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Peter Gregory
with Tom Doria, Chris Stegh, Jim Su
Foreword by Alan B. Johnston

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Contents at a Glance

Foreword

These days, in communications circles, Session Initiation Protocol, or SIP for short, is seemingly everywhere.

SIP is supported by practically every manufacturer of IP Phone, Gateway, Call Manager, and IP PBX. It is part of the IP Multimedia Subsystem (IMS). It is powering the fastest growing VoIP residential and enterprise providers. It is a part of PC operating systems and has been enthusiastically adopted by the open source movement.

Years ago, someone proposed a usage of SIP that was dubbed "SIP for Light Bulbs"! Don't laugh, it may happen yet.

So what is SIP and why is the industry buzzing about it? This book will tell you. What can you use SIP for? This book will tell you that too. Why is SIP so important? You'll find that here, too.

Before I leave you in the competent hands of the authors, I will add a few of my own answers here. SIP can be called a "rendezvous" protocol. That is, it allows endpoints on the Internet to discover, locate, negotiate, and establish sessions. What kind of sessions? Any kind of sessions. SIP is used to establish VoIP (of course), video, gaming, text, call control, and others I'm sure I've left out. Recent extensions to SIP add in instant messaging and presence capability. What is presence? This book will tell you, but presence stands ready to revolutionize enterprise communications the same way public Instant Messenger networks have revolutionized consumer communications.

Besides all these applications and uses, SIP is also generating its own ecosystem. In the same way that the Internet opened up networking by displacing closed, proprietary networking protocols, SIP has opened up communications and displaced closed and proprietary signaling protocols. It has created an entire ecosystem of interoperable and configurable devices and services that is revolutionizing the way communications is done.

Not bad for a little protocol developed in academia back in the mid-1990s by such thinkers as Henning Schulzrinne and Jonathan Rosenberg.

The authors have done an excellent job of explaining the what, why, and how of SIP in an understandable way. Enjoy your read of *SIP Communications For Dummies, Avaya Custom Edition*!

Alan B. Johnston

February 2006

Introduction

Imagine a communications environment where a central directory server not only knows how to reach an individual's work phone, cell phone, and pager, but also her instant messaging (IM) program, e-mail, and PDA. Not only that, but also imagine that the central directory server also knows a party's communication preferences and capabilities, and can intelligently alert a called party when someone is trying to reach her. Finally, imagine that phone calls to an unavailable person can be intelligently rerouted to another person or group depending upon a number of interrelated factors such as time of day, whether the called person is scheduled to be in a meeting, or whether one or more of her modes of communication is unreachable.

These capabilities aren't some dream of a far-off utopian future, but are available today thanks to a remarkable advance in communications: *Session Initiation Protocol (SIP)*. SIP is the glue — and the intelligence — that makes these advanced communications capabilities possible.

Vendors are rushing to incorporate SIP into their products, including those that work with Voice over Internet Protocol (VoIP). Here's a short list of the kinds of products you can expect to become SIP-enabled:

- VoIP phones, gateways, proxies, and servers
- VoIP *softphones* — phone software programs that run on PCs, PDAs, and other devices
- VoIP PBXs
- Instant messaging (IM) programs
- Videoconferencing systems

SIP is an open standard, with an active working group on the Internet Engineering Task Force (IETF) that has given SIP tremendous legitimacy and momentum. Avaya and other major companies are active in the IETF SIP working group, as well as in other industry groups working to make sure that SIP works across enterprises that have a variety of architectures, standards, and products in use.

About This Book

This book describes SIP from both business and technical perspectives. You can read about SIP architecture and operations, as well as its impact on business. You can discover how SIP can improve internal and external communications as well as the basics of how SIP technology works, and how to build a SIP environment.

Foolish Assumptions

We assume that you have a keen interest in ensuring that your company's networking and telecommunications systems are up to the challenges of intelligent communications today and tomorrow. Regardless of your role in your organization, this book can help you quickly get up to speed on how SIP promises to revolutionize electronic communications.

How This Book Is Organized

Each part of this book considers a different aspect of SIP environments. You may want to read the book cover to cover to gain a fuller understanding of SIP, or you may prefer to skip around to find out what you need when you need it.

Part I: The Case for SIP gives you the high-level view of what SIP is and what it can do for your company's communications. If you're unfamiliar with SIP's benefits, this is a great place to begin.

Part II: SIP at a Glance explains how being based in current standards makes SIP compatible with existing systems. It also introduces presence, the SIP feature that adds intelligence to communications at many levels, and it describes the components in a SIP network that make intelligent communications possible.

Part III: How SIP Transforms User Communications describes how SIP "addressing" works with the concept of presence to make reaching users easier, regardless of which device they're using or where they are.

Part IV: How SIP Transforms Enterprise Communications explains how SIP drives down the cost and complexity of intra-enterprise and inter-enterprise communications by permitting their consolidation with IP data communications. This part delves into the details of basic SIP calls, including proxy-mediated voice calls, presence-enabled calls, instant messaging, and videoconferencing. Here we explain the concept of trunking — that is, connecting enterprises' IP-based communications systems together over long distances. We also discuss ENUM, the protocol that bridges the new SIP-based URI system with the old TDM phone number system.

Part V: SIP Interoperability discusses the principles needed to support an enterprise migration to SIP. No two enterprises are alike, so it would be difficult to come up with a single works-for-all recipe for migrating to SIP. Instead, this part explains what vendors are doing to make multi-vendor integration with SIP as straightforward as possible.

Part VI: SIP and Server-Free Communications discusses how small businesses and distributed offices can take advantage of SIP features, even without dedicated servers.

Part VII: SIP and the Future of Intelligent Communications looks into the future of SIP and how it will continue to evolve and improve, embracing more communication technologies and supporting more enhanced communications capabilities.

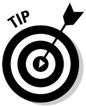
Part VIII: Top Ten Reasons for SIP-Enhanced Communications offers a condensed list of the most important reasons to put your communications on steroids. SIP will make your communications more intelligent than anything else available today. This intelligence pays off in more effective communications that result in happier customers and more productive employees. If you're still sitting on the fence about SIP, start here.

Icons Used in This Book

Throughout this book, I occasionally use icons to call attention to material worth noting in a special way. Here is a list of the icons along with a description of each:

Some points bear repeating, and others bear remembering. When you see this icon, take special note of what you're about to read.

This icon indicates technical information that is probably most interesting to IT professionals.



If you see a tip icon, perk up — you're about to find out how to save some aggravation.

Where to Go from Here

No matter where you are in your SIP project, always keep your eye on the big picture. Avaya has keen vision and leadership in the communications industry. You can learn a lot from this book, but a lot more from Avaya professionals. Turn the page and discover for yourself why Avaya is one of the leaders in converged voice and data environments powered by SIP.

Part 1

The Case for SIP

In This Part

- Defining SIP
- Connecting people anywhere, anytime, on any device
- Facilitating interoperability
- Streamlining communications with or without servers
- Preparing for the future of SIP

Do you increasingly feel like your communication devices are holding you hostage? Just as you finish checking your multiple voice mailboxes you get an instant message from someone that you just left a message for, saying, "I'm off the phone now, can you call me back?" Or have you left a greeting on your work voice mail saying, "If you've missed me, you can call me on my cell phone, or better yet, send me an e-mail if this is after hours." You may be rapidly reaching the conclusion that everyone has too many devices, too many numbers, and too little time. Sometimes modern communications technology seems to have forgotten the main reason for using it: to communicate with another person more conveniently.

People have more options available to them today for communicating with each other than ever, yet they often have a harder time getting through. The selection of choices spans a dizzying array of technologies and devices that can deliver voice, e-mail, instant messages, and even video. Simply picking up the phone still works, but now users have more choices for deciding how to reach somebody — and, ironically, that is the problem.

With people more widely available and more connected than ever before, an unintended and unexpected communications paradox has emerged. Users need to manage multiple identities for each of the devices and networks that they want to use. Determining the best way to reach a person and managing a contact list for these multiple identities can be staggering. Simply put, communications today has become device-centric, not user-centric.

This part outlines how SIP is one innovative technology that actually helps improve communications without complicating these issues.

What Is SIP?

Session Initiation Protocol (*SIP,* pronounced just like *sip,* as in sipping from a fire hose on a hot day) is an open signaling protocol for establishing any kind of real-time communication session. The communication session can involve voice, video, or instant messaging, *and* can take place on one of many devices that people use for communicating: laptop computer, PDA, cell phone, IM client, IP phone, and so on. SIP has been developed in the *Internet Engineering Task Force (IETF)* by common participation from various vendors, including Avaya.

SIP builds on a number of existing communications protocols. Developers and system administrators can easily work with SIP, customize it, and program with it. It is rapidly becoming a standard for service integration (how new services and applications are created and combined) within a variety of wireless and carrier networks, and is gaining momentum within enterprises. This growing acceptance in both enterprise and service provider networks offers the promise of a single unifying protocol that will transform not only communications within an enterprise, but communications between the enterprise and its ecosystem of partners, suppliers, and customers. Is SIP a refreshing solution to simplifying communications? For companies who need to sort out and reconnect the current tangle of disparate communications protocols and programs: Yes!

To understand the power behind this protocol, you need to examine some of the key factors that are driving SIP's momentum across all aspects of enterprise communications.

A brief history of SIP

SIP traces its origins to the mid 1990s in the Internet's experimental multicast backbone, or "Mbone." This network was used to facilitate the distribution of streaming multimedia content including seminars, broadcasts of space shuttle launches, and IETF meetings. The original draft of the SIP specification was published in the IETF in 1996, and eventually standardized in 1999. Today, the most up-to-date core SIP specification can be found in RFC (Request for Comment) 3261.

