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About this document

Overview

This SIP Enablement Services (SES) Implementation Guide provides references to the documentation needed to deploy the Avaya SES solution with Avaya SIP IP telephony products. This document is a road map.

Intended audience

This document is for field technicians, services technicians, and installers who are deploying the Avaya SES solution over an IP telephony network.

Using this document

Use this document as a reference point for deploying the Avaya SES solution. The document is high-level, task-oriented and guides you through the documentation that contains installation, administration, and deployment instructions for all of the components of the Avaya SES solution.

Document tables

At the beginning of each chapter, related documents are listed in a table as follows:

<table>
<thead>
<tr>
<th>Document Title</th>
<th>DocumentID</th>
</tr>
</thead>
<tbody>
<tr>
<td>4600 Series IP Telephone LAN Administrator Guide</td>
<td>555-233-507</td>
</tr>
<tr>
<td>4610SW SIP Telephone User’s Guide</td>
<td>16-300472</td>
</tr>
<tr>
<td>4610SW SIP Telephone Quick Reference</td>
<td>16-300473</td>
</tr>
</tbody>
</table>
Reference points

A key component of this documentation is the reference point. The reference point refers to the document that has the information that is needed to complete the task at hand.

Reference point—Setting up SES server(s)

- For detailed instructions on the initial setup of Avaya SIP Enablement Services on a standalone S8500-series server, refer to this document:
  *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 3: “Setup and configuration”—”Configuring a new server”.

The sample reference point above refers to the Avaya document, the relevant chapter within the document, and the section heading within the chapter to complete the task.

Related sources

The following documents are referenced within this document:

- Avaya application notes
- Administration for Network Connectivity for Avaya Communication Manager, 555-233-504
- Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508
- Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768
- Avaya Toll Fraud and Security Handbook, 555-025-600
- Security and the Avaya S8700 Media Servers, Avaya white paper, Resource Library
- Avaya one-X™ Deskphone SIP for 9620 IP Telephone User Guide, 16-601945
- Avaya one-X™ Deskphone SIP for 9630/9630G IP Telephone User Guide, 16-601946
- Avaya one-X™ Deskphone SIP for 9640/9640G IP Telephone User Guide, 16-602403
- Avaya one-X™ Deskphone SIP for 9620 IP Telephone Quick Reference, 16-601947
- Avaya one-X™ Deskphone SIP for 9630/9630G IP Telephone Quick Reference, 16-601948
Conventions used in this document

Become familiar with the following terms and conventions.

- Media server—In the SES solution, Avaya media servers may be S8300, S8400, S8500 series, or S8710 series. When this document refers to a media server, it means a Linux-based media server running Avaya Communication Manager R3.1 or later.

- Commands are printed in bold face as follows: command.

  We show complete commands in this book, but you can usually type an abbreviated version of the command. For example, list configuration station can be typed as list config sta.
About this document

- Screen displays and names of fields are printed in constant width as follows: `screen display`.
  A screen is any form displayed on your computer or terminal monitor.
- Variables are printed in constant width, bold, italic: `variable`.
- Keys and buttons are printed as follows: `KEY`.
  To move to a certain field, you can use the `TAB` key, arrows, or the `ENTER` key (the `ENTER` key may appear as the `RETURN` key on your keyboard).
- In this book we use the terms “telephone” and “voice terminal” to refer to phones.
- Reference point—Defined
  The reference point refers to any related documentation that details the tasks at hand.
- Tips look like this:
  ❯ Tip:
  Draws attention to information that you may find helpful.
- Notes look like this:
  Note:
  A general note calls attention to neutral information or positive information that supplements the main text.
- Cautions look like this:
  ⚠ CAUTION:
  Denotes possible harm to software, possible loss of data, or possible service interruptions.
Chapter 1: SES Solution Overview

This chapter provides an overview of the process and components needed to deploy an Avaya IP convergence solution using Session Initiation Protocol (SIP) as its enabling technology. SIP is a text-based protocol that is designed to set up, modify, and tear down communication sessions between users. Once sessions are established, the content of these sessions can be voice, video, instant messaging, or any other communications method.

