SIP Communications FOR DUMMIES®

AVAYA 2ND CUSTOM EDITION

by Lawrence Miller, CISSP, and
Peter H. Gregory, CISA, CISSP

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 (Voice over IP) residential and enterprise providers. It is a part of PC operating systems and has been enthusiastically adopted by the open source movement. It is a key part of fixed mobile convergence (FMC) and Unified communications.

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 2nd Custom Edition!

Alan B. Johnston

September 2008

Introduction

You've probably heard a lot about voice over IP ("VoIP") or IP telephony in recent years. There's a lot of industry buzz and excitement about it, but you may be at a loss to explain some of the benefits of VoIP for your business. The real advantage of VoIP and, more broadly, real-time IP communications, is in the way it can transform your business communications architecture into an intelligent communications system using a simple but powerful protocol, known as the *Session Initiation Protocol,* or *SIP.*

SIP makes it possible to easily connect the various components of an overall communications system, rapidly deploy applications, reduce costs, and improve customer service and employee productivity by simplifying — or "SIP-fying" — your organization's communications architecture.

Vendors are increasingly incorporating SIP into their various IP communications products, including:

- Phones, gateways, proxies, and servers
- *Softphones* — software applications that provide voice communication capabilities on PCs, PDAs, and other mobile devices
- PBX (Private Branch Exchange) systems
- Instant messaging (IM) programs
- Videoconferencing systems
- Collaboration systems

SIP is an open protocol standard, with an active working group on the Internet Engineering Task Force (IETF). Avaya and other major companies are active in the IETF SIP working group and in other industry groups, working to ensure that SIP-enabled solutions work across businesses and enterprises with a variety of legacy, new, and evolving networking and telecommunications infrastructures and products to enhance and revolutionize the world of real-time business communications.

About This Book

This book explains SIP from both a business and a technical perspective. You not only learn what SIP is and how it works but, more importantly, how SIP can benefit your entire organization by transforming your real-time business communications to gain a real competitive advantage.

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 into the foreseeable future. Regardless of your role within your organization, this book will help you quickly get up to speed on how SIP can revolutionize real-time electronic communications for your business or enterprise.

How This Book Is Organized

Each chapter of this book covers a different aspect of SIP. You may want to read the book cover to cover to gain a more complete understanding of SIP, or you may prefer to skip around to find out what you need, when you need it. But be forewarned, we didn't just save the best for last. This book is chock full of good information throughout!

- **Chapter 1: SIP at a Glance** provides a brief overview of the SIP protocol including some history about its development as an industry standard and its basic components.
- **Chapter 2: The Case for SIP** describes some of the key features of a SIP-enhanced IP communications infrastructure and its potential benefits and advantages for businesses and enterprises.
- **Chapter 3: How SIP Transforms Communications** explains how SIP enhances customer service and employee productivity, and how SIP scales from the

smallest businesses to the largest enterprises. We cover topics such as presence, AORs, peer-to-peer SIP, SIP trunking, the IP Multimedia Subsystem (IMS), and the service-oriented architecture (SOA).

- **Chapter 4: SIP Interoperability** explains what vendors are doing to make multi-vendor integration with SIP as straightforward as possible. We also cover topics such as hybrid infrastructures, connection points, platforms and applications, endpoints, the communications "ecosystem," survivability, security, and federated presence.
- **Chapter 5: SIP in the Contact Center** describes how SIP transforms your call center into a full-service contact center to help your business better serve its customers.
- **Chapter 6: SIP and Intelligent Communications** looks at how SIP continues to evolve and improve, incorporating more communications technologies and supporting more enhanced communications capabilities.
- **Chapter 7: Ten Reasons to Use SIP-Enhanced Solutions by Avaya** cuts straight to the chase — why your business needs SIP-enhanced communications and how Avaya can help!

Icons Used in This Book

Throughout this book, we occasionally use icons to call attention to material especially worth noting. Here is a description of the icons you'll encounter:

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 highlights technical information that will either make your pocket protector curl or help you fall asleep!

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

Where to Go from Here

Whether you're just hearing about SIP for the first time, considering a SIP project, neck-deep in it, or looking to take your existing telecommunications infrastructure to another level, always keep your eye on the big picture. Avaya has keen vision and a strong commitment to SIP and unified communications. Turn the page and discover for yourself why Avaya is a leader in intelligent communications.

Chapter 1

SIP at a Glance

In This Chapter

- Defining SIP
- Playing nice with others . . . through standards
- Sketching out a simple SIP architecture

People have more options today for communicating with each other than ever, yet we often have a harder time getting through to anyone. We now have a dizzying array of technologies and communication devices, literally at our fingertips, that can deliver voice, text, and even video, in real-time. It seems everyone has too many devices, too many numbers, and too little time.

Yet with more people more connected than ever before, an unintended and unexpected communications paradox has emerged — in our quest to make it convenient for anyone to reach us anywhere, anytime, and any way, it has actually become more difficult to simply communicate with each other. Determining the best way to get in touch with your customers, clients, and partners at any given time can be a daunting task, and letting people know how to get in touch with you at any given time is no easy feat, either. Simply put, communication has become device-centric, not user-centric.

This chapter introduces SIP, a widely adopted industry standard protocol that is helping businesses and enterprises of all sizes solve these issues and enhance communications capabilities.

What Is SIP?

Session Initiation Protocol (*SIP,* as in sipping from a fire hose on a hot day) is an open signaling protocol standard developed by the *Internet Engineering Task Force (IETF)* in cooperation with many industry leaders, including Avaya, for establishing, managing, and terminating real-time communications over large IP-based networks, such as the Internet. Communications via voice, video, or text (instant messaging), may take place using any combination of SIP-enabled devices, such as a softphone on a laptop computer, a wireless handheld device or PDA, a mobile phone, an instant messaging client on a desktop PC, or an IP phone with videoconferencing capabilities.

SIP is an application layer peer-to-peer communication protocol for establishing, manipulating, and tearing down communication sessions. But, you can do a lot more with SIP than just setting up telephone calls. The protocol is *extensible* — meaning developers can easily write custom applications for SIP to accommodate video, instant messaging, and other emerging communications media and features, using tools and programming languages, like Java, that are already familiar to Internet developers. Using SIP, simple-to-develop and quick-to-deploy custom applications can be easily integrated into your communications sessions.

TECHNICAL STUFF

SIP is used to identify, locate, and enjoin parties who want 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.

SIP builds on a number of existing communications protocols and has rapidly become the standard for service integration (how new services and applications are created and combined) within most large fixed and wireless carrier networks. Thus, SIP is positioned as a single unifying protocol that will transform not only communications within an enterprise, but communications between an enterprise and its ecosystem of customers, clients, partners, and suppliers. For businesses that need to sort out and reconnect their current tangle of disparate communications protocols and programs, SIP is a refreshing solution that can simplify and enhance your communications capabilities.

What does SIP have to do with music?

Absolutely nothing! But the original IETF draft of the SIP protocol, published in February 1996, was titled "draft-ietf-mmusic-sip-00" — not quite as catchy as, say . . . *SIP Communications For Dummies!* Those wild and crazy guys at the IETF gave us "mmusic" instead — an acronym for **M**ultiparty **MU**ltimedia **S**ess**I**on **C**ontrol.

Although SIP was originally the brainchild of the telephony industry, which envisioned a better way to set up and manage telephone calls, the computer industry saw the potential to revolutionize *all* real-time communications with SIP. Initially, SIP was developed as a means to invite people to large multimedia broadcasts on the Internet's multicast backbone, known as "Mbone." Mbone was used to facilitate the distribution of streaming multimedia content including educational seminars, broadcasts of space shuttle launches, and riveting IETF meetings.

Today, SIP enables a wide array of services and applications that enhance real-time communications for businesses and enterprises. You can find the latest core SIP specification in IETF RFC (*Request for Comments*) 3261 (now simply titled "SIP: Session Initiation Protocol"), and it's not just a one-hit wonder: Currently, some 289 RFCs related to SIP have been published, showing just how much of a factor SIP is in the communications industry today — and will continue to be for tomorrow and beyond.

Setting the Standard

Although you may not be familiar with SIP by name, it's actually based on many protocols that are widely used across the Internet and in many enterprise applications today. And, just as common standards and interoperability have been key to the success of the Internet and IP networks, SIP is a widely adopted standard that promotes interoperability and drives down costs in communications networks.

You're probably already familiar with a protocol very similar to SIP — HTTP (*HyperText Transfer Protocol*) — yep, we affectionately refer to that bit before the "www" as "H-T-T-P, colon, slash, slash" when browsing to an Internet Web site. In effect, SIP is to intelligent, unified communications, as HTTP is to information exchange on 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 Web site enables you to play a video, download a picture, or read text, SIP has been designed to support multimedia communications in real-time.