Presenting Presence Services

SIP introduces a new model for communications through its support of *presence*. Presence enables you to locate a user and determine his willingness and ability to participate in a session, even before you initiate communications. This information, reflected across multiple devices such as IP phones, cell phones, and instant messaging clients, makes communication simple and efficient by helping you to reach the right person at the right time, on the right device.

Presence and preference features enabled by SIP are discussed in more detail in Part 3.

Celebrating User-Centric Communications

Communications today are *device*-centric. Every device has its own phone number, address, or alias. The more devices you use, the more addresses others need to remember in order to reach you. And without presence, as described in the previous section, communication becomes a guessing game when trying to connect with people, wherever they may be and whatever they're doing.

Do we need fold-out business cards?

You probably don't consider yourself a communications geek, but like most modern business people, you probably have many ways to communicate with others. For example, a typical salesperson may have:

- Three phone numbers (home, work, cell)
- Text messaging and e-mail on a PDA
- IM identities on Yahoo!, Google, MSN, and AOL
- IM capabilities on a laptop and on a cell phone
- Four e-mail addresses

All of these identities operate in silos — none is aware of any other. The salesperson's communications capabilities have not become easier, but more difficult, because of the lack of integration between all of these media types. SIP promises to bring all of these capabilities (and more) back together.

With SIP, communications become *user*-centric once again. A SIP *address of record (AOR)* provides one unifying identifier that can be mapped across multiple devices and media types. You can think of an AOR as the user's "public address." Part 2 explains more about AORs.

Simply put, SIP-based communications are between *people,* connected together without needing to know which device they happen to be using. No more tracking of multiple phone numbers, e-mail addresses, and IM contact names.

TIP

SIP is particularly suited to facilitating communications with mobile devices such as laptop computers, cell phones, and PDAs. Part 3 describes how SIP enables mobile communications.

Nevertheless, SIP still supports the legacy *public switched telephone network (PSTN)* with its numeric dialing because it's going to be around for quite a long time. An effort to map PSTN telephone numbers with SIP's newer user-centric identifiers is discussed in Part 4.

Encouraging Interoperability

SIP uses a *text-based language*. That doesn't mean that SIP supports only text; it means that SIP's messages are easy to program and interpret, making it easier to achieve interoperability between different vendor implementations. SIP is also very modular and *extensible,* allowing for the integration of existing legacy protocols. These properties make SIP an ideal protocol for implementing a standards-based converged communications network.

The SIP standard is defined in RFC 3261 by the Internet Engineering Task Force (IETF). The IETF is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and its operation. Several neutral consortiums, including SIPit, SIP Foundry, and SIP Forum, arrange meetings and events where companies with SIP-based hardware and software products can test interoperability with other SIP-based products. This testing helps to promote smoother integration of SIP-based products in carrier and enterprise networks. We tackle this subject in depth in Part 5.

Some vendors have gone beyond these efforts with active SIP interoperability and *ecosystem* programs. In a well-run SIP ecosystem, devices and services obviously need to work together seamlessly, and the only way to ensure that they do is for many kinds of vendors to test their SIP-based products together. For example, Avaya has made a commitment to establishing openness and interoperability for SIP through its Developer Connection program, which support software developers, systems integrators, and service providers in testing interoperability and developing SIP-based solutions with Avaya products and services. You can find more information about interoperability efforts in Part 5.

The impact of SIP goes beyond internal communications within an enterprise. SIP has become a signaling standard for carrier networks. Service providers have started to provide SIP-based trunk services that can reduce costs and extend an enterprise's SIP environment into the public network. The adoption of SIP for external connectivity will lead to a transformation in communications between an enterprise and its ecosystem of partners, suppliers, and customers. SIP may eventually become the unifying protocol for all communications. You can find more information about SIP trunks, connectivity to PSTN networks, and connecting enterprise "islands" in Part 4.

Simplifying Communications Architecture

Communications networks today are complex and costly to operate. When you begin investigating ways to transition to SIP telephony, you may feel rather bewildered by the array of protocols, gateways, security constraints, and quality of service issues. A considerable effort is required to plan, build, and operate these multiple media streams that often coexist on shared physical networks. Rush-hour traffic gridlock, by comparison, is an easy problem to solve.

SIP offers the promise of a single unifying protocol for all communications. With SIP being widely deployed in both service provider and enterprise networks, the need for gateways that translate one protocol to another (for example, IP to Time Division Multiplexing, or TDM) is eliminated. Proprietary signaling protocols give way to a single standard interface for all connectivity, whether for adding endpoints, deploying contact center adjunct services, or even connecting to trunk services for external communications. See Part 4 for more details on trunking with SIP.

An equally important foundation of SIP is the concept of *distributed intelligence.* This concept, evident in exciting new peer-to-peer (P2P) architectures such as Avaya one-X Quick Edition, creates a new paradigm in communications, requiring no PBX or communication server, only intelligent phones and other endpoint devices as the mechanism for establishing a working communications system. We discuss Quick Edition in more detail in Part 6.

SIP scales well for the smallest businesses, where SIP-enabled endpoints can be established in the absence of centralized proxies and registrars. We discuss how SIP can power small offices in Part 6.

Building a Foundation for Standardized, Intelligent Communications

SIP, defined in IETF standards, is a structured, text-based protocol that is modeled after HTTP, or HyperText Transport Protocol, the language that powers the World Wide Web. Because SIP is text-based and similar to HTTP, application developers and system engineers will have an easier time developing and integrating applications with communications systems.

SIP's architecture consists primarily of SIP endpoints and SIP servers. Endpoints are also called *user agents* — the programs and devices that actually perform the communications between end-users. In small organizations, the user agents can be smart enough to communicate to one another without the need for servers. In large enterprises, centralized SIP servers such as proxies, registrars, and presence servers, facilitate user agent communications. SIP components, and even an example call scenario, are found in Part 2.

SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) is an important standard that facilitates instant messaging communication. SIMPLE is essentially a standardization of SIP's *presence* features. We describe SIMPLE in more detail in Part 2.

Where Will SIP Take Communications in the Future?

SIP, extensible and versatile as it is, continues to grow and evolve. In the near future, you'll likely see SIP become integrated into business applications with several types of functionality — far beyond simple click-to-call hyperlinks. Applications will be able to make communications routing and other decisions based upon interaction with users.

We're predicting that the multiple addresses associated with various modes of communication (IM, text messaging, e-mail, phone) will collapse into a single SIP user id. This single user id, coupled with SIP presence servers, will put communications (with the right people and in the right medium) at your fingertips, no matter what kind of communications device you or they are currently connected to.

SIP may also follow the lead of E911 emergency location services in cellular networks by using a user's known physical location to make even better decisions about the initiation of communication sessions.

We don't want to give away all of our predictions here. Turn to Part 7 for more prognostications about SIP.

Part 2

SIP at a Glance

In This Part

- Working with existing protocols
- Extending SIP to multimedia sessions
- Getting to know SIP presence
- Examining SIP components
- Following an example of basic operations

SIP is an application layer Internet protocol for establishing, manipulating, and tearing down communication sessions. You can do a lot more with SIP than set up telephone calls. The protocol is designed to be *extensible* — meaning SIP can be easily extended to accommodate video, instant messaging (IM), and yet-to-be-invented communications media and features. (XML is an example of another extensible language.) Aside from supporting communication call setup and tear down, SIP also currently supports extensions for instant messaging as well as advertising and tracking user availability (both familiar today in Yahoo! and AOL Instant Messenger).

SIP is used to identify, locate, and enjoin parties who wish to communicate using any peer-to-peer media type. However, SIP does not transport the media itself: that is handled by codecs within the communications programs or devices.

Based on Existing Internet Standards

Although SIP may seem new, it's actually based on many protocols that are widely used today across the Internet and in many enterprise applications. The IETF community took Internet standards as a model, and used a text-based request/response model at the heart of the SIP protocol.

If you use Web browsers (and who doesn't?), then you already depend on a protocol very similar to SIP, called HTTP (HyperText Transport Protocol) — yep, that bit before a standard Web address that you usually take for granted. SIP is modeled after HTTP, and in fact uses much of HTTP's syntax and semantics. Both are *text-encoded* protocols, which means that they are easy to read and debug. This readability promotes integration across a decentralized architecture (such as the Internet) and interoperability across a distributed network. In effect, SIP is to converged communications what HTTP is to information exchange for the World Wide Web (WWW) — it makes the communications infrastructure transparent to end-users and enables ready access to many modes of communication. Just as pointing your browser to an HTTP site enables you to play video, download pictures, or upload files, SIP too has been designed to support multimedia communications.

SIP goes beyond HTTP by embedding in communications the intelligence to sense the media capabilities of the end device as well as the availability of a user to communicate.

Getting Down to One Address for Everything

One key feature of SIP is its ability to use an end-user's *address of record (AOR)* as a single unifying public address for all communications. So, in the world of SIP-enhanced communications, a user's AOR becomes her single address that links the user to all of the communication devices or services that she uses. For example, user Eileen Dover's AOR would look like `SIP:eileendover@company.com`. Using this AOR, a caller can reach Eileen's multiple communication devices (known as *user agents* or UAs to techno-types) without having to know each of Eileen's unique device addresses or phone numbers.

To complement the AOR, SIP provides a mechanism called the *Uniform Resource Identifier (URI)* that establishes a common addressing scheme for all of an individual's user agents. The format of a URI address follows the same basic format as a Web or e-mail address: `contact-address@domain`. By applying this style of addressing, SIP can map the unique addresses of a user's multiple devices and services to a communication domain, and then link all the user agents to a user's single AOR for that domain. Following are some examples of how this URI might be applied:

- **A phone:** `sip:408-555-1212@company.com;user=phone`
- **A fax:** `sip: 408-555-1214@company.com;user=fax`
- **An IM user:** `sip:eileendover@company.com`

REMEMBER

A user typically has just one SIP AOR, such as `eileendover@domain`. Then, each of the user's devices has its own URI, such as `sip:408-555-1214@company.com; user=fax`.

