll put on it in the future may be underestimates. As mobile proliferates and new UC apps flourish, there may be a lot more devices using a lot more applications than you can possibly predict — and this is going to happen very soon. Make sure your SBC vendor can grow with you without breaking the bank. Look at:

  - **The peak performance of the SBC:** How many sessions can it handle and how does mixing in things like transcoding and transrating affect performance?
  - **How the SBC is upgraded:** When you do need more than your current SBC can provide, is it upgradeable via processor or line card additions? Or do you need to buy a whole new chassis?

- **Manageability:** Before you buy *any* piece of network equipment you should ask yourself this question: How will I manage this device, and how does that management integrate into how I manage everything else? An SBC is no different in this regard. Look for SBCs that use

> standardized network management systems and don't require you to station an expensive engineer at each location to keep them up and running.

If you're going to need to manage SBCs in multiple locations — likely for an enterprise moving to SIP trunking — look for SBCs that can be centrally managed, so policies and configurations can be set once and propagated across the network to all your SBCs automatically.

Chapter 5

Ten Reasons to Choose Sonus SBCs for SIP Trunking

In This Chapter

- Leveraging leadership
- Maximizing performance

When it comes time to choose your SIP trunking hardware solutions, you may have many choices. In this chapter, I show you ten reasons why to choose Sonus over the other guys.

A Leader in SIP Communications

How many minutes of Session Initiation Protocol (SIP) traffic is a lot to you? A million? A billion? How about 50 billion minutes a month? That's how much SIP traffic is currently deployed by Sonus. The number is growing every month, too. If you're looking for a vendor with experience in SIP communications — and you should be — keep that number in mind.

Networked Policy Management for SBCs

Many Session Border Controller (SBC) solutions require hands-on configuration and hands-on policy management. If

you're looking at a big deployment, with multiple SBCs, that's both a pain in your wallet and a pain somewhere else entirely. Sonus SBCs can be centrally managed using Sonus' networked policy management. Set a policy one time, in one place, and it automatically flows through to each and every SBC in your network. That's the way to do it!

Highest Performance

Sonus SBCs are built to handle the load — with plenty of overhead to spare. They've been proven in the field and they've been tested in the labs under conditions that simulate a massive network attack without faltering. Third-party lab evaluations have confirmed it — Sonus SBCs performance is excellent.

Sonus divides its SBC labor into three categories:

- General computing for policy management and call control
- Network processing for packet routing, security, and interworking between IPv4 and IPv6
- Media processing for media transcoding and transrating (see Chapter 4 for more on this)

This three-dimensional approach means that a heavy load in one area (like an unusually large number of sessions requiring transcoding) won't impact the other elements of the SBC.

Better Transcoding Support

Sonus SBCs have dedicated processors for transcoding and transrating (and other media processing tasks) — processors optimized for this task and not bogged down with other day-to-day SBC functions.

Additionally, Sonus doesn't rely on off-the-shelf firmware for Digital Signal Processors (DSP) but instead writes its own, optimized firmware. Not only does this make transcoding work better and faster today, but also it means that you have someone to turn to when a new codec comes along that needs support. Other vendors are stuck waiting for the chip vendor to upgrade their own firmware; Sonus can just write its own!

Scalability

SBC performance doesn't end the day you buy an SBC and install it — it also includes what you need to do when you finally max out the capacity of your installed SBC base. That day may not happen for a long time with the performance offered by Sonus SBCs, but eventually it will.

With a Sonus SBC, you can add capacity in any of the three processing functional areas (general computing, network processing, and media processing) individually. So you can "grow" an SBC based on your usage patterns and the areas that are causing you issues.

Support for Future Apps through Media Support

There are all sorts of telecommunications megatrends that are moving communications beyond just voice and into richer media types — like videoconferencing, high-definition audio, and rich-media collaboration. Many first generation SBCs can't handle this — they end up requiring additional boxes to pick up the slack. Sonus SBCs are ready *today* for rich media and are easily upgradeable as new types of media become prominent.

Ironclad Security

The network firewall and intrusion prevention system that you have in place for your data network are probably pretty darn effective. They also are designed to prevent data-centric attacks. VoIP and SIP have their own, unique attack vectors that those data devices aren't really designed to identify and stop.

Sonus SBCs are designed to do the following:

- Provide encryption (end-to-end) to protect the content of your SIP sessions from prying eyes
- Hide the "insides" of your private network from the outside with *topology hiding* — so people making SIP session connections to your network can't "see" past the front door to tell what's happening inside

- Implement blacklists to keep known bad actors on the outside
- Protect against DoS/DDoS attacks by repelling the flood of requests without losing the ability to complete legitimate sessions

Guaranteed Interoperability

Different vendors have slight variances in how they implement SIP in their products. This can result in poor performance/call quality or even a straight-up inability to complete some calls. Sonus has tested its SIP normalization engine and made sure it works with all major vendors on the market — so you don't need to worry about SIP interoperability.

Robust Session Management

As you expand your SIP service portfolio beyond just a few simple VoIP apps (like conferencing or call recording), you may find that you're beginning to spend way too much time managing systems, customizing applications to work on different platforms, and more. When that happens, look into session management (more info in Chapter 1).

Sonus has a robust and industry-leading session management solution that's already deployed in some leading and marquee customers' networks, and which is ready to be part of your SIP network as you grow it.

TIP

If you want to dig into SIP session management in more detail, grab yourself a copy of *Session Management For Dummies,* Sonus Special Edition.

A Complete Package

Whatever you need to outfit your network for SIP trunking — be it SBCs, a session management solution, or VoIP gateways — Sonus has it. Sonus lets you build a SIP trunking solution on top of what you've already got or helps you design, build, and implement your SIP trunking plan from the ground up.

Consider a SIP trunking solution for your company

Move beyond expensive T1s to an IP network by using SIP trunking. This book helps you understand that transition and the service providers and equipment that enable it. You also discover the benefits of SIP trunking when you decide to make that move.

- ***Enable new UC services* — adopt a SIP trunking solution**
- ***Investigate your ROI* — SIP trunking is a huge money saver for many enterprises**
- ***Commit to a SIP trunking deployment* — discover the questions to ask and the things to consider before doing so**
- ***Make the move to SIP trunking* — leave costly T1 connections in the past**

Open the book and find:

- **The building blocks of a SIP trunking solution**
- **The benefits of deploying SIP trunking**
- **The importance of SBCs in a SIP trunking solution**
- **How to choose the right SIP trunking solution**

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