SIP is modeled after HTTP, and in fact uses much of HTTP's semantics and syntax. Both SIP and HTTP use a plain text-based language. What does this mean for your business? SIP's messages are easy to program and interpret, making it easier to achieve interoperability between disparate networks and different vendor solutions, saving you money and enabling you to rapidly deploy new applications to support your business and customer needs. SIP is also very modular and extensible (like *XML*, or the *Extensible Markup Language*), allowing for integration with legacy systems and new and evolving technologies. These properties make SIP an ideal protocol for implementing a standards-based unified communications network.

Although the protocols are similar in their simplicity, SIP goes well beyond the capabilities of HTTP, for example, by embedding intelligence in a communications session to sense the media capabilities of an end device and the availability of a user to communicate.

The SIP standard is defined in RFC 3261 by the *Internet Engineering Task Force (IETF)*. For anyone familiar with the battle between Blu-Ray and HD DVD formats, the importance of industry standards cannot be overstated. SIP has been a widely adopted industry standard for more than ten years. Additionally, several neutral consortiums, including SIP Forum and SIP Center, arrange meetings and events, such as SIPit, 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.

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.

Some vendors have gone above and 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 absolutely ensure that they do is for vendors to test their SIP-based products together. For example, Avaya has made a long-standing commitment to establishing openness and interoperability for SIP through its Developer Connection program, which supports software developers, systems integrators, and service providers in testing interoperability and developing SIP-based solutions in conjunction with Avaya products and services. You can find more information about interoperability efforts in Chapter 4.

The potential impact of SIP goes beyond internal communications within a business or enterprise. SIP has become a signaling standard for carrier networks. Service providers now 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 real-time IP-based communications. You can learn more about SIP trunks, PSTN (*Public Switched Telephone Network*) and PBX (*Private Branch Exchange*) connectivity, and connecting disparate enterprise "islands" in Chapter 3.

SIP goes well beyond traditional telephony by facilitating any type of peer-to-peer communication session, including instant messaging, video gaming, voice and video conferencing, and collaboration.

SIP is not designed to be a one-stop shop for all your protocol needs. SIP is used to set up, manage, and tear down *media sessions* (for example, voice, text, and video). SIP works 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).

Basic SIP Components

When you are ready to enhance your enterprise communications with SIP, you'll need to understand the basic building blocks that form the foundation of your new SIP-enabled enterprise.

User agents

User agents (UAs) are applications installed on SIP endpoints, such as an IP phone, mobile phone, wireless device or PDA, or a laptop or desktop PC (see Figure 1-1), that interface between the user and the SIP network. A UA can act as either a client or a server. When sending SIP requests, the UA acts as a *user agent client (UAC),* and when servicing a request, 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 1-1: Some typical SIP user agents.

SIP devices can communicate directly if they know each other's *URI (Uniform Resource Identifier)* or IP address, but more commonly, SIP servers are used in an enterprise network to provide an infrastructure for routing, registration, and authentication and 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 System (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.

SIP servers

SIP servers provide centralized information and enablement services in a SIP ecosystem. The core SIP servers and an overview of their basic functions are described here.

- **Registrar Server.** When users come online, they need to make sure that others are aware they're available to take and make calls. The Registrar authenticates and registers users when they come online, and stores information on the users' logical identities and the communications devices or physical entities (IP address) of the communication devices they can use. 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 need to be redirected to them. The redirect server maps a SIP request destined for a user to the URI of the device "closest" to the user. For example, if a call is destined for `eileendover@avaya.com` and the user is

on the road, the company's redirect server may reply to the caller's UA (or to the requesting proxy server) with the contact address of the user's mobile phone, so that the incoming call can be redirected to the mobile 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 may modify certain parts of 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.** Presence servers accept, store, and distribute presence information that allows users to see the availability of people they want to contact. 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 instant messaging users choose which "buddies" to add to their list.

TIP

Presence is a key feature in SIP-enabled communications networks. Don't worry if you don't yet understand the concept: You can read more about it in Chapter 2.

Now, you may be saying to yourself, whew, that's a lot of servers! However, these functions can often be provided by a single appliance, such as Avaya's SIP Enablement Services platform or the development environment provided by the Avaya SIP Application Server.

A Basic SIP Call Example

This section walks you through a basic SIP communication session — how it works, and how SIP supports it. Figure 1-2 illustrates the session.

In Figure 2-2, two people — `Michelle@smallcompany.com` and `Tony@bigcompany.com` — use SIP user agents (UAs) to make a point-to-point call 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:

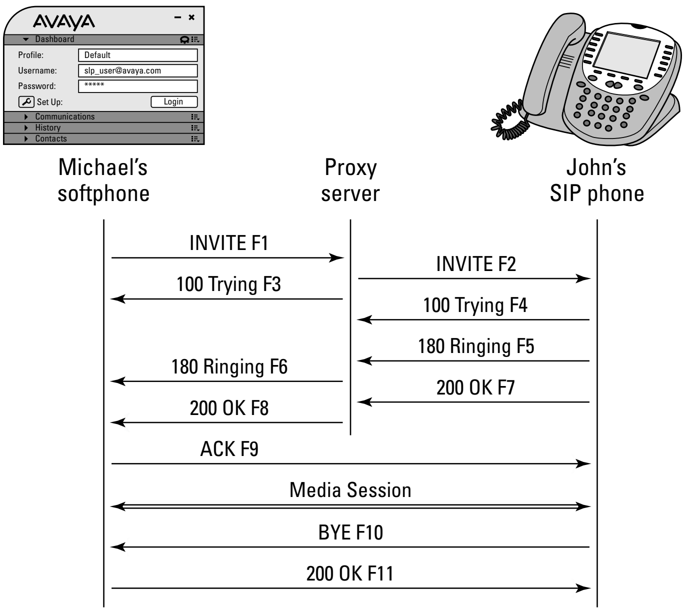

Figure 1-2: A typical SIP session's ladder diagram.

1. `Michelle@smallcompany.com` (the UAC) initiates a session by inviting `Tony@bigcompany.com` and sending this request to the proxy server at `smallcompany.com`.

   Michelle's UA generates an INVITE request, which is sent to the proxy at `smallcompany.com`. The INVITE message contains *Session Description Protocol (SDP)* parameters that define the types of media she is capable of accepting and where she wants the media to be sent.

2. The proxy at `smallcompany.com` performs a DNS SRV record lookup for SIP services at `bigcompany.com` since `bigcompany.com` is a foreign domain. This

record lookup returns `proxy.bigcompany.com`, which is then resolved to a physical IP address by DNS. Michelle's INVITE request is then forwarded to the proxy server at `bigcompany.com`.

3. The `bigcompany.com` proxy server receives and processes the invitation, and looks up Tony's contact in the location database of the Registrar (physical IP address of the UA).
4. The location database of the Registrar returns `host@192.168.1.100` where Tony is currently located.
5. The `bigcompany.com` proxy server forwards the INVITE request to Tony's UA at `host@192.168.1.100`.
6. The UAS at `host@192.168.1.100` asks Tony whether he wants to accept the call. Tony may hear a ring, see a text message, or see a blinking LED.
7. Tony's acceptance is sent back through the `bigcompany.com` proxy, which forwards it to the `smallcompany.com` proxy, which forwards it to Michelle's UA. The body of Tony's acceptance includes SDP parameters defining the selected media chosen from what Michelle had originally offered and where Tony wants the media to be sent.
8. Michelle's UA responds to the acceptance with an ACK (acknowledgement) directly to Tony's UA, which tells Tony's UA that Michelle is ready to start the call.
9. At the end of the conversation, Tony hangs up his phone. His UAC sends a BYE message directly to Michelle's UA.
10. Michelle's UAC responds with a 200-OK message directly to Tony's UA, which ends the session.

Though this call flow describes the initiation of a basic phone call, that simple call flow would be the same for establishing video conferencing or other media sessions using SIP.

Chapter 2

The Case for SIP

In This Chapter

- Connecting people anywhere, anytime, on any device
- Introducing AORs — one convenient address for everything!
- Using voice, video, text, and more!
- Simplifying communications architectures for businesses large and small
- Preparing for a bold new future with SIP

What business initiatives are driving your organization's technology strategy? Perhaps it's improving your customers' experience? Or, increasing the productivity of your employees? Or, minimizing your operating costs? Gee, those all sound good! How about — *all of the above*!

SIP is one solution that can help your business achieve these goals. For example:

- *Presence* allows your employees and customers to communicate more efficiently.
- *User-centricity* gives your users more flexibility and options while improving control of technology and greatly simplifying device (and communications) management.
- *Multi-modal capabilities* provide a richer communications experience and empower your business to better serve your customers.
- *Simplified architectures* promote interoperability while simultaneously allowing your business to reduce costs, rapidly deploy new applications and solutions, and grow with your business.