SIP "business cards"

One nifty benefit of SIP is how easy it makes the transition for end-users who are still using traditional communications devices, like, um, telephones. For example, imagine professors at a university who want to make it easier for their colleagues, including researchers at other universities, to contact them. To do so, they are replacing multiple contact numbers on their business cards with a single easy-to-remember SIP address. The problem? The university has started to deploy SIP, but most of the professors are still using traditional (analog) phones, and a complete update to SIP is still years away. As a solution, Avaya Handle-Based Dialing works with the university's *Lightweight Directory Access Protocol (LDAP)* directory server to reach any non-SIP phones by translating the SIP AOR in real time to a standard telephone number. Now, the professors can hand out their business cards and be reached at their existing phone through one simple address, such as: `SIP:CoolestProf@oncampus.edu`. (Part 3 explains how you can also still dial this number on a traditional phone.)

Because a SIP URI supports both numeric (phone numbers) and alphanumeric (Internet-style addresses) formatted contact addressing, the *public switched telephone network (PSTN)* and the Internet can be seamlessly linked together. With SIP, users can potentially contact any user, whether they are on the PSTN or on the Internet.

Enabling Voice, Video, and IM, Oh My!

In keeping with the IETF philosophy of defining simple protocols with powerful functionality, SIP follows a peer-to-peer architecture containing a small set of different methods (types of messages). At the same time, SIP is also very modular and extensible, enabling you to integrate SIP into your existing legacy communications environment. As a result, SIP can interoperate with many traditional telephony protocols and scenarios, as well as with emerging communications services. These properties make SIP an ideal protocol for any company implementing a standards-based converged communications network.

SIP is not designed simply to replace the PSTN. Rather, SIP goes well beyond traditional telephony by facilitating any type of peer-to-peer communication session, instant messaging, video gaming, conferencing, and collaboration.

SIP is also not designed to be a one-stop shop for protocol needs. You essentially use SIP to set up and tear down *media sessions* (for example, IM, text, voice, or video communication sessions). SIP combines with other network protocols as well as application-layer technologies to provide complete end-to-end functionality. One such protocol is the Session Description Protocol (SDP), which carries within it information about the session that you're setting up (namely, the type of media, the codec to use, and the protocol for actually transporting the media).

A SIMPLE Presence Solution

One of the key SIP extensions defined by the IETF is SIP for Instant Messaging and Presence Leveraging Extensions (or even more simply, SIMPLE). SIMPLE defines functions for presence and instant messaging.

In an advanced communications network, users can specify their communications preferences and availability. This feature is known as *presence* — an intelligent "network overlay" that makes it easy for a calling party to reach an available and willing-to-be-called party. Presence streamlines the communication process by enabling users to inform others of their status, their availability, and how they can be contacted before a communication session even begins. Many devices can provide presence information, and it becomes extremely powerful when integrated across all of the user's communication devices such as IP phones, cell phones, softphones, PDAs, and wireless/Bluetooth appliances. As Part 5 explains, realization of such a vision requires not only open interoperability, but also well-built ecosystems that promote cooperation between many industry players, standards, and protocols.

Presence is not limited to a single user; it can also apply to a group of users (for example, Finance Group) or a device (for example, Phone Status = "Off-hook" or "On-hook"). Even more exciting, both users *and applications* can access presence information, providing the opportunity to create next-generation converged communication applications. For example, your network can deliver new capabilities such as *polite calling* (calls that are less disruptive for the party being called) as when, based on information from her calendar application, calls to an executive can automatically be routed to an assistant if the executive is scheduled to be in a meeting.

Components in a SIP Network

When you are ready to enhance your communications with SIP, it's important to understand the building blocks that you will need to form the foundation of your new SIP-enhanced enterprise.

Agents and Servers. These components can take the form of an additional software program (on a laptop computer, for example), or as an inherent part of a mobile device such as a PDA or cell phone.

Using user agents

User agents (UAs) are applications in SIP endpoints (such as a SIP phone, cell phone, PDA, or workstation, as shown in Figure 2-1) that interface between the user and the SIP network. A user agent can act as either a client or a server. When sending SIP messages, the UA acts as a *user agent client (UAC),* and when receiving messages, it acts as a *user agent server (UAS).* A *back-to-back user agent (B2BUA)* is an application that acts as an intermediary between two parties, but appears as an endpoint to both parties — like a middleman. It serves as both UAS and UAC simultaneously to process session requests.

Figure 2-1: Some typical SIP user agents.

SIP devices can communicate directly if they know each other's URI or IP address, but in practice SIP servers are often used in the network to provide an infrastructure for routing, registration, and authentication/authorization services.

IP-based devices can identify and communicate with one another using IP addressing alone. However, in most cases, your network uses the *Domain Name Service (DNS)* to establish sessions with device names, which DNS translates into IP addresses. Similarly, SIP devices frequently consult directory servers (often by name), which provide endpoint addresses that the devices then contact to set up a call.

Serving up SIP servers

SIP servers provide centralized information and enablement services in a SIP ecosystem. The core SIP servers and their functions are summarized here.

- **Registrar Server.** When users come online, they need to make sure that others are aware that they're available to take and make calls. The Registrar authenticates and registers users when they come online, and then stores information on the users' logical identities and the devices that they can use for communications. The devices are identified by their URIs.
- **Location Service.** As users roam, the network needs to be continually aware of their locations. The location service is a database that keeps track of users and their locations. The location service gets its input from the registrar server and provides key information to the proxy and redirect servers. A SIP proxy or redirect server uses this information to obtain the mapping from logical SIP addresses to physical SIP addresses, so that communication sessions can be properly established and maintained.

- **Redirect Server.** If users are not in their home domains, sessions bound for them needs to be redirected to them. The redirect server maps a SIP request destined for a user to the URL of the device "closest" to the user. For example, if a call is destined for `eileendover@company.com` and the user is on the road, the company's redirect server may reply to the caller's user agent (or to the requesting proxy server) with the contact address of the user's cell phone, so that the incoming call can be redirected to the cell phone.
- **Proxy Server.** A proxy server takes SIP requests, processes them, and passes them downstream while sending responses upstream to other SIP servers or devices. A proxy server may act as both a server and a client, and can modify a SIP request before passing it along. A proxy is involved only in the setup and teardown of a communication session. After user agents establish a session, communications occur directly between the parties.
- **Presence Server.** In order for users to see the presence of their buddies to improve communication, they often refer to a presence server. Presence servers accept, store, and distribute presence information. The presence server has two distinct sets of clients:
    - *Presentities* (producers of information) provide presence information about themselves to the server to be stored and distributed.
    - *Watchers* (consumers of information) receive presence information from the server. Watchers can subscribe to certain users, much like IM users choose which buddies to add to their list.

Now, you may be saying to yourself, whew, that's a lot of servers! However, these functions are usually embedded in a single appliance, such as Avaya's SIP Enablement Services platform, which also supports standards-based instant messaging.

Basic SIP Operations

This section walks you through a basic communication session — how it works, and how SIP supports it. Figure 2-2 illustrates the path of a typical session.

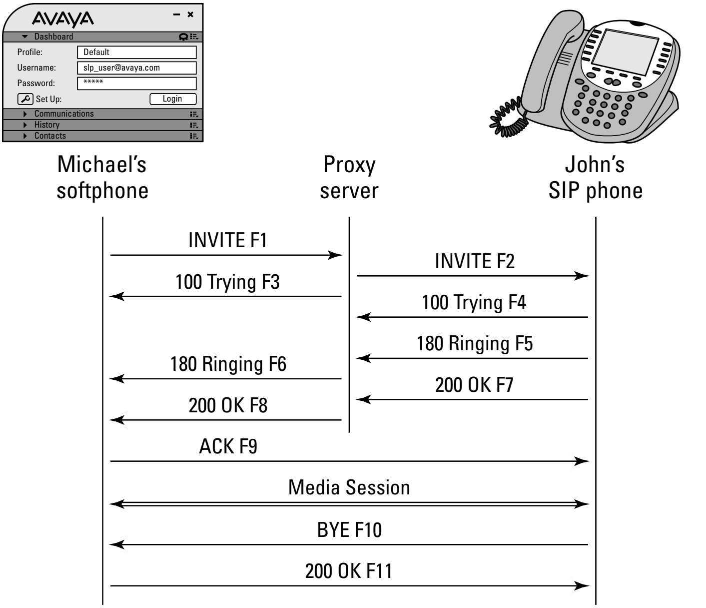


Figure 2-2: A typical SIP session's ladder diagram.

The scenario in Figure 2-2 shows two users — `Michael@smallcompany.com` and `John@bigcompany.com` — using SIP user agents, calling point-to-point through a proxy server. Examples of UAs could be an Avaya SIP Phone, a SIP softphone, or a PDA phone. The proxy server works to connect the two UAs. The communication then follows these steps:

1. `Michael@smallcompany.com` (the UAC) initiates a session by inviting `John@bigcompany.com`.

An INVITE request is generated and sent to John. The INVITE message contains Session Description Protocol (SDP) parameters that define the types of media the caller can accept and where it wishes the media to be sent.

2. A DNS SRV record lookup for SIP services resolves to John's proxy server, `proxy.bigcompany.com`. The INVITE request is sent to the proxy server.
3. The server receives and processes the invitation, and looks up Michael's contact in the Registrar.
4. The Registrar returns `host@officephone.bigcompany.com` where John is currently located.
5. The proxy server generates and sends an INVITE request to the `server host@officephone.bigcompany.com`.
6. The UAS at `host@officephone.bigcompany.com` asks John whether he wants to accept the call. John may hear a ring, see a text message, or see a blinking LED.
7. John's acceptance is sent to the proxy server.
8. The proxy server sends the acceptance to Michael.
9. Michael's UA responds to the acceptance with an ACK (acknowledgement), which tells the proxy server and John's UA that Michael is ready to start the call.
10. At the end of the conversation, John hangs up his phone. His UAC sends a BYE message to Michael.
11. Michael's UAC responds with a BYE message which ends the session.