To illustrate the potential of SIP for your business or enterprise, we examine some of these key features and advantages and show you how SIP can help you address this myriad of business challenges!

The Presence Is Now!

SIP supports a new model for communications through its use of the *Extensible Messaging and Presence Protocol* (*XMPP*, which can also be used for presence) and *SIP for Instant Messaging and Presence Leveraging Extensions* (or more simply, *SIMPLE*). Both are widely adopted open protocol standards defined by the IETF.

In an intelligent communications network, people can specify their communications preferences and availability. This feature is known as *presence* — an intelligent "network overlay" that makes it easier for a calling party to reach an available and willing-to-be-called party. Presence enables people to inform others of their status, their availability, and how they can be contacted before a communication session even begins, thereby increasing productivity by making it easier to reach people more efficiently. Many devices can provide presence information, which is particularly powerful when integrated across all of a person's communication devices, such as IP phones, mobile phones, softphones, wireless devices, and PDAs.

TIP

Presence is not limited to a single person; it can also apply to a group of people (for example, "Finance") or a device (for example, Phone Status = "Off-hook" or "On-hook").

Even more exciting, both people 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) — based on information from her calendar application, calls to an executive can automatically be routed to an assistant if the executive is busy. This can eliminate the need for the caller to leave a voice mail and allows the executive to stay focused on her current meeting.

Making the World Revolve around You with User-Centric Communications

Over the last few years, business communications have become largely *device*-centric. The more devices you use, the more numbers and addresses others need to know in order to reach you. And without presence, as described in the previous section, communication has become a guessing game when trying to connect with people, wherever they may be and whatever they are doing.

Is it a business card or a phone book?

Like many busy people, you probably have a number of ways for others to get in touch with you and you want a convenient way for them to know how to reach you without your business card starting to look more like a phone directory! For example, you probably have some combination of the following:

- At least four phone numbers including home, office (main and direct), and mobile
- An SMS address for text messaging on your mobile phone
- Various instant messaging (IM) screen names or aliases
- A fax number
- Numerous corporate and personal e-mail addresses and aliases (such as `eileen.dover@avaya.com`, `edover@avaya.com`, `eildover@yahoo.com`, and `eileen_dover@hotmail.com`)

Contacting you may have become a guessing game for your customers, clients, and partners. It shouldn't be this hard! The user now controls how people communicate with you, and SIP can help by providing a single address for all your communication devices.

With SIP, your business communication becomes *user*-centric, once again making it easier for you to reach your customers and for your customers to reach you. A SIP *address of record (AOR)* provides one unifying identifier that can be mapped across multiple devices and media types. No more tracking of multiple phone numbers, e-mail addresses, and IM contact names. You can think of an AOR as your "public address."

One Protocol, Many Modes of Communication

New SIP-enabled converged communications solutions enable users to interact with each other or with an application, in a variety of ways. Input can be via speech, keyboard, telephone keypad, or mouse. Various modes of output may include synthesized speech, audio, plain text, motion video, and/or graphics. And best of all — *you* control how people reach you. 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.
- **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 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 cards 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 (GUI) in a screen phone

- **Speech-to-text translation:** In situations where the caller only has a phone and the called party only has a text device (PC, laptop, wireless device), 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 example.

Because SIP uses the *Session Description Protocol* (*SDP*) to determine what type of media stream the answering UA can support, SIP can make intelligent choices for modality.

SIP can also 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 the situation and the 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 can deliver either a text or voice message.

Streamlining Communications Architecture

Communications networks today are complex and costly to operate and maintain. By comparison, a SIP communications architecture consists primarily of SIP endpoints and SIP servers. This means that your business can simplify its communications network and reduce associated costs. Endpoints are also called *user agents* — the programs and devices that actually perform the communications between end-users. In smaller businesses, the user agents can be smart enough to communicate with one another without servers. In larger enterprises, SIP servers such as proxies, registrars, and presence servers, facilitate user agent communications.

SIP scales well for even the smallest businesses, where SIP-enabled endpoints can be established in the absence of centralized proxies and registrars. We discuss SIP for small offices in Chapter 3.

SIP offers a single unifying protocol for all real-time communications. For example:

- 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 TDM, or *Time Division Multiplexing*) is minimized.
- 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.
- In addition to its forward-looking innovative uses, SIP supports the legacy *public switched telephone network* (*PSTN*) and *Private Branch Exchange* (*PBX*) systems, which will be around for some time to come.

See Chapter 3 for more on SIP trunking.

An equally important foundation of SIP is the concept of *distributed intelligence.* This concept, evident in peer-to-peer (P2P) architectures such as Avaya 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.

The Future Is Now!

SIP, an extensible and versatile open protocol, has wide adoption throughout the industry. New solutions that will empower your business and employees to better serve your customers now make intelligent communications routing and other decisions based upon interaction with users, such as a person's known physical location linked to presence.

Chapter 3

How SIP Transforms Communications

In This Chapter

- Enhancing end-user communications with presence, preference, user-centric communications, and mobile devices
- Understanding peer-to-peer SIP for small offices
- Taking a look at SIP trunking, federation, IMS, and SOA for the enterprise

SIP can revolutionize real-time IP communications in any organization, large or small. This chapter takes a look at how SIP-enabled features like presence and addresses of record can enhance the productivity and quality of communications for end users. We also examine SIP for small and medium businesses and discuss some of the unique challenges for small and distributed business environments. Finally, we see how SIP scales to even the largest enterprises and service provider networks, providing the framework for new and evolving architectures such as IMS and SOA, and making SIP a truly universal solution for intelligent communications.

How SIP Transforms End-User Communications

Are you spending more time managing your communications devices and looking up phone numbers or addresses than actually communicating with others? If so, SIP is about to simplify your life and let you control your communications devices, rather than having them control you.

To Be or Not To Be . . . Available — with Presence

When a user activates a communications device (*user agent*, or *UA*), the device registers its presence on the network, indicating its ability to communicate. The concept of presence is somewhat analogous to the telephone network's busy signal, signaling to a caller that you are unable to talk right now because you're already talking with someone else. But, SIP takes presence a step further.

Presence distributes the following information:

- User *status* (that is, online or offline)
- User *availability* (such as `Available`, `Away`, `In a Meeting`, `On the Phone`, and `Busy`)
- User's *desired* contact method (such as instant messaging, desk phone, mobile phone, pager, and so on)

SIP's presence states also permit predictable rules-based routing decisions to be made. These decisions are based on a user's specific presence state, and on customizable preferences that include any information the user wants to share.

REMEMBER

Presence doesn't just apply to people and need not only apply to a single entity; presence can also be associated with a device or 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.

Using presence to route communications

SIP can make call-routing decisions based on presence information by enabling users to inform others of their status, availability, and how they can be contacted — before a communication session even begins. A user can communicate status and availability to others through multiple devices such as IP phones, mobile phones, softphones, instant messaging, pagers, video conferencing, e-mail, wireless devices, and even TDM phones connected to an intelligent IP PBX.

Presence can span a number of different communication channels and provide an aggregate view of a user's presence (that is, availability across all of an individual's SIP-enabled devices). 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 mobile phone if the user has indicated that he is *roaming* and prefers calls routed as such
- Classifying non-urgent incoming communications as *polite calls* that the user can choose to answer, forward, or ignore
- Routing urgent incoming calls and e-mail to others 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 sending multiple simultaneous INVITEs to other parties to initiate a communication session.

Presence means "being there" for your customers

Every successful company strives to provide superior customer service, with call centers or contact centers (you can read more about contact centers in Chapter 5), which are often the first contact an upset customer may have with your company when dealing with a particular issue. But how can your customer service agents get to all the information they need to provide quick, accurate responses for your customers?

A credit card company that provides ongoing support for its customers through a contact center offers one example of how SIP presence can help you deliver superior customer support. The names, places, and events are fictitious, but the possibilities are real:

A customer planning an overseas trip calls with a question about monetary conversion rate policies. The customer service agent checks her "finance expert" presence tab and sees that an internal resident expert is off the phone and available for consultation. The agent clicks the IM tab and is automatically routed to the available expert. The agent then gets, and quickly relays, the expert's answer to the customer. The customer then asks about a disputed transaction with a merchant. The agent brings up the merchant information, which displays the presence and availability of the *merchant's* customer service agents for phone calls or IM, and contacts an available agent who can look up details of the transaction and send it back via a Web-page push. Complete customer service with a smile (or maybe just a smiley face icon)!

AORs — One address to rule them all

Another 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. With 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, Eileen Dover's AOR might be `sip:eileendover@company.com`. Using this AOR, you can reach Eileen on any of her multiple communication devices (her UAs) without having to know each of her unique device addresses or phone numbers.