Though this call flow describes the initiation of a phone call, the beauty of SIP is that the same basic call flow would also apply for establishing video conferencing or other media sessions.

Part 3

How SIP Transforms User Communications

In This Part

- Getting familiar with presence-enabled communications
- Routing communications with presence
- Choosing your modes of communication freely
- Recognizing benefits for the mobile user

This part describes in detail how SIP, presence, and user preference can enhance the productivity and quality of communications and explains the concept of *user-centric* communications.

To Be Available or Not to Be Available . . . Presence-Enabled Communications

When a user activates his or her communications device (user agent, or UA), it registers its presence on the network, indicating its ability to communicate. Presence distributes the following information:

- User *status* (that is, online or offline)
- User *availability* (such as available, busy, on the phone, or out to lunch)
- User's *desired* contact means (such as instant messaging, desk phone, cell phone, pager, and so on)

Probably the earliest manifestation of presence is the telephone network's "busy signal", signaling to a caller that the party is unable to communicate right now because he or she is already communicating with someone else.

Instant messaging has taken presence a step further, with states that include `Available`, `Away`, `On the Phone`, and `Busy`, plus customizable preferences that include any message that the user wants to share (see Figure 3-1). But unlike IM's customizable states, SIP's presence states are generally predefined, which permits predictable routing decisions to be made based upon a user's specific presence.

Figure 3-1: Enterprise presence and buddy list.

Presence doesn't just apply to people and need not apply to just one entity; it can also be associated with a device or a group. For example, a presence status might capture the status of a device (`Phone Status = Off-Hook`) or the status of a user (`User Status = Online`). Presence for *composite entities* like contact center groups or shared documents can be similarly represented.

Presence means "being there" for your customers

Every company wants to retain customer loyalty through superior customer service, but how can your agents keep all of the info they need for accurate responses? A credit card company is especially under pressure to perform 24/7 for helping customers keep accurate tabs on transactions. Imagine how SIP presence can help in this scenario: A customer planning a trip overseas calls his credit card company with a complex question on monetary conversion rate policies. The call center agent checks her "finance expert" presence tab and sees that internal resident experts are off the phone and available for consultation. The agent clicks the IM tab and is automatically routed to one of the available experts for an instant messaging session. The agent then gets and quickly relays the expert's answer to the customer's question.

Surprised at the quick response, the customer then asks a question regarding a disputed transaction with a merchant. The call center agent brings up the merchant information, which displays the presence and availability of the merchant's call center agents for phone calls or IM, and quickly identifies an available agent who can look up details of the transaction and send it back via a Web-page push. The customer, provided with this information, now remembers not only the transaction, but also leaves with a lasting impression of the responsiveness and expertise of the credit card company.

Using Presence to Route Communications

SIP is unique in its ability to make call routing decisions based upon presence. As Part 2 explains in detail, presence enables users to inform others of their status, availability, and how they can be contacted — before a communication session even begins. When integrated with telephony, a user can communicate status and availability to others through multiple devices such as IP phones, cell phones, softphones, pagers, video conferencing, e-mail, wireless devices, and even TDM phones when tied into an intelligent IP PBX.

Presence can span a number of different communication channels. The aggregated view of a user's presence (that is, the availability across all of an individual's SIP-enabled devices) is called Multiple Points of Presence, or MPOP. MPOP becomes powerful when presence is inferred from observation of a user's actions. Possibilities include:

- Setting the user's status to `Away` when his phone and keyboard are inactive for some time
- Making inferences about a user's presence through mobile device location information
- Checking a user's calendar to see whether he is in a meeting or on vacation
- Checking a user's e-mail to see whether he is reading or sending e-mail, or whether he has an `Out of Office` setting

SIP uses presence to make routing decisions for a variety of incoming communications including:

- Routing incoming calls from a desk phone to a cell phone if the user has indicated that he is *roaming* and prefers calls routed as such
- Classifying nonurgent incoming communications as *polite calls* that the user can choose to answer, defer, or ignore
- Routing urgent incoming calls and e-mail to backup support if the user is on vacation or in an extended meeting

When a SIP proxy (a server that processes and forwards SIP requests between calling and called parties) receives an INVITE (request to communicate), it uses the called party's presence to make a routing decision, sometimes called *forking.* The forking decision may be to a specific party *(an intelligent fork),* or it may send several INVITEs to different addresses *(parallel forking).*

REMEMBER

Forking is an old UNIX term where a process "clones" itself into two or more new processes. In the SIP context, forking refers to SIP's use of sending multiple simultaneous INVITEs to other parties in an attempt to initiate a communication session.

The Internet Engineering Task Force (IETF) has a work group that is working to standardize the SIMPLE protocol. SIMPLE is based on RFC 3428, 3265, and 3856 but is being enhanced. SIMPLE is increasingly used to interconnect with previously closed public IM networks.

Another protocol called XMPP, or Extensible Messaging and Presence Protocol, was designed and is currently maintained by the Jabber Software Foundation. In 2004, the XMPP working group at IETF published the RFC documents 3920, 3921, 3922, and 3923, to standardize the core XMPP protocol. Gateways can be used to interconnect XMPP and SIMPLE IM networks.

As SIMPLE progresses and as companies like Avaya continue to add versatility features to communications products, presence will likely give way to *rich presence*, where communications are available anytime, anywhere, and through any application. For example, a company's materials handling application can automatically notify a manager when a particular shipment has been received; the application can use SIP presence to make a decision on how to construct and deliver the notification, depending upon the mode of communication that is available at the time.

One Protocol, Many Modes of Communication

Instant Messaging (IM) is one of the most popular modes of real-time communication. Consumer IM solutions, using proprietary protocols, have been available for some time, but such systems suffer from security, privacy, reliability, and functionality gaps. Although the consumer industry has made some efforts to address these problems with small-scale Enterprise IM solutions, scalability and business continuity issues remain.

A SIP-based, open standards implementation of IM enables interdomain IM via an IM gateway that routes messages to other popular public domain IM services — even those that are external to the enterprise. Additionally, SIP also enables the possibility of sending IMs to next-generation cell phones and Wi-Fi devices via a Service Provider Gateway.

Because SIP can determine through the Session Description Protocol (SDP) what type of media stream the answering UA can support, SIP can make intelligent choices on what type of modality to use. (Part 2 explains more about SIP and SDP.)

However, IM is just the beginning. SIP has the unique and natural ability to support multiple media types within a single communication session. This broad support creates a natural solution for providing communications that adapt to the user based on his or her situation and communication device being used.

The term *multimodality* refers to the ability for a user (or device or application) to communicate through more than one mechanism. For example, a user may be able to input text via voice or typing on a keyboard. An application such as the Avaya SIP Softphone may be able to deliver either a text or a voice message.

New SIP-enabled converged communications solutions will enable users to interact with each other or with an application in a variety of ways: input with speech, keyboard, telephone keypad, mouse, and/or stylus, and output such as synthesized speech, audio, plain text, motion video, and/or graphics. Here are some examples that SIP-enabled solutions can facilitate:

- **Voice/IM/video:** A common interface provides access to instant messaging, voice, and video services, tied together with presence. Initiating communications is the same for all modalities, and users can switch from one communication method to another on the fly to best meet their needs, perhaps within the context of a single communication session!
- **Inline translation services:** A SIP request, originating with an English-speaking user, might contain a Web Services request to translate a message into another language for a non-English-speaking recipient.
- **Multimodal messaging:** A SIP-enabled voice messaging system could provide additional features such as:
  - Voice-mail headers to the end-user via text.
  - Display-enhanced voice-mail by delivering text or graphics menus instead of voice-based menus.

- Virtual Business Card with every voice call so that the user can have the caller's contact information available.
- Playback, skip, rewind, pause, slowdown, and speedup buttons using a Graphical User Interface in a screen phone, as opposed to using the telephone keypad to control the presentation of the voice messages.

- **Speech-to-text translation:** In situations where the caller has only a phone and the called party has only a text terminal, a SIP-enabled translation service could provide text-to-speech and speech-to-text translation.
- **Web-based Interactive Voice Response (IVR):** Users may surf the Web as opposed to working their way through IVRs. Such systems could be used from a hotel room to order services, for instance.

Making the Most of User Preference

Another differentiating aspect of SIP is user *preference*. In the SIP world, you can specify all options for communication, and those options function as a single tool. In other words, the user controls how calls are handled, where they are routed, and the type of communication used. Some of the ways you can specify preferences include:

- **Buddy lists:** People on your communications buddy list can be given preference, priority, or additional choices for communication mode.
- **Time-of-day:** You can specify modes of communication based upon the time of day of communications. For example, you might accept a nonurgent call on a cell phone during the day, but after hours direct it to a voice-mail or IVR system.
- **Preference-influenced multimodal communications:** You can choose which medium of communication you want to use or respond with, based upon a wide number of parameters.

By linking multiple modes of communication with user preference, SIP provides a unifying solution that helps to reduce user and operating costs by making even advanced communications more intuitive and more consistent to the user. Instead of requiring the user to interpret and interact with multiple applications, interfaces, and addresses, the communication system adapts to the need of the user.

Catering to the Mobile User with SIP

Because SIP enables a user to associate a single address with multiple communication devices, communication types, or User Agents (UAs), as discussed in Part 2, SIP natively enables mobility-based communications.

Because an AOR can be associated with any number of devices and/or phone numbers, SIP can help your company leverage all kinds of mobile communication devices as part of a SIP-enabled enterprise. Applying the concepts of intelligent forking described earlier in the section "Using Presence to Route Communications," SIP can direct communications to any number of mobile UACs (user aagent client devices) including cell phones, mobile IM devices (such as BlackBerry devices and Treos), and SIP softphones on laptops.

TIP

As with e-mail addresses, users probably won't memorize other users' SIP AORs. Instead, they'll have address books and buddy lists, just like they do on their e-mail systems, cell phones, and IM clients today. A SIP AOR will be just another data field associated with each person or group. When used by a SIP device, the URI will be retrieved and used to start communicating with another party.

Initial efforts to develop SIP-enabled converged seamless mobility solutions might appear to be aimed mainly at wireless voice calls within wireless networks, to lower usage charges and require fewer phones. For example, Avaya and its partners have pioneered the development of multimode SIP phones, with both cellular and Wi-Fi antennae. Although a company can certainly gain cost savings and other efficiencies, the real benefit is that these efforts also serve as a steppingstone towards next-generation communications applications powered by SIP.

Consider a business communications solution where SIP is the common interface providing integration between enterprise networks and service provider networks. While roaming in a service provider network, users can stay in touch with their virtual enterprise anywhere, anytime, any place — as though they had never left the premises.

- Users in such an environment can instantly receive enterprise voice message notifications while out of the office.
- Users, such as doctors, who work across multiple locations, don't have to carry yet another phone, pager, or PDA.
- Services support improves because service management can locate field technicians within a customer location very quickly and provide better service to customers.
- Administrative assistants can use presence to quickly locate their staff members to get attention to a matter that requires immediate action.
- SIP-enabled user devices can respond to a phone call by responding with a short IM that lets the caller know about the person's availability.

SIP is well suited for mobile environments. SIP's registration function is similar to that in cell networks. When a user turns on a SIP device, it registers the user and sends the device's URI to the registrar server, which routes calls to and from the user. This system ties together multiple communication silos (for example, e-mail, IM, desk phone, cell phone, and so on) using a single address that can reach the user regardless of location. Native mobility is one of the reasons that the Third-Generation Partnership Project (3GPP), which is defining specifications for third-generation (3G) mobile systems, has adopted SIP as the primary signaling protocol.

What about the user's highly personalized services such as address lists, buddy lists, and speed dials? SIP preference tools features can make these personal services mobile. For example, Avaya's Personal Profile Manager provides a centralized service that communicates with SIP endpoints to receive, store, and distribute contact lists, access control lists for user presence, and device parameters such as speed dials and feature button mappings — to the active SIP endpoint being used. A Web-based interface, the SIP Personal Information Manager, enables users to securely manage and view their profile and device information using any standard Web browser. The user simply logs in through the endpoint; after the user is authenticated, his stored data is securely downloaded into the device to create a customized user environment.

Keeping pace with the mobile user

SIP enables seamless mobile communications — anytime, anywhere. For example, suppose a bank executive adds a new contact to her personal profile and assigns it a hot button or speed-dial. She then initiates a SIP call through the internal network using her dual-mode cell phone to check voice mail. While still listening to messages, she walks out of the office to her car — and the call switches on the fly to her cellular service provider network.

She then arrives at headquarters still on the cell phone. The intelligent network using SIP detects her presence and switches the call back to the company's wireless network automatically. The executive then finds a mobile user cubicle with a PC and softphone application; upon authentication, her entire contact list and phone features are downloaded. She then checks her buddy list, sees via presence that her new contact is online and available for a phone call, and she initiates a connection. Her contact is out in the operations center, so the network intelligently forks the connection over to his PDA (which contains a SIP telephony client), and the call begins.

Part 4

How SIP Transforms Enterprise Communications

In This Part

- Understanding how SIP and the PSTN work together
- Getting familiar with SIP trunks
- Federating SIP “islands”
- Matching up phone numbers to URIs

SIP will fundamentally improve the efficiency of communications between enterprises and their partners, suppliers, and customers. The initial wave of VoIP benefits has been primarily limited to intra-enterprise communications. Communications between enterprises, even those that are VoIP-enabled, still require a circuit-switched handoff that impacts voice quality, adds complexity, and introduces unnecessary expense through intermediate carriers. But SIP promises to change all of that.

This chapter discusses how the worlds of SIP and the PSTN (public switched telephone network) will be interconnected and how enterprise communications will change through the introduction of SIP trunk services and federation services.

Working Together: SIP and the PSTN

Clearly the telco world with its country codes, area codes, city codes, and telephone numbers will continue to serve many people for some time to come. So, how do you call SIP users with URIs from old push-button phones, and how do you call landline users from SIP-enabled devices?

Fortunately, the mapping between SIP and telephony protocols has been defined. Gateways that link the Internet with the PSTN are widely deployed and used by VoIP users every day. SIP URIs can also be used to carry telephone numbers. For example, `sip:+12025551212@example.com;user=phone` contains the phone number for directory assistance in Washington, D.C.

By *porting* a PSTN telephone number to a SIP/PSTN gateway, incoming telephone calls can be routed to SIP phones, call managers, and PBXs.

Service providers have already adopted SIP for their internal PSTN telephone call routing. With *SIP-to-PSTN interworking* (providing connectivity between these two systems through a defined interface standard) now in place, carriers such as Global Crossing, AGN Networks, and others are working with equipment vendors like Avaya to offer exciting new SIP trunk services to the enterprise market.

Trunking with SIP

Trunking refers to the means used to transport inbound and outbound calls between the enterprise and external entities (including branch offices and other remotely located parties such as business partners, customers, and suppliers). In this section we describe the differences between traditional and SIP trunks, as well as some of the characteristics of SIP trunks that are attractive to businesses.

Marveling at what a SIP trunk can carry

SIP technology can alter the way that a retail business services its customers. Suppose a retailer with a number of store locations wants to offload the task of handling phone calls from its store employees so they can focus on in-store customers, but the retailer has no direct connectivity between its call center and each store.

SIP-based trunking enables a reconfiguration of communications to address the problem. Through SIP-based DID mobility inbound service, the service provider transports local calls to each store over the SIP network directly to the call center.

Without making any changes to the local stores, the retailer is now able to free up store employees to serve in-store customers, improve customer service over the phone by reducing hold times and busy signals, and still retain a local presence to its customers through a local access number. And with SIP trunking, the call center can replace dozens of TDM trunk lines with a single SIP link!

Traditional network model

Before SIP and VoIP, enterprises connected their internal PBX-based telephone systems to carriers via dedicated TDM (Time Division Multiplexing) trunks. Companies paid for them whether they were idle or overflowing, and incurred toll and tariff charges, especially expensive for long-distance calls.

Today, many companies integrate voice and data over IP and link their sites using IP WANs to reduce costs for communications within the enterprise. However traditional PSTN circuits are still used to communicate with their customers and suppliers, partners, and the outside world.

Transforming trunking with SIP

SIP trunks enable enterprises to carry their voice data over a pure IP connection to carrier clouds, rather than through separate circuits as has been the practice for decades. An Enterprise SIP proxy peers with a Carrier SIP proxy, with the appropriate federations and security protections established between them. The IP circuit continues to carry e-mail, Internet, and other corporate traffic as it does today, and voice is simply layered on top of the circuit as another IP application. SIP sets up and tears down voice calls to and from the enterprise over this IP circuit.

On-net calls traverse the carrier's VoIP backbone (which is typically dedicated to voice so that voice quality can be guaranteed). Off-net calls ride the carrier IP network until the last mile, where a gateway converts VoIP to TDM for calls to PSTN parties (see Figure 4-1).

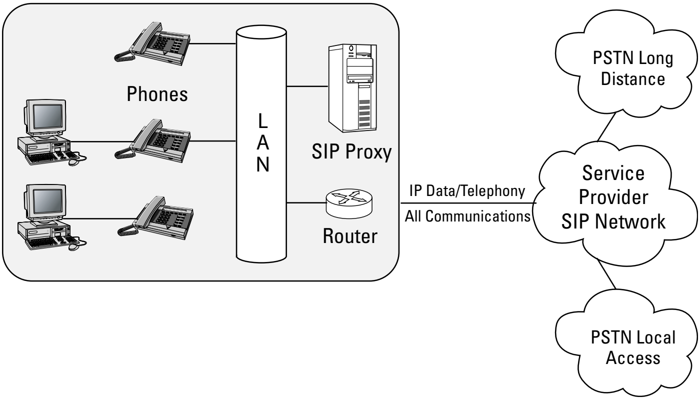


Figure 4-1: SIP trunks change how you make connections to telco carriers.

SIP trunks offer a number of benefits, including:

- **PSTN origination/termination:** Many SIP service providers support origination/termination services directly to the PSTN from their SIP networks. This practice enables the enterprise to reduce the monthly recurring costs associated with multiple TDM circuits by deploying a single IP pipe to the service provider network.
- **DID and 800-Number Mobility:** These features take advantage of the fact that SIP is geographically agnostic. They allow calls destined to a local or 800 number to be

automatically rerouted over the service provider SIP network to another enterprise location. For enterprises, this system offers great flexibility in providing a local presence in all their markets while routing calls to a centralized call center for more efficient service.

- **Cost savings:** For enterprises, SIP networking means reducing the monthly recurring cost of separate PSTN and data circuits to the premises. When you remove voice circuits, you reduce the number of TDM T1 interfaces on the IP PBX, because hundreds of VoIP calls can come from the same hardware footprint as a single T1 interface. Service providers may also offer reduced toll charges to customers when SIP is used as the interface to the PSTN.

Session Border Controllers (SBCs for short) are devices that sit at the edge of a SIP enterprise network and a carrier or ISP network cloud. Among the multiple functions these devices can provide, some SBCs have been designed to support both inward and outward translation of URIs to E.164 (phone number addressing).