To complement AORs, SIP supports *Uniform Resource Identifiers (URIs)* that establish a common addressing scheme for all of an individual's user agents. A URI address follows the same basic format as a Web or e-mail address: `contact-address@domain`. Using this format, 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. Some examples of how a URI might be applied include:

- **A phone:** `sip:908-555-1212@company.com;user= phone`
- **A fax:** `sip: 908-555-1214@company.com;user=fax`
- **An IM user:** `sip:eileendover@company.com`

REMEMBER

A user typically has just one SIP AOR, such as `sip:eileendover@domain`. Each of the user's devices then has its own URI, such as `sip:908-555-1214@company.com;user=fax`.

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 the Internet.

TIP

As with e-mail addresses, users probably won't memorize other users' SIP AORs. Instead, they'll use address books and buddy lists, just like they do on their e-mail systems, mobile 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 communicate with another party.

Better mobility with SIP and IMS

Because an AOR can be associated with any number of devices and/or applications, SIP can leverage all kinds of mobile communications devices as part of a SIP-enabled enterprise. By applying intelligent forking , SIP can direct communications to any number of mobile UACs (user agent client devices) including mobile phones, wireless devices, and softphones or other applications installed on a laptop computer.

Initial efforts to develop SIP-enabled mobility solutions focused on voice calls within wireless networks to lower usage charges and require fewer phones. For example, Avaya and its partners pioneered the development of multimode SIP phones with both mobile and WiFi capability. These developments served as an important step towards next-generation SIP-enabled communications applications.

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 can instantly receive enterprise voice message notifications while out of the office.
- Users working across multiple locations don't have to carry yet another phone, pager, or PDA.
- Services support improves because managers can quickly locate field technicians within a customer area to provide better responsiveness.
- SIP-enabled user devices can respond to a phone call by responding with a short text message that lets the caller know the person's availability.

SIP is well suited for mobile environments. SIP's registration function is similar to that in GSM and 3GPP 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, and mobile phone) using a single address that can reach the user regardless of location.

What about customized options such as address lists, buddy lists, and speed dials? SIP preference features can make these customizations 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 SIP endpoint currently in use. A Web-based interface, the SIP Personal Information Manager, allows users to securely manage and view their profile and device information using any standard Web browser. The user simply authenticates through the endpoint and his stored data is securely downloaded to create a customized user environment.

Native mobility is one of the reasons the *Third-Generation Partnership Project (3GPP)* has adopted SIP as its primary signaling protocol for the *IP Multimedia Subsystem (IMS)*. Other reasons for using SIP as a core underlying technology for supporting IMS include simplicity, flexibility, extensibility, and familiarity — recall that SIP is a plain-text, open protocol standard similar to HTTP that simply establishes, manages, and terminates real-time IP communications sessions over a wide array of mediums including voice, video, and text (refer to Chapter 1).

IMS provides a framework for innovation within a service provider network, enabling rapid development of new and innovative multimedia applications and content over a mobile network. SIP is the key to delivery of these innovations from IMS to mobile networks and users.

Keeping up with mobile users

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 initiates a SIP call through the internal network using her dual-mode mobile 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 mobile service provider network. She then arrives at the main office still on her mobile phone. The intelligent network using SIP detects her presence and switches the call back to the company's wireless network automatically. The executive 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.

How SIP Transforms Small Office Communications

Small offices — including small or mid-sized businesses and small branches of large enterprises — are becoming more dynamic in form and function and are becoming increasingly distributed. These work environments must address challenges that although not unique, can be nonetheless daunting for small offices, 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.
- **Operating and administrative costs:** Communications solutions often require on-site technical installation and maintenance 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. Developing new applications that integrate with complex communications architectures through open but proprietary APIs (*application programming interfaces*) can be a time-consuming event requiring detailed planning, staging, testing, and debugging.
- **Business continuity:** Small businesses are often more sensitive to disruptive events than larger businesses. The survivability of a small business may be threatened by even a relatively minor or short-term event lasting only a few days.

TIP

Peer-to-peer (P2P) SIP is one solution for small- and mid-sized businesses. P2PSIP collapses some of the more complex server functions into the phones (or other endpoints) themselves. P2PSIP relies on the core SIP philosophy that intelligence in communications solutions should reside in the endpoint (refer to Chapter 1 for more about the variety of available SIP endpoints, or user agents). Contrast this approach with that of old-fashioned analog telephones that do little more than amplify voice signals and rely upon complex and fixed-cost PBX switches to provide communications functionality.

Avaya IP Office

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked branch and head offices for small and mid-size businesses. The award-winning IP Office gives growing companies a complete solution for telephony, messaging, networking, conferencing, customer management, and unified communications, thereby helping businesses reduce costs, increase productivity, and improve customer service.

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 yet with redundant edge devices that understand multiple home-proxies. But, SIP endpoints can also function without all of these central services in small-office settings bringing the advantages of SIP to small and medium businesses, such as cost reduction, converged voice/video/data over SIP trunking, presence, and UC applications.

Avaya is an active member of the IETF P2PSIP working group that is working to define a P2PSIP protocol standard and to address issues such as security and privacy in a peer- to-peer communications network.

P2PSIP effectively replaces the registration, location, and lookup steps of SIP. It handles three functions:

- **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

With P2PSIP technology, you can drastically simplify telephone system setup and installation. Plug an IP telephone into your local area network, and it configures itself. In minutes, all users have access to the most commonly used features, including voice mail, conferencing, and auto-attendant. A simple PSTN gateway also acts as a peer to the phones and can provide access to the PSTN. And a peer-to-peer solution easily grows with your business. As you add employees, simply add telephones.

How SIP Transforms Enterprise Communications

SIP fundamentally improves the efficiency of communications between enterprises and their partners, suppliers, and customers. The initial benefit of IP communications has been primarily limited to intra-enterprise communications. Communications between enterprises, even those that are VoIP-enabled, still largely require a circuit-switched handoff that impacts voice quality, adds complexity, and introduces additional expense through intermediate carriers. SIP changes all of that by interconnecting SIP communications architectures and the PSTN, and with SIP trunking and federation services.

Enterprises can benefit from the simplification of enterprise networks through SIP standardization for both internal and external communications. As SIP becomes ubiquitous in both service provider and enterprise networks, a single standard interface for all connectivity is available for adding endpoints, deploying contact center adjunct services, or even connecting trunk services for external communications. Proprietary signaling protocols, including variants of voice-centric T1 and E1 standards, 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.

Working together: SIP and the PSTN

Clearly, the telecommunications industry's system of country codes, area codes, city codes, and telephone numbers will continue to serve many people around the world for some time to come. So, how do you call SIP users with URIs from a plain old telephone system (POTS) using a dial-up telephone, and vice versa?

Fortunately, the mapping between SIP and telephony protocols has already 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:+12125551212@example.com;user=phone` contains the phone number for directory assistance in New York, New York.

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

Many 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) in place, carriers throughout the world are working with SIP 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 trunks and SIP trunks, as well as some of the characteristics of SIP trunks that are attractive to businesses.

How SIP trunking can help you improve customer service

SIP can improve the way a retail business services its customers. Suppose a retailer with several 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 customer service center and each store.

SIP-based trunking enables a reconfiguration of communications to address the problem. Through SIP-based DID (*Direct Inward Dialing*) mobility inbound service, the service provider transports local calls to each store over the SIP network directly to the customer service 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 customer service center can replace dozens of traditional *TDM* (*Time-Division Multiplexing*) trunk lines with a single SIP link!

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 busy, and incurred toll and tariff charges, particularly expensive for long-distance calls.

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

SIP trunks enable enterprises to carry their voice data over a pure IP connection to carrier clouds, rather than through separate “voice-only” circuits. 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, Web, data, and other corporate traffic as it does today, and voice is simply added to the mix 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 3-1).

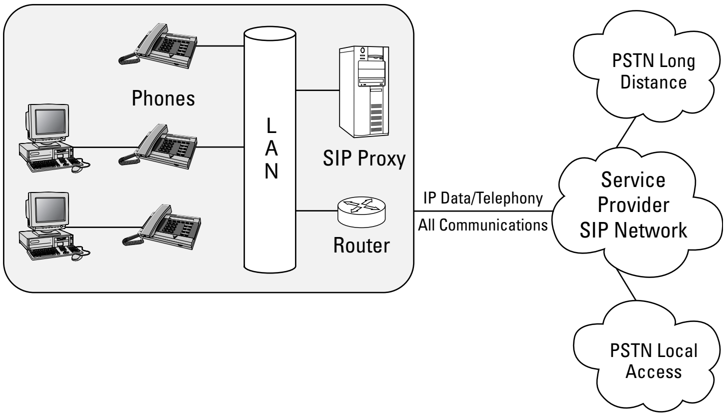

Figure 3-1: SIP trunks change how you make connections to carriers.

SIP trunks offer a number of advantages, including:

- **PSTN origination/termination and cost savings:** Many SIP service providers support origination/termination services directly to the PSTN from their SIP networks. This allows the enterprise to reduce monthly recurring costs associated with multiple TDM circuits by deploying a single IP pipe to the provider network.
- **DID and Toll-free Number Mobility:** These features take advantage of the fact that SIP is geographically agnostic. Calls destined to local or toll-free numbers can 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.

Session Border Controllers (*SBCs* for short) are edge security 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).

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

Where no one has gone before — connecting enterprises to federated services

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 answer: federation.

In recent years, businesses have enthusiastically adopted IP for their internal communications. For example, today far more IP PBXs are sold than TDM PBXs. However, most of these IP communication infrastructures 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, IP communications "islands" are growing, but they are not interconnected. With SIP, interconnection of these islands over the Internet is accomplished. The benefits of interconnection are enormous. For example, you can

- Extend SIP-enhanced services beyond the enterprise boundary
- Make additional modes of communication available, including multimedia and instant messaging
- Establish higher-quality connections
- Reduce costs by bypassing TDM network interconnections

If two enterprises both use SIP, they can interconnect and enable SIP-to-SIP calling, interact via multimedia sessions and instant messaging, and exchange presence information.

What are the obstacles to 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 interconnections so that spam and abuse do not become rampant

One solution to these issues is federation. *Federation* builds a network of open communications within the ecosystem of SIP communications. Accomplishing this interconnectivity may require one or more of the following:

- A federation service that 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. One such database is ENUM, described in the next section.
- A federation service built on top of SIP that provides interoperability over a wide range of services and features such as voice, video, presence, and IM. A federation service may also provide gateways for interoperability with non-SIP devices.
- A federation service that authenticates users and sets policies for acceptable use. A SIP equivalent of caller ID is possible in this model, enabling users to trust a calling party. This helps avoid the anarchy of e-mail in which anyone can claim to be anyone else (spoofing).

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 enterprises, direct peering is cumbersome when connecting multiple enterprises and obviously does not scale like a federation.

Mapping phone numbers to URIs

To interconnect SIP islands, your network needs a method of discovering a SIP URI from a telephone number. The ENUM protocol has been developed by the IETF Telephone Number

Mapping working group for this purpose. 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 (`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 associated with that number.

As SIP continues to be adopted by service providers, ENUM will 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 mobile phone, and the carrier's ENUM server will seamlessly connect you.

An alternative to ENUM is LDAP *(Lightweight Directory Access Protocol),* which can store both a user's URI and all associated phone numbers (E.164 addresses). SIP applications can then reference entries stored in LDAP to resolve and translate phone numbers to URIs. Many applications already support LDAP interconnectivity. For example, Avaya Communication Manager with either Avaya SIP Enablement Services or Avaya SIP Application Server development environment can interface with LDAP to send calls bound for SIP URIs to standard telephones on the enterprise's IP PBX.

Enterprise SOA and intelligent communications

Service Oriented Architecture (SOA) is a methodology that focuses on three major objectives:

- Designing IT infrastructures that allow different applications to interoperate and exchange data via a common communications protocol
- Leveraging existing applications by exposing their capabilities as services that can be easily used by other applications
- Using these services to automate business operations

Avaya Unified Communications

Avaya Unified Communications combines SOA and SIP, along with business-intelligent technologies that integrate business-context knowledge and user communication context knowledge. The end vision is a comprehensive, multi-channel communication access architecture that helps increase business agility through rapid, intelligent responses to business events that can find and connect the right people, at the right time, with the right devices. Unified Communications has intelligent integration that enables businesses to keep their decision-making processes moving towards resolution, whether for application-to-application, human-to-application, or application-to-human communications related tasks.

SOA is possible in an advanced, intelligent communications infrastructure based on a value proposition of strategic value versus economic efficiency. At the heart of the strategic value proposition is the promise that communications capabilities, when closely integrated with business processes and applications, can greatly enhance the speed and ability of enterprises to respond to changes in customer and market demands.

SIP-enabled communication services extend and simplify communications within enterprise processes with a portfolio of telephony, messaging, speech, and contact center solutions.

SIP integrates easily with the Web service environments being developed for many business critical applications. For enterprises, this means the introduction of powerful multimodal communications, embedded as a service within their business applications. It also means that information on user presence can be incorporated into application business logic. Communication becomes a tool not just for users, but also for the enterprise's critical business applications.

Chapter 4

SIP Interoperability

In This Chapter

- Getting acquainted with industry and vendor interoperability initiatives
- Examining interoperability within the enterprise
- Looking at interoperability beyond the enterprise
- Appreciating SIP security and survivability

As previously discussed, SIP is an open and extensible IETF protocol standard that has gained widespread support and acceptance among a growing community of hardware and software vendors. In this chapter, we discuss interoperability within enterprises and among enterprises, including topics such as hybrid infrastructures, the communications "ecosystem," survivability, federated presence and security.

Proving That We Can "All Just Get Along!"

SIP is an open protocol standard developed to help facilitate real-time IP communications and other forms of peer-to-peer and group communications. Communications hardware and software vendors have recognized the universal appeal of the SIP protocol and are adding SIP capabilities to a rapidly growing number of products. Various interoperability initiatives currently under way 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

Several organized groups and communities are actively working to promote SIP interoperability, including:

- **Developer Connection** is Avaya's program to promote interoperability between Avaya products and others in the market. Go to `http://devconnectprogram.com`.
- **SipCenter** promotes the development of SIP-based products and interoperability. Go to `http://www.sipcenter.com`.
- **SIP Forum** promotes industry interoperability by hosting live testing events, defining and creating compliance tests, and developing industry-wide technical recommendations. More information is available at `http://www.sipforum.com`.
- **SIPconnect** is a standards-based initiative of the SIP Forum to promote direct peering between SIP-enabled IP PBX systems and VoIP service provider networks by ensuring interoperability between SIP trunks and legacy TDM environments. Go to `http://www.sipforum.com/sipconnect`.
- **SIPit** (SIP Interoperability Tests) is a week-long semi-annual test event held at locations around the world for the purpose of promoting global SIP interoperability. More information is available at `http://www.sipit.net`.
- **SPEERMINT** is an IETF working group established to address peering and operational issues in real-time IP communications sessions. Learn more at `http://www.ietf.org/html.charters/speermint-charter.html`.

These are just a few examples of the interoperability initiatives that are accelerating the already rapidly growing adoption and reliability of multi-vendor SIP environments.

Multi-Vendor Integration

One challenge many enterprises face is 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 does provide limited interoperability between systems, but it doesn't address the management issues caused by the duplication of features and systems, or the training complexity arising from different user experiences with each system.

Although SIP isn't a silver bullet, it does provide enterprises with more options for multi-vendor integration within an IP telephony environment based on open standards. For example, a variety of user agents (such as a softphone) can be connected to the PBX using SIP. And, with federated services (refer to Chapter 3), you can support basic connectivity between different vendors' systems, devices, and applications using SIP. Soon, SIP will enable enterprises to combine multiple PBXs into a single system, which reduces complexity for both users and administrators. Interoperability within the SIP ecosystem, which spans a wide range of telecommunications and networking technologies and services, is key to SIP's widespread adoption and rapid growth.

REMEMBER

Avaya is committed to fostering SIP interoperability, reflected both in its extensive portfolio of SIP products and applications and its interoperability testing and certification programs such as DevConnect, the SIP Ecosystem Partner program, and the BusinessPartners program.

TECHNICAL STUFF

Avaya is clearly committed to SIP interoperability among various vendor products. At a recent industry interoperability event (Interop 2007), Avaya was the only major telecommunications vendor to successfully demonstrate basic SRTP interoperability for both PBX's and phones. We discuss SRTP in more detail later in this chapter.

Avaya Intelligent Presence Server

With the introduction of Avaya Intelligent Presence Server in early 2008, Avaya became one of the first SIP vendors to "crack the code" on federated presence, providing a scalable, high performance presence aggregation service that collects and disseminates rich presence from Avaya and other third-party sources. Using rich presence, users gain the ability to more effectively reach the people they need, leveraging the multiple channels of communications available to them, no matter what vendor applications, platforms, or end-devices they are using.

REMEMBER

SIP is designed to *simply* set up, manage, and tear down calls and sessions. Although the SIP protocol is expanding, other protocols are still necessary to address interoperability with systems and applications such as:

- Interactive Voice Response *(IVR)* systems
- Firewalls
- Conferencing bridges
- Instant messaging platforms

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

Internal to the Enterprise

Enterprise telecommunications infrastructures can often be classified in one of three distinct phases in their evolution.