A simplified communications Architecture

Enterprises can benefit from the simplification of enterprise networks through the standardization on SIP — this is true for both internal and external communications. As SIP becomes ubiquitous in both service provider and enterprise networks, it introduces a single standard interface for all connectivity, whether for adding endpoints, deploying contact center adjunct services, or even connecting trunk services for external communications. Proprietary signaling protocols and hardware-intensive digital/analog interfaces give way to a simple, logical SIP interface that connects application servers residing on industry-standard platforms. With SIP as a unifying protocol, you can dramatically reduce the need for dedicated hardware gateways and devices. Variants of voice-centric T1 and E1 standards will begin to diminish, as SIP globally standardizes the interface.

REMEMBER

In a converged network, voice becomes an IP application, sharing the common network infrastructure and services.

Connecting SIP Enterprises

A common concern for companies contemplating the switch to SIP is how quickly all these nifty advances are going to come together in a truly comprehensive way. In particular: how can isolated enterprises exchange directory information? Sure, each step toward SIP is a step in the right direction, but how do you get everyone marching in step? The SIP world calls reaching that higher level of inter-connectedness *federating*.

In recent years, businesses have enthusiatically adopted IP for their *intracompany* communications. For example, today more IP PBXs are sold than conventional TDM PBXs.

However, today, most of these IP communications tools stop at the enterprise boundary. IP PBXs "fall back" to TDM and revert to phone calls for communication outside of the enterprise. As a result, "islands" of IP communication are growing, but they are not interconnected. As SIP has matured as a standard, the interconnection of these islands over the Internet has become a technical possibility.

The benefits of such interconnection are enormous. For instance, you can

- Extend enhanced services beyond the enterprise boundary.
- Make available additional modes of communication including multimedia, presence, and IM.
- Enjoy higher-quality connections.
- Reduce costs by bypassing TDM network interconnections.

If two enterprises both have SIP, they can interconnect and enable SIP-to-SIP calling. Beyond calling, they can now interact via multimedia sessions, presence, and instant messaging.

What are the obstacles to this interconnection? To enjoy the full benefits of SIP, you need to overcome three main barriers:

- Enabling SIP islands to discover the existence of one another
- Ensuring that the two islands interoperate
- Managing and controlling this interconnection so that spam and abuse does not become rampant

One answer to all of the above issues is federation. Federation is the mechanism by which you/your company can provide end-to-end SIP communications for end-users. Federation builds a network of open communications within the ecosystem of SIP communications.

Accomplishing this interconnectivity may require one or more of these processes:

- A federation service provides discovery services enabling other SIP islands to be discovered from just a telephone number or address. A number of databases and directories are available today. One such database is ENUM, which is described in the next section, “Mapping Phone Numbers to URIs.”
- A federation service built on top of SIP provides interoperability over a wide range of services and features, from voice, video, presence, IM, and others. A federation service may also provide gateways to inter-operate with other non-SIP devices. For example, a SIP to Jabber gateway could interconnect the SIP federation with a Jabber federation.
- A federation service authenticates all the users and sets policies for reasonable use. A SIP equivalent of caller ID is possible in this model, enabling users to trust calling party indications. This practice avoids the anarchy of e-mail in which anyone can claim to be anyone else, and the resulting avalanche of spam. With policies in place, an enterprise can control how much of its communication is routed over SIP and to whom.

TECHNICAL STUFF

An alternative to federation is *direct peering*. In this mode, two enterprises get together, exchange directory and routing information between them, and set up policies and procedures for communication. Although this system may work among a handful of large enterprises, this bilateral arrangement does not work for hundreds and thousands of enterprises. It obviously does not scale like a federation does.

Federation allows these islands of SIP communications to grow and expand and interconnect. Businesses can maximize the services and features from their SIP products and systems and participate in the growing SIP communication ecosystem.

Mapping Phone Numbers to URIs

When interconnecting islands of SIP, your network needs a method of discovering a SIP URI from a telephone number. The ENUM protocol has been developed to fulfill this purpose.

The ENUM protocol is the result of work of the Internet Engineering Task Force (IETF) Telephone Number Mapping working group. The charter of this working group was to define a Domain Name System (DNS)–based architecture and protocols for mapping a telephone number to a Uniform Resource Identifier (URI), which can be used to contact a resource associated with that number.

TECHNICAL STUFF

ENUM has a DNS-based architecture and protocol by which standard telephone numbers can be expressed as a Fully Qualified Domain Name (FQDN) in a specific Internet domain defined for this purpose (`e164.arpa`). The result of the ENUM query is a series of records (defined in RFC 3402) that gateways, proxy servers, and even SIP endpoints can use to contact one or more resources (for example, URIs) associated with that number.

ENUM and service providers

As SIP becomes increasingly popular with service providers, ENUM will likely become part of the suite of services that a service provider connecting to a SIP Enterprise Network can deliver through the PSTN. On your SIP phone you'll be able to punch in a URI for your friend's TDM landline (or cellular) phone, and the carrier's ENUM server will magically connect you.

ENUM alternatives

Until ENUM (or a similar standard) is widely adopted by service providers, enterprises using SIP can leverage existing solutions such as LDAP (Lightweight Directory Access Protocol) to store both a user's URI and all associated phone numbers (E.164 addresses). SIP applications can then reference entries stored in LDAP structures to resolve and translate phone numbers to URIs.

Vendors are already expressing support for LDAP interconnectivity. For example, with a simple plug-in, Avaya's SIP Enablement Services platform can interface with LDAP to send calls bound for SIP URIs to standard telephones on the enterprise's IP PBX.

Part 5

SIP Interoperability

In This Part

- Getting greater interoperability within a company
- Examining interoperability between companies
- Looking at industry interoperability efforts
- Integrating multiple vendors

SIP is successful and widespread because it is open, extensible, and driven by the IETF. Great strides have already been made to make SIP interoperable among the growing community of SIP-enabled hardware and software products. In this part we discuss interoperability from the perspectives of operating within enterprises and among enterprises.

Internal to the Enterprise

SIP is the most interoperable multimedia signaling protocol to date. SIP has approximately 20 standard features that function admirably across multiple vendors' equipment, as tested periodically at *SIPit* and *InteropNet Labs* (iLabs) events. Avaya and other vendors actively test interoperability of their SIP services and endpoints at these events, further advancing the openness of their solutions.

Although SIP is considered a mature protocol, it isn't 100 percent complete as a standard. Standards are still being developed, especially in the privacy and security areas, and enhancements to the feature set will continue for the foreseeable future. For example, the IETF is still finalizing some fundamental architecture decisions regarding SIMPLE (see Part 3 for more about this instant messaging protocol).

Generally, SIP-adopting companies can expect a greater likelihood of interoperability among proxies, phones, and gateways. This integration is quicker and easier because these devices are modeled after traditional telephony devices with more predictable features and logic.

Interoperability for other services such as firewalls, Interactive Voice Response systems (IVRs), and conferencing servers is less proven because these applications aren't as rigorously tested at interoperability events. Because these types of products tend to have more advanced and proprietary features than familiar voice-centric products, interoperability with these types of applications will be more complicated. In such cases, the application-level logic built into these systems is much more complex than simply setting up a session. Important supporting protocols, such as those being developed within the XCON working group for centralized conferencing, are being developed in the IETF.

External from the Enterprise

Interoperability with the PSTN is achieved through gateways and through SIP-T, a protocol used to carry ISDN signaling within SIP messages.

Connecting carrier SIP networks to enterprises is possible today, as described in Part 4. However, because so many implementation options in carrier and enterprise devices exist, several interoperability hurdles remain.

Some enterprise communications vendors, including Avaya, are actively promoting programs that encourage interoperability with carrier networks. Avaya cofounded SIP Connect (now part of SIP Forum) to define rigorous standards for the carrier-to-enterprise interface. Through its SIP certification program for service providers, Avaya has tested and certified interoperability with a number of service providers.

SIP Interoperability Efforts

SIP is an open standard — and the up-and-coming standard for VoIP and other forms of peer-to-peer and group communications. Like TCP/IP in the 1980s and the World Wide Web in the 1990s, communications hardware and software product vendors are adding SIP capability to their products.

Issues being worked out today include:

- **Development of a reference architecture** of common network elements
- **Specification of the basic protocols** (and protocol extensions) that must be supported by each element of the reference architecture
- **Specification of the exact standards** associated with these protocols
- **Specification of standard methods** for negotiating protocols, protocol extensions, and exchanging capability information between endpoints
- **Definition of authentication methods** to ensure user security and accurate billing

SIP vendors are tackling these interoperability issues now, so that you won't have to face them on your own, especially should your company adopt SIP technology in the near future.

Several groups are actively working on SIP interoperability, including the following:

- **Developer Connection** is Avaya's program that promotes interoperability between Avaya products and others in the market. Go to `http://devconnectprogram.com`.
- **iLabs** (InteropNet Labs) engineers test the interoperability of hundreds of commercial and open source products. The 2005 SIP iLabs is designed to show the state of vendor interoperability using SIP. You can read more at `http://www.networkworld.com/research/2005/050205-ilabs-sip.html`.
- **SipCenter** promotes the development of SIP based products and interoperability. Info can be found at `http://www.sipcenter.com`.
- **SIP Connect** morphed into the SIP Forum as the IP PBX and Service Provider Interoperability Task Group. Information is available from `http://www.sipforum.com`.

- **SIP Foundry** promotes interoperability of SIP products. More information can be found at `http://sipfoundry.org`.
- **SIPit** (SIP Interoperability Tests) test events at locations around the world. More information is available at `http://www.sipit.net`.
- **SPEERMINT** is a new Operations Area working group in the IETF that is working on peering and operational issues of SIP. Find out more at `http://www.ietf.org/html.charters/speermint-charter.html`

These are just a few of the efforts that are helping to accelerate the already rapidly growing adoption and reliability of multi-vendor SIP environments.

Multi-Vendor Integration

One of the challenges many enterprises deal with today is the issue of multi-vendor PBX networking. Traditionally, interconnection has required the use of *Q-interface Signaling protocol (QSIG)* to enable support of supplementary services between systems. This approach provides limited features between systems, but it doesn't address the management complexity caused by the duplication of features and systems, or the user training complexity due to different user experiences with each system.