In the first, or traditional phase, enterprises have separate infrastructures for voice and data networks. *Time division multiplexing (TDM)* is used for voice, and *Internet protocol (IP)* is used for data.

This paradigm is sooo twentieth century (literally)! Prior to 2000, ISDN and TDM voice networks used the Signaling System #7 (SS7) set of protocols to set up and tear down telephone calls on the PSTN. Applications were developed to interface directly with connection-management protocols, which interfaced directly with ISDN- and TDM-based access protocols.

In this phase, interoperability between voice and data systems is largely non-existent. Your telecommunications circuits and data circuits are separate and unable to share excess capacity or offload excess traffic. Thus, your network may be crawling while your phones sit idle, or your Internet connection may be blazing fast but your increasingly frustrated customers get continuous busy signals when trying to call you. And, you get to pay the same circuit rates whether they are busy or idle!

This also means that your various devices and applications don't interoperate. Your office phone is just your office phone and your mobile phone is just your mobile phone — two separate phone numbers to reach one person, and two separate voicemail systems. And e-mail, instant messaging, video conferencing, and on and on!

In the second, or converged networks phase, enterprises build out their IP networks to leverage a common infrastructure that flattens, consolidates, and extends their voice and data networks. This enhances the IP network to meet enterprise-class criteria: improving *quality of service (QoS)* and increasing the reliability of real-time, mission-critical business and communication applications. Applications, built on the *Computer Telephony Integration (CTI)* overlay, interface directly with connection management protocols, which interface with H.323 and IP-based access protocols.

"Ripping and replacing" everything in one step is often too expensive for organizations, particularly given their large capital investments in legacy PBX equipment. Still, organizations can attain significant cost savings now by implementing core SIP routing (using a gateway if needed) and trunking, by deploying selected SIP applications such as conferencing, mobility, and Unified Communications, and by taking advantage of SIP features such as centralized management of dialplans and on-net calling.

Finally, in the third phase, enterprise communications capabilities are closely linked to business processes and applications to deliver real business advantage, allowing an organization to quickly adapt to changing customer needs and market situations.

In this phase, interoperability of systems and applications is the key to success. Peering and loose coupling promotes rapid development and deployment of applications with SIP as the unifying protocol between applications, connection management, and access-layer protocols. In this model, innovative uses of SIP include intelligent customer routing in contact centers (discussed in Chapter 5) and voice portals “at the edge” providing customers with self-service options.

External from the Enterprise

Interoperability with the PSTN is achieved through gateways, SIP trunking (explained in Chapter 3), and through SIP-T, a protocol used mostly by service providers to carry ISDN signaling within SIP messages.

Many enterprise communications vendors, including Avaya, actively promote interoperability programs with carrier networks. Avaya co-founded SIPconnect (now part of SIP Forum) to define interoperability specifications 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 Security and Survivability

Because SIP relies heavily on an IP-based network and utilizes a plain-text language similar to HTTP, SIP-enabled applications are potentially vulnerable to many of the same security threats that plague corporate networks and the Internet today. These include authentication, authorization, and privacy issues, denial of service and buffer overflow attacks, and *SPIT (SPAM over Internet Telephony)*.

TECHNICAL STUFF

Fortunately, many of the same solutions to these problems are effective for securing SIP implementations as well. These may include:

- HTTP Digest Authentication using MD5 for challenge/response user authentication
- Signaling channel encryption using *Transport Layer Security (TLS)* for end-to-end session security

- *Certificate Authorities (CA)* for authentication in networks using SRTP and TLS
- *Secure Real-Time Transport Protocol (SRTP)* or *IP Security (IPSEC)* using *Advanced Encryption Standard (AES)* encryption to provide authentication, confidentiality, and integrity for protection of the media (to prevent eavesdropping, for example)

Other security concerns are not necessarily unique to SIP, but are nonetheless threats that must be addressed with innovative solutions. For example, spam has become an unfortunate part of our modern e-mail lexicon, and SPIT may be an equally vexing problem, if not even more ubiquitous than spam.

Another major concern for enterprises is survivability of their telecommunications and network systems. Traditional TDM telephone equipment and the PSTN are commonly perceived as highly reliable, dedicated networks and systems compared to distributed, “best effort” networks such as the Internet.

Although a high degree of redundancy is built into the individual components of expensive TDM PBX systems, having redundant PBX systems in remote locations capable of providing seamless failover during a major systems malfunction or failure, or a catastrophic event is rather uncommon.

By comparison, IP-based communications systems and networks, in addition to having built-in redundant components, are commonly deployed as duplexed systems or server farms in multiple locations. SIP survivability and redundancy includes benefits like re-routing DID’s and quickly redirecting calls to alternate data centers (or having the Service Provider do so after a specified timeout) — something traditional PSTN circuits don’t provide. Even SIP endpoints can failover to third party SIP gateways as part of the “survivable intelligent edge.”

REMEMBER

Avaya is #1 in the industry for survivability and redundancy and is firmly committed to promoting interoperability among all SIP-enabled vendor systems.

Chapter 5

SIP in the Contact Center

In This Chapter

- Defining a contact center
- Using inbound, outbound, self-service, and blended centers
- Transforming the contact center with SIP

No company operates in a vacuum, devoid of contact with customers or the general public. If you have the need to communicate with others outside of your company (who doesn't?), then you are a prime candidate for developing or using a contact center.

This chapter introduces contact centers — what they are and how they benefit customers and companies, and how SIP can truly transform your contact center into a world-class service organization.

Call Center Versus Contact Center

You may be asking yourself "What's the difference between a call center and a contact center?" You may think of a call center as a group of people sitting in neat rows beside their phones, answering customer calls. But contact centers are much more than large rooms filled with headset-wearing switchboard operators. The modern contact center handles phone calls, e-mail, and online communication — including instant messaging.

Traditionally, contact centers *have been* known as call centers. The newer name — contact center, and even customer service center — reflects the fact that more than just phone calls are being handled, including e-mails, IM/chat, Web, video, and video kiosks.

Some companies choose to separate the handling of customer contacts by communication medium. For example, a company may establish a department for inbound calls, one for outbound calls, and another for e-mail. Some companies opt to create "universal" or "blended" agents who handle all contact types. Companies create universal contact agents for efficiency and service. Training agents to use multiple communication methods to provide product or service information is often more effective than training multiple agents to use a single communication method to provide the same product or service information.

Contact centers deal with almost any type of contact for a company including phone calls, e-mails, online communications (such as instant messaging or chat windows), Web, one way and two-way video, and video kiosks.

Inbound and Outbound Contact Centers

Contact centers communicate with customers in a number of ways, but who initiates the contact defines the type of contact center. If the outside world initiates contact, then the contact center is said to be an inbound contact center. Conversely, if the contact center itself is responsible for initiating contact, then the contact center is said to be an outbound contact center.

Customers contact inbound centers to buy things, such as airline tickets; to get technical assistance with their personal computer; to get answers to questions about their utility bill; to get emergency assistance when their car won't start; or for any number of other reasons for which they might need to talk to a company representative.

Increasingly, companies are looking to inbound call centers for proactive customer service that could be used for cross-selling and up-selling opportunities.

In outbound centers, representatives from the company initiate the call to customers. Companies might call to notify a customer that a product ordered has arrived, to collect an unpaid invoice, to follow up on a problem the customer reported, or to survey customer satisfaction or solicit ideas for product or service enhancements.

Outbound contact centers are, most often, very telephone-centric because of tradition and perception. It is not unusual for a company representative to call a customer on the phone; by contrast, simply sending the customer an e-mail may seem too impersonal and ineffective.

Self-Service Contact Centers

A new breed of inbound contact center is starting to emerge — the *self-service* center. In traditional contact centers, all interaction between the customer and the center is done with human agents. However, in self-service centers a good portion of the load is being shifted toward non-human automated response or speech-enabled systems.

Automated response systems prompt customers to use their phone keypads or voices to answer questions that help to route their calls appropriately. Each button push or response brings the caller closer to the desired information. Automated response systems have been around for years, giving the customer access to simple (and common) information, such as mailing addresses, driving directions, account balances, and procedural instructions. These systems can also be used to route calls to the most appropriate human agent able to answer the call.

Transforming the Call Center

There are many ways to transform your call center into a world-class contact center and thereby gain efficiencies and greatly improve your customer service capabilities.

Many companies are adding more channels of contact to create an enhanced experience resulting in higher customer satisfaction. Market research indicates that the new standard for contact centers includes a combination of phone, e-mail, self-service, and Web communications capabilities.

If your contact center is running separate call-handling groups (customer service and collections, for example), then by merging these groups you can take advantage of the economies of full-service contact centers. You can continue to have call-handling groups logically separated, but with the ability for available agents in one area to handle overflow in another area.

Implementing an IP contact center can help make your operation more scalable, which means your company can grow without costly build-outs or investments. Instead, you can utilize and manage remote agents — through your IP infrastructure — as if they were physically located in your office.

You can also benefit from improved agent occupancy by blending work into your contact-handling queue. A classic example of blending is mixing outbound telemarketing into an inbound sales or service queue. In this case, you make use of agent idle time — time agents spend waiting for incoming calls — to do your outbound work. As a result, your agents are busier (more occupied) overall. Should inbound volumes increase or spike, your agents can stop making outbound calls while they handle inbound calls. Other work such as e-mail and chat can also be blended.

SIP in the Contact Center

The transformation from call centers to IP-based contact centers has moved beyond the early adoption stage to the mainstream as IP communications matures and its benefits are recognized.

Businesses are leveraging the power of open IP standards to simplify application logic and to eliminate needless redundancy in the infrastructure and operations of their contact center. The result is that every agent and expert has full access to the required features, functionality, and information needed to serve customers better on each call or contact.

IP-based contact centers are simultaneously lowering networking costs, distributing the agent workforce, and enhancing agent productivity through multi-modal communications. Customers can reach agents by whatever means they choose (e-mail, instant messaging, Web chat, video, or phone call) and the agent can then route the request as appropriate.

SIP further extends these advantages by facilitating a more efficient and collaborative communication model between the enterprise and its communications ecosystem.

For example, SIP allows agents to take advantage of presence information to determine the status of enterprise subject-matter experts and then to easily connect with them. With integrated instant messaging, agents can communicate with supervisors, peers, and experts while remaining on the call with the customer. Bottom line: Customers make one phone call and they're done! No need for frustrating follow-up calls to get through to the right person.

The productivity gains can be immediate. When answers are needed, an agent can contact an expert either within or outside of the contact center via a choice of communication options. The agent simply checks the availability of experts for phone consultation or instant messaging and contacts the expert using either method. This capability allows enterprise knowledge workers to become on-demand experts whose availability can be determined in real time and then easily located and contacted.

A common metric for assessing contact center efficiency and customer satisfaction is the number of first-call resolutions. Integrated SIP and presence helps raise this resolution rate because experts are now just a mouse click away. Agents don't have to transfer customers to other agents, forcing callers to repeat their questions or issues. Talk times are often reduced since the agent doesn't have to hunt for experts. Overall call volumes can be lowered when customers get their answer on the first try and don't have to call back. The contact center is more efficient, and customers are happier.

Extending Customer Service Beyond the Contact Center

Many contact centers tend to operate as standalone operations, separate from the rest of the enterprise. Expertise within the enterprise, and among its suppliers and partners, typically is not available as a resource to assist with customer interactions.

The costs for contact centers to replicate knowledge, already present in the enterprise and its ecosystem, too often strain already limited resources. The skills-based routing paradigm of transferring calls in search of expertise can result in long waits, dropped connections, and customer frustration.

SIP presence facilitates a more efficient and collaborative communication model that will help merge not only enterprise communications with contact center processes, but will also integrate the enterprise ecosystem. Through a standard mechanism for communicating presence, contact center resources across enterprises can also be linked.

SIP fundamentally changes the reach of the contact center call-routing decision. In effect, everyone within the enterprise and its ecosystem of partners and suppliers can become a contact center resource. The contact center no longer has to rely solely on dedicated agents. No longer is the contact center an isolated business function, but an essential part of a business process that involves and leverages all resources to improve the customer experience.

Building an Open, Modular, and Simplified Contact Center Architecture

Today, contact center and call center architectures reflect a strong hardware-centric model based on a PBX and ACD *(Automatic Call Distributor)* heritage. The current state-of-the-art contact center includes a variety of applications such as skills-based routing, screen pops with CRM *(Customer Relationship*

Management) applications, multimedia customer input channels, and ICR *(Intelligent Customer Routing)*.

Many companies have multiple contact centers located across the country or around the world. While this approach provides a local presence for the business, it creates additional costs and introduces operational inefficiencies. Each location requires servers and applications: the switch, the contact center software, reporting, workforce management, and so on. When these components are replicated for each contact center site, the cost and complexity can be significant. Add the requirements for communications, call transfers and call coverage between sites, and the cost and complexity rise even more.

Contact center solutions based on IP have been addressing these issues by introducing a "flatten, consolidate, and extend" approach to consolidating multiple contact centers over an IP network. A single central location (and perhaps a second for survivability) serves as the heart of the operation, providing the intelligence and the contact center applications. Other sites serve as a gateway off of the centralized hub. Cost-effective communications between the central site and satellite locations is achieved by using IP for the communication path. With centralization, the network is flattened and costs associated with multiple instances of each application are drastically reduced. One consolidated contact center now serves the business — eliminating the need for network pre-route solutions and their associated costs and complexities.

With Intelligent Customer Routing (ICR), the consolidated IP contact center also provides a new paradigm for customer service. ICR is a tool that provides businesses a new way to treat customers and respond to their specific, individual needs by leveraging customer data to route every customer interaction in the most efficient and cost-effective manner to provide the highest-touch, highest-value customer experience.

In addition, this approach becomes cost effective for extending the contact center to areas of the business that previously did not have coverage. Companies gain a larger agent pool by eliminating the geographical constraints they've previously had to consider. Providing consistent customer support and a consistent brand image becomes easier. IP becomes an enabler of business transformation and saves real dollars.

Session management with Avaya Communication Manager and SIP Enablement Services

Avaya Communication Manager and SIP Enablement Services continues to evolve as an open, highly reliable, and extensible real-time communications platform that enables Unified Communications and Contact Center applications. Reduced TCO, in the short term, and increased ROI on the entire SIP investment, in the mid- and longer term, is achieved through expanded functionality, which includes

- Routing SIP sessions across the network
- Centralizing SIP registration and location services
- Extending geographic redundancy with a distributed, highly reliable, presence-enabled SIP and SOA architecture that enables massive scalability from the smallest deployments to more than 250,000 users per system
- SIP trunk termination at the access point controlled by session management and redirected to any application
- Adding session management, enhanced presence, common management and a standards-based software development environment to existing voice and video features
- Enabling applications to be decomposed and distributed across the network and introducing application sequencing

SIP can take this a step further by introducing a single standard interface for all connectivity, including 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. This new modular server architecture can be software-centric, which will simplify upgrades and promote greater flexibility by enabling the rapid deployment of new services.

Chapter 6

SIP and Intelligent Communications

In This Chapter

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

Internet-centric design allows SIP to integrate easily with the Web service environments being developed for many business-critical applications. For enterprises, this means the introduction of powerful multi-modal communications, embedded as a service within their business applications. It also means that information on user presence can be incorporated into application business logic. Communications now becomes a tool not just for users, but also for the enterprise's critical business applications.

This chapter looks at intelligent communications as part of unified communications and how SIP facilitates greater productivity for your business and customers.

Presence-Enabled Business Applications

Together with Web services and XML-based applications, SIP enables presence *within business applications*. Desktop programs that have references to business contacts within them will be able to show the presence of those contacts, on the screen, within the application. In other words, you don't have to switch to another application, such as an IM client, to view the presence of a contact.

Applications do not need to be customized to take advantage of new and evolving SIP capabilities. Rather, common libraries (such as dynamic linked libraries — DLLs in Windows, and shared libraries in Linux) contain function calls that know how to query presence servers and return rich and meaningful information.

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 quickly determine the availability of all of the individuals necessary to resolve the shortage issue.

Integration with Business Applications

Taking the scenario described in the previous section a step further, SIP enables the line worker to click-to-conference with all the appropriate contacts that are available, initiating 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. This bridge may also include Web or video conferencing yet remove the need for a human to start the process. Each participant may have mini-applications operating within his collaboration software. These mini-applications can 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 can improve the productivity of businesspeople by enabling them to seamlessly use multiple communication devices. Today, most users view the presence status of a contact, or “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 contact list, but the *presentity* can now be a laptop computer, a mobile phone, or another PC application. With its unified approach, SIP shows the presence for the *user*, regardless of the device she happens to be using, rather than simply being “idle” or “out-to-lunch” based only on the presence status of a single application, such as IM.

Also, SIP-enabled devices are smarter and more aware of their owners’ *preferences,* including which modes to communicate in depending on a variety of conditions. Regardless of whether you’re in the office or on the road, available by phone or IM/chat, or prefer to receive still images on your data-enabled lower-speed mobile phone instead of streaming video, callers only need to know one “number” to reach you — your SIP AOR or logical URI.

REMEMBER

Presence refers to the ability for SIP-based communications to become smarter by facilitating communications (including voice, video, text, and Web) based on a user’s preferences and ability to communicate.