SIP promises to make all of this much easier — with some caveats, of course! Because it is so open, SIP provides enterprises with more choices for user devices and connecting applications. But although SIP enables basic functionality between some vendors' proxies and other vendors' phones, for example, it isn't yet the silver bullet for all interoperability challenges. Some applications are just too complicated for SIP.

REMEMBER

Remember, SIP is designed to *simply* set up and tear down calls. Although SIP's role is expanding and it now does things for which it wasn't initially intended, it needs help from other standards when it comes to

- Interoperability among IVRs.
- Firewalls.
- Conferencing bridges.
- Instant messaging platforms.

In essence, these systems must rely on much more than standard SIP to make them work together.

Nevertheless, SIP offers new options for multi-vendor integration within an IP telephony environment based on open standards. First, when needed, a specific user agent (a client device such as a PDA phone) can be connected to the PBX through SIP. Next, through federation services, you can support basic connectivity between different vendors' PBX systems using SIP. Soon, SIP will enable enterprises to combine multiple PBXs into a single system, which reduces complexity for both users and administrators.

Part 6

SIP and Server-Free Communications

In This Part

- Implementing SIP communications for small offices
- Getting the big picture on peer-to-peer communications
- Using peer-to-peer SIP for maximum benefits

Deploying SIP in the smallest business settings presents unique challenges. Models for deploying and operating SIP in larger enterprises (distributed or not) may not be optimal for smaller businesses. In this part we discuss some of these challenges and the ways in which SIP-based solutions can be effective in small and distributed business environments.

Meeting Challenges for Small Offices

Small offices — including very small businesses and small branches of enterprises — are becoming more dynamic in form and function and are becoming increasingly distributed. These new work environments introduce challenges including:

- **Capital Costs:** As small offices seek to maintain a more dynamic form that focuses on the localized needs of their markets, they often find themselves balancing the need for adaptability with the upfront capital costs of communications solutions.

- **Deployment and Administrative Costs:** Communications solutions often require on-site technical installation services. Additional costs are incurred when local support is required to fix problems, add capacity, or perform basic administrative tasks.
- **Speed:** For many small offices, competitive advantage is all about speed — time to deployment drives time to market. In contrast, communications solutions deployment is a relatively time-consuming event often requiring detailed planning, staging, and manual administration.
- **Business Continuity:** Business relies on resilient communications that work. Though available, business continuity often requires additional upfront capital costs.
- **Risk Aversion:** Small businesses are more sensitive to disruptive events than are larger businesses. For small businesses, service and responsiveness are critical, and the consequences for disruptive events are great.

Peer-to-Peer SIP Communications Overview

In large enterprises, SIP is most often implemented using centralized services such as proxy servers, presence servers, gateways, and location servers. But SIP endpoints can also function without all of these central services in small-office settings — those environments that can least afford the investment and maintenance required of these big-company services.

A new, simpler variant protocol of SIP, called *peer-to-peer (P2P)* SIP, promises a solution. P2P SIP collapses some of the more complex server functions into the phones (or other endpoints) themselves. P2P SIP relies on the core SIP philosophy that intelligence in communications solutions should reside in the endpoint (refer to Part 2 for more about the variety of SIP endpoints available). Contrast this approach with that of old-fashioned analog telephones that did little more than amplify voice signals and rely upon intelligent switches to function.

The peer-to-peer layer effectively replaces the registration, location, and lookup steps of SIP. It handles three things:

- **Registering** a phone or a user with the peer-to-peer overlay network (when the phone or user joins the network)
- **Looking up** a phone or a user in the peer-to-peer overlay network (when a call to the phone or user is made)
- **Dynamically sharing information** when peers join and leave, so that the load is balanced across peers, and so that the sudden loss of one or more peers doesn't cause the peer-to-peer network to lose track of its current registrants

Figure 6-1 shows how this process works.

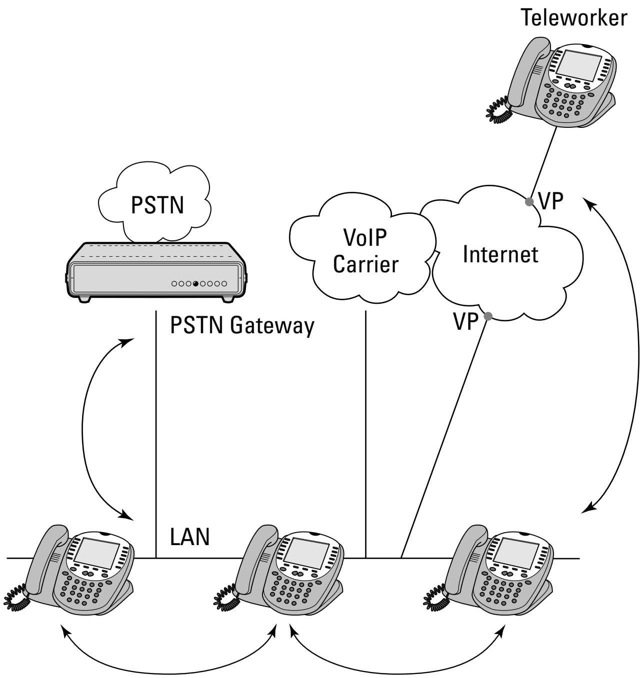


Figure 6-1: The peer-to-peer SIP discovery process.

Reaping the Benefits of Peer-to-Peer SIP Communications

With peer-to-peer SIP technology, you can dramatically simplify telephone system setup and installation. Plug the IP telephones into the local area network, and the system configures itself. In minutes, all users have access to the most commonly used set of features, including voice mail, conferencing, auto-attendant, and call management. A simple PSTN gateway also acts as a peer to the phones and can provide access to the PSTN. By distributing the workload out to the telephone system, peer-to-peer SIP increases reliability by eliminating the problem of a single point of failure wreaking havoc on the entire system.

Avaya one-X Quick Edition

Avaya recently announced a new solution, Avaya one-X Quick Edition, that is based on peer-to-peer technology and can serve both very small businesses and small branches of larger enterprises. Quick Edition is best suited for small offices who desire select communications features. The solution complements Avaya's existing SIP product portfolio, fits into the vision of providing customers one common user experience across devices and applications (one-X), and is another step in Avaya's plan to make intelligent communications applications available to all customers regardless of size. Peer-to-peer technology is an important communications style, and this new solution clearly illustrates Avaya's determination to maintain leadership in IP telephony innovation.

For example, with Quick Edition, setting up a phone system has never been easier. All you need to do is plug your Quick Edition phones into the network. Within minutes, the phones themselves

- Discover each other.
- Assign extensions and populate a company directory.
- Set up voice mail automatically.

You're done! You can pick up the phone and use it.

Peer-to-peer technology also enables sophisticated backup of phones and voice mail. Remarkably, you can achieve all this with no significant additional performance burden on the network. The peer-to-peer solution can easily grows with the needs of your business. As you add employees, simply add additional telephones — it's that simple.

Although peer-to-peer SIP does not require a server, it can work with SIP proxy servers, such as Avaya SIP Enablement Services, to provide a networked solution for enterprises with distributed small offices. Essentially, this solution uses SIP routing capabilities to efficiently connect distant locations that have peer-to-peer SIP.

Part 7

SIP and the Future of Intelligent Communications

In This Part

- Increasing business applications' productivity with presence
- Improving the ease and speed of multimodal communications
- Becoming smarter about prioritizing messages
- Roaming around with location-aware services
- Maintaining privacy
- Enabling new services

To write this part, we got out our crystal ball and polished it up a bit, waved a cell phone over it, and chanted, "Presence, Presence, Presence, PRESENCE!!" Then we just typed what appeared . . .

Presence-Enabled Business Applications

Combined with Web services and XML-based applications, SIP will enable presence *within business applications*. Desktop programs that have references to business contacts within them will be enabled to show the presence of those contacts, on the screen, within that application. You won't need to toggle to an IM client to view the presence of the contact.

Applications will not need to be customized to account for new SIP capabilities. Rather, common libraries (such as dynamic linked libraries (DLLs) in Windows) will contain function calls that know how to query presence servers and return rich and meaningful results.

Here's an example: A factory assembly line worker is viewing the parts needed for an upcoming order in an inventory program, and notes that there is a shortage. He can immediately view the presence of the line manager and procurement manager of his own company, and possibly even that of the parts supplier, from within the inventory management application itself. Using this data, the worker can communicate status to the individual who is most likely to be able to act quickly.

Integration with Business Applications

Taking the scenario described in the previous section, "Presence-Enabled Business Applications," a step further, SIP will enable the line worker to click-to-conference with all contacts that are present, firing up a collaborative conference call to discuss the inventory problem. Open Web services will even allow the inventory application itself to check the presence of all relevant contacts, and interact with a SIP-enabled audio bridge to proactively out-call to the parties, removing the need for a human to start the process. In fact, each participant will have mini-applications (that provide a view into the part of the application being discussed) operating within his conference calling software. These mini-applications will provide views back into the inventory application, permitting participants to submit queries or make changes in the application. Cool huh!?

When coupled with unified communications across multiple access devices, presence will fundamentally change the way people communicate.

Seamless Use of Multiple Devices

SIP presence will improve the productivity of businesspeople by enabling them to seamlessly use multiple communication devices. Today, most users view the presence status of a buddy only as it pertains to their IM/desktop status, resulting in inefficiencies in the way they communicate. For example, they waste time playing voice/e-mail tag and by instant messaging someone simply to ask, “can u talk?”

SIP provides options to collect and advertise presence and preferences in a buddy list, but the *presentity* can now be a desktop, a mobile phone, or another PC application. With a unified approach, SIP will show the presence on whatever device the user happens to be using, instead of simply being “idle” or “out-to-lunch” based only on the desktop status.

Also, devices will be smarter and aware of their owners’ *preferences,* including which modes to communicate in depending upon a variety of conditions. For example, if you were present on your cell phone, but not your desk phone, your buddy would know to click-to-call you instead of instant messaging you while you’re moving 60 miles per hour down the freeway.

REMEMBER

Presence refers to the ability for SIP-based communications to become smarter by facilitating communications based upon a user’s preferences and ability to communicate.

Presence-PBX Integration

When SIP/SIMPLE IM is integrated with an IP-PBX, desktop presence will include the on/off-hook status of the buddy’s phone, all in the same buddy list. This use of presence means that a caller can see whether his buddy is on the phone or not, eliminating blind phone calls or e-mails.

Combining presence from multiple SIP devices will inform the caller that the user is present or not, but the caller will not need to know on which number to call his buddy. He simply launches a message to `sip:buddy@company.com`, and the SIP server will start the session with the correct device (cell phone, desk phone, IM) at the correct time using the correct mode of communication (voice, text, video, and so on).

Further Optimization

As more applications, devices, and networks become SIP aware, communications will be further optimized.

For instance, presence-aware messaging servers could sense that the called party is available on his IM client, but not on the desktop phone that was being called. As a caller leaves a voice-mail message, instead of simply dropping the message in an inbox, an IM may be sent telling the called party that a message has arrived and to the calling party that the called party has been alerted.

A speech-to-text tool can convert the voice-mail message to a text-based IM or e-mail message (again, depending upon the presence of the recipient). Coupling presence servers with application servers will add much more decision-making intelligence about where the message should be delivered, improving the speed at which users respond and communicate.

Communication Systems That Learn

As users acquire more SIP-enabled means of communication and as applications become SIP- and presence-aware, users *could* be crushed under an avalanche of messages, some of which are vital, and others of which are not. So SIP will help devices and applications become smarter about prioritizing.

REMEMBER

With more devices and applications using SIP, the rules for presence will become increasingly complex. Software applications for managing presence (and, what to do when users or applications want to communicate at all times of the day and night) will improve accordingly. It is likely that tools that manage and aggregate presence will emerge in the coming years to manage and aggregate presence from multiple sources.

As presence management tools evolve, they may take on "learning" characteristics (that is, catching on to user habits from heuristics or patterns of usage). Instead of relying upon a static set of business rules, they will adapt to changing conditions (such as volume and source of messages) and make changes to user preferences automatically.

Location-Based Services

Presence will enable new location-aware services for consumers and enterprises. Devices and SIP presence servers can interface with cellular carriers' location-based services (which are used for E911 among other things) to obtain and act on a user's approximate geographic location. Also, presence-enabled wireless access points and micro-cellsites can sense that a user is "roaming" within his service area and inform SIP presence servers of a user's approximate location.

For example, when a mobile user enters a conference room with a presence-enabled Wireless Access Point or micro-cellsite, his presence can be sent to a SIP-enabled room-scheduling system. The system can check to see whether the meeting room is reserved — if it's not, the system can IM the user asking how long he would like to reserve it. If the room was already booked, an IM can be sent to the user's PDA informing him that the room is booked and by whom, and offering alternate locations that are close by.

Ensuring Privacy

Privacy becomes a major concern as presence is enabled on multiple user devices. Users need ways to control their environments so as not to be buried by spam, spyware, or interruptions. With a personal profile manager, users will be able to control presence settings for all their devices from a secured personal portal. They can set preferences to allow or deny others to see their presence based on time of day, location, the device they're currently using, and other factors. Such control will help to address big-brother type privacy concerns.

Personal profile managers will need to be aware of privacy requirements in various parts of the world, and alter their behavior accordingly.

The Internet Multimedia Subsystem

The Internet Multimedia Subsystem (IMS), the Third-Generation Partnership Project's (3GPP) next-generation wireless/wireline communication services reference architecture, is a highly visible example of how SIP is enabling new services. IMS is a standard architecture for telecom operators to provide next-generation mobile and fixed multimedia services. It uses a VoIP implementation based on a 3GPP standardized implementation of SIP, all running over IP. IMS will not only provide new services but all the services that the Internet provides. Users will be able to access their services when roaming as well as from their home networks.

Part 8

Top Ten Reasons for SIP-Enhanced Communications

In This Part

- Simplifying communications
- Gaining more control
- Working with divergent systems
- Making mobility easier
- Saving money

SIP is a key enabling technology that can advance communications to an unrealized level of flexibility and usefulness. Even though SIP may end up being invisible to end-users, SIP is the glue that will help peer-to-peer communications work better than most people can imagine today.

We believe that most organizations will be using SIP within the next few years. If you're considering using SIP in the near or distant future, this part helps you understand the most important benefits of using SIP — and will help you explain those benefits to other decision-makers in your organization.

Presence-Based Communications

SIP adds intelligence to communications by enabling users, as well as applications, to intelligently connect parties based on their presence (availability). We explain how presence works in Part 3.

SIP also has the ability to support intelligent forking — that is, the ability to route communications to the right person, using the right medium (voice, video, IM), on the right device, and at the right time. Turn to Part 3 for more information on SIP forking.

Preference-Based Communications

Like SIP presence, SIP adds intelligence to communications through giving users control over the parameters of their communications (such as time of day, preferred medium, preferred callers, and so on). This concept is best exemplified through SIP's ability to enable you to control who contacts you on what devices. If your users want more control over their communications environment instead of being held hostage by their communication devices, you can find out more in Part 3.

An Open Standard

The SIP standard is defined in RFC 3261 by the Internet Engineering Task Force (IETF). The IETF is a large open international community of network designers, operators, vendors, and researchers (including Avaya) who are all concerned with the evolution of the Internet architecture and the smooth operation of the Internet. As a result, enterprises have choices for their platforms, devices, and applications. Want to discover more? Turn to Part 2.

Interoperability

Several working groups, including SIPit, SIP Foundry, and SIP Forum, arrange events where companies with SIP hardware and software products can test interoperability with other SIP products. This process helps to promote smoother integration of SIP products in enterprise networks.

SIP's ability to work across a range of systems helps enterprises enjoy more seamless integrations between platforms, devices, and applications, so your company can get more done with less CTI and API programming. Part 5 is the place to go for more information.

Unified Addressing

A single SIP AOR (address of record) provides a unifying identifier that can be used for routing all communication to a user. Simply put, an AOR allows for a single user identity to be mapped across multiple devices so that people connect with people, without needing to know which devices they have and are presently using.

This address eliminates the need for tracking users' multiple phone numbers, e-mail addresses, and IM contact names. Turn to Part 3 for the scoop on AOR.

Operational Cost Savings

SIP trunks are IP trunks from service providers that use SIP for call control and routing, enabling enterprises to create a single, pure IP connection to carrier clouds. Voice traverses the network just like other IP applications.

SIP trunks reduce operational costs by enabling the enterprise to eliminate hardware, software, and recurring network charges associated with using traditional PSTN trunks for voice communications. If you're keen on cutting costs, you can find out more in Part 4.

Simplified Communications Architecture

At the foundation of SIP's philosophy is the concept that intelligence should reside in the endpoint. This concept is evident in SIP's native ability to support peer-to-peer communications. Peer-to-peer environments don't rely on communications servers, gateways, or other intermediate devices to support communications between users.

Peer-to-peer SIP networks are easy to set up and administer yet can include features such as automated attendant, voice mail, and three-party conferencing. This architecture is best exemplified by Avaya's Quick Edition solutions. We explain this in detail in Parts 4 and 6.

Creation of New Services

SIP is a structured, text-based protocol that is modeled after HTTP, or HyperText Transport Protocol, the language that powers the World Wide Web. SIP opens the door to a much larger developer community than traditional CTI, and so offers your company the potential to create competitive advantages with intelligent communications.

Because SIP is based on HTTP, application developers and system engineers will have an easier time developing and integrating applications into their communications environments. Part 2 explains this more thoroughly.

Native Mobility

As more devices become SIP-capable, users will be able to pick up and go at will, but still communicate as if they were in their office. Their presence and readiness to communicate will still be visible to their buddies.

For example, SIP's awareness of a user's communication capabilities will aid international travelers who have to use different cell phones and other messaging devices and protocols in different countries. A caller trying to locate such a traveler need not know the traveler's availability or location: SIP by nature will know how a person can be reached, and facilitate the connection. Take a trip to Part 3 to find out more.

Ease of Implementation and Support

Because SIP is modeled after HTTP as a text-based language, it is easy to learn, troubleshoot, and support. From analyzing network packets to application code, SIP's structured language stands out so that IT people can understand and interpret it. But, if you need security, you can simply encrypt this easy-to-read information.

Streamline your system setup times and troubleshooting processes in no time. Turn to Part 5 if you want to learn more.

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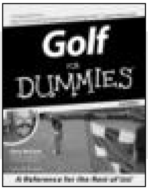

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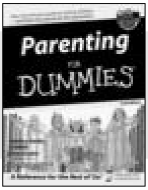

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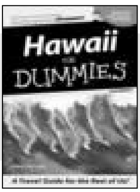

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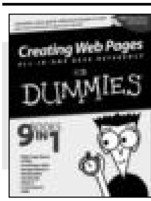

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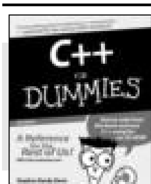

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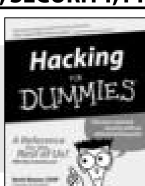

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Realize your full communications potential with SIP!

Integrate SIP with existing systems and simplify your communications architecture!

SIP is transforming instant messaging, voice, and video communications, by adding intelligence and reducing cost. The new world standard for facilitating communications, SIP makes it easier to communicate with the right people, at the right time, on the right device. This book shows you how SIP can be easily integrated into any large or small enterprise to lower cost, improve productivity, and simplify communications.

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Take advantage of SIP for PBX-less communications

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