Presence-PBX Integration

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

Combining presence from multiple SIP devices informs the caller that the user is present or not, but the caller does not need to know on which number to call. He simply sends a message to `sip: coach@company.com`, and the SIP server starts the session with the right device (mobile phone, desk phone, IM, video-enabled endpoint) at the right time using the preferred mode of communication (voice, text, video, and so on).

Further Optimization

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

For example, 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 on the presence of the recipient). Coupling presence servers with application servers adds much more decision-making intelligence about where the message should be delivered, improving the speed at which users respond and communicate.

Not only can you convert voicemail messages to e-mail messages with speech-to-text tools, but with Avaya's DevConnect partner tools you also can convert e-mails to voice messages.

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 communications.

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. More tools to manage and aggregate presence information from multiple sources likely will emerge.

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 on a static set of business rules, they will adapt to changing conditions (such as volume and source of messages), user profiles, security settings, and software capabilities, to make changes to user preferences automatically.

Location-Based Services

Presence enables 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 emergency dialing and E911, among other things) to obtain and act on a user's approximate geographic location. Also, presence-enabled wireless access points and microcellsites 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 microcellsite, 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 if he would like to reserve it and for how long. If the room was already booked, an IM can be sent to the user informing him that the room is booked and by whom, and offering nearby alternate locations.

Ensuring Privacy

Privacy is 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 can 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 need to be "aware" of privacy laws and regulations in various parts of the world, and alter their behavior accordingly.

Chapter 7

Ten Reasons to Use SIP-Enabled Solutions by Avaya

In This Chapter

- Simplifying communications for your users . . . and your customers
- Improving customer service
- Imagining the possib"ilities" — flexibility, scalability, productivity and more!
- Doing more with less (money)

SIP is a key enabling protocol that can revolutionize your IP communications infrastructure. As more and more organizations deploy SIP-enabled intelligent communications architectures, they are realizing very real and immediate benefits. This chapter gives you a quick overview of some of SIP's most important benefits — and will help you explain those benefits to other decision-makers in your organization.

Easier Manageability

SIP networks are relatively easy to set up and administer, yet include many advanced features, such as automated attendant, voice mail, and three-party conferencing.

Because SIP is modeled after HTTP as a text-based language, it is easy to learn, develop, troubleshoot, and support. From analyzing network packets to developing custom applications, SIP's structured language makes it easier for IT systems engineers and developers to understand and interpret it. The ability to rapidly deploy new technologies and applications will give your business a definite competitive advantage.

As the industry continues to evolve, so will interoperability and manageability across third-party software, devices, and so on, and integrated management across the entire SIP environment will become even easier!

Turn to Chapters 1 and 2 to find out more.

User-Centric Communications

SIP works with the Extensible Messaging and Presence Protocol (XMPP), or *presence*, to intelligently connect communicating parties based on their ability and willingness to participate in a communications session, as well as their preference, based on parameters such as time of day, day of week, desired mode, and type of caller.

Unified addressing, through the use of a SIP AOR (*address of record*), provides a single URI (Uniform Resource Identifier) that can be used for routing all communications 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 eliminates the need for tracking users' multiple phone numbers, e-mail addresses, and IM contact names.

SIP also supports intelligent forking — that is, the ability to route a communications session to the right person, using the right medium (voice, video, text), on the right device (or application), and at the right time. We explain all of these key features in more detail in Chapters 2 and 3.

Native Mobility

SIP builds the foundation for native mobility in applications and devices for the (not-too-distant) future. 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 everyone.

For example, SIP's awareness of a user's communication capabilities will aid international travelers who have to use different mobile 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 will know how a person can be reached and will facilitate the connection. Roam to Chapter 3 to find out more.

Unlimited Scalability

No matter the size of your business or the state of your current telecommunications infrastructure, Avaya has a solution to fit your intelligent communications and your unified communications needs and budget.

SIP-enhanced communications solutions can easily scale from a small branch office deployment to carrier-class enterprise networks spanning multiple continents. Whether you're looking to consolidate a mix of legacy TDM PBX telecommunications equipment and evolve your IP communications network incrementally, or "rip and replace" it all at once, SIP can easily scale to any deployment scenario and support the future growth of your business. See Chapters 3 and 4 for more information about the flexibility and scalability of SIP communications architectures.

Better Survivability

The ability to communicate is important to any business, but real-time communications are critical to the survival of your business when disaster strikes. SIP-enhanced intelligent communications architectures help ensure that disaster recovery and business continuity plans actually work.

Through SIP, enterprises can control their redundant platforms and fail-over paths and, with the advent of both User-to-User Information (UUI) and Network Call Redirect (NCR), SIP trunks offer feature parity with transport mechanisms such as Integrated Services Digital Network (ISDN). SIP trunking and survivability are covered in Chapters 3 and 4.

Endpoint Flexibility

End users appreciate choice so they benefit from the fact that the SIP protocol works on a wide variety of communications devices. Users preferring a desk phone, a unified communications softphone on a PC, or a favorite mobile phone can connect to a SIP-based communications network seamlessly and easily. This means that the same communication functionality available in the office extends to a user via any device.

For example, your desk phone and mobile phone can ring simultaneously. This enables you to remain in contact with customers and business associates no matter where you are. SIP also gives you "mobility within the office," allowing you to sit down at any desk or in any office, log in, and automatically download your communications profile to your current location — making it appear as if you were sitting at your own desk! Call logs, conference calling, call transfer — and other features you use on your office phone are extended to your other communication devices and your current location. With SIP, you get endpoint flexibility without sacrificing endpoint functionality. Learn more about SIP endpoints in Chapter 4.

Unprecedented Interoperability

SIP is an open standard defined in RFC 3261 by the IETF, an international community of network designers, operators, vendors, and researchers, all concerned with the evolution of the Internet architecture and the development of standards to ensure the smooth operation of the Internet.

Several working groups, including SIPit, SIPconnect, SIP Foundry, and the SIP Forum — the board of directors of which includes Avaya — arrange regular 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. The SIPconnect Compliant designation helps customers identify solutions that provide interoperability among multiple vendors. Avaya is one of the first five vendors to be certified as SIPconnect Compliant.

SIP's ability to work across a range of systems helps enterprises enjoy more seamless integrations between platforms, devices, and applications, so that companies can rapidly deploy new technologies, applications, and services. See Chapter 4 for more information.

Lower Total Cost of Ownership

SIP-enhanced intelligent communications architectures deliver lower total cost of ownership (TCO) to businesses and enterprises through SIP trunks. 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. In fact, Avaya customers typically see a return on investment (ROI) of 6–12 months for their trunking solutions. If you're keen on cutting costs, you can find out more in Chapter 4.

Enhanced Customer Service

More than ever, customer service is a competitive differentiator for any successful business. Customers expect and deserve more than just a call center that routes them to the "next available agent" or an impersonal interactive voice response (IVR) system that routes them in circles until they get frustrated and take their business elsewhere.

SIP has accelerated the migration from call center to contact center, allowing businesses to truly leverage the power of real-time IP communications, including voice, IM, chat, Web, and video, to ensure that customer service agents and technical experts all have the features, functionality, and information necessary to best serve their customers.

Get more information about contact centers and customer service in Chapter 5.

Increased Productivity

It's easy to imagine how any of the features and advantages already described in this chapter could increase individual productivity just by simplifying your communications world. But SIP can actually do a lot more to improve productivity, increase efficiency, and save the whales! Well, that may be a stretch, but SIP can help make your office "greener," for example, by facilitating virtual meetings using multimedia collaboration tools (voice, video, and data through the Web) such as Avaya's Meeting Exchange, Avaya Web Conference, and Avaya one-X portal together, thereby reducing the need for costly travel. With rising fuel costs alone, many companies can expect an ROI of six months or less using SIP's conferencing/ video conferencing and collaboration capabilities.

By deploying SIP-enabled intelligent custom applications, businesses can offer new services for their customers and improve existing business processes. For example, we've already described the advantages of SIP in the contact center in terms of customer service, but what about its implications for employee productivity? With SIP-enabled applications, a customer service agent can be more productive by improving his response and resolution times. Rather than simply escalating calls to various technicians, sales reps, or supervisors, the agent can communicate with all the necessary parties over a number of SIP-enabled applications and platforms while simultaneously maintaining contact with the customer. This saves valuable time (and frustration) for everyone involved since the customer doesn't have to repeat the issue each time a call is "handed off" to someone else. You can read more about contact centers in Chapter 5.

REMEMBER

Here's a good way to add the "green factor." Meeting Exchange with Avaya Web conferencing and Avaya one-X Portal enhance employee productivity, lower cost of travel, and provide a multimedia (voice, video and data through Web) experience today. Or perhaps you simply want calls to automatically find the device you're using while working remotely to cut down on the commute. This is all possible today.