SIP, an Internet-centric protocol that provides basic functionality beyond that of H.323, was originally designed to place the intelligence in the endpoints rather than within the network. Practical necessity requires that some intelligence be in the network. SIP architecture promises increased resiliency, scalability, and rapid application development.

Presently, Avaya’s SIP Enablement Services (SES) solution offers these advantages:

- The ability to make and receive SIP telephone calls
- Many advanced features and services
- Secure instant messaging
- To subscribe to and receive presence notifications.

This document outlines how to deploy an SES solution.

Reference point—System architecture, server requirements, and local failover

- Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 3: "Introduction", provides details of the SES solution architecture and topography, server hardware and software requirements, and local failover design.

The tasks required to deploy this solution are described in this document, in these sections:

1. Set up the server(s) running Avaya Communication Manager.
   See Chapter 2: Setting up the Communication Manager server on page 19.

2. Install and configure Avaya SIP Enablement Services on the SES server(s).
   See Chapter 3: Setting up the server(s) running SES on page 25.

3. Integrate Communication Manager, SIP Enablement Services, and the 4600/9600 series SIP IP telephones via Extension to Cellular and Off-PBX Station (OPS) mapping.
   See Setting up OPS stations on page 28.

4. Establish SIP Trunks on the server running Communication Manager.
   See Chapter 4: Establishing SIP trunks in Communication Manager on page 29.

5. Configure supported SIP IP telephones for use with the SES solution.
   See Chapter 5: Setting up a SIP Endpoint on page 31.
Reference point—Co-resident system architecture and server requirements

- Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508, provides details of the SES solution architecture and topography, server hardware and software requirements, and local failover design.

Overview of Avaya SES solution components

The Avaya SIP Enablement Services (SES) solution consists of the following components:

- Avaya SIP Enablement Services server or servers
- One (or more) Servers running Avaya Communication Manager 5.0
- Gateways
- SIP endpoints:
  - Avaya one-X™ Deskphone supporting SIP for 9600 Series phones
  - Avaya 4600 Series IP Telephones supporting SIP
  - Avaya softphones, including one-X™ Desktop Edition
- Third-party SIP clients

Avaya SIP Enablement Services server

The SES application runs standalone on Avaya’s S8500-series hardware platform. This platform supports Edge, Home, and combination Home/Edge servers. The SES application also supports the Avaya S8300C Server, running co-resident with Avaya Communication Manager 5.0 or later releases. In this configuration, only Home and Home/Edge SES server types are supported.

The SES edge server is dedicated to performing proxy, registration, and redirection functions associated with SIP applications, such as instant messaging. When the SES home server communicates with one or more servers running Communication Manager, communication between and among the various non-SIP endpoints is supported. Endpoints supported by Communication Manager include analog, DCP or H.323 stations and analog, digital or IP trunks, as well as SIP-enabled endpoints. SIP-enabled endpoints may be any from this list:

- Avaya 9600 Series one-X™ Deskphone, Avaya 4600 Series SIP IP telephones or Avaya Agent Deskphone 16CC
- Avaya one-X™ Desktop Edition Release 2 or later (or Avaya SIP Softphone)
- Avaya IP Softphone Release 5.1 and later with the IM client
SIP-enabled endpoints must register with the SES server. Although technically an Avaya IP Softphone is not managed by Communication Manager, it is required to register H.323 with the server running Avaya Communication Manager. Although IP softphones use H.323 for voice, SIP is also used for communications. For example, instant messages on IP softphones use SIP.

Reference point—System requirements, installation and administration for standalone SES servers

The document Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, provides detailed system requirements, installation instructions, and administrative tasks for S8500 series servers running SES. See the respective chapters.

- System requirements - Chapter 3: "Introduction to SIP and SIP Enablement Services"
- Installation - Chapter 4: "Setup and configuration"
  Chapter 5: "Getting ready to install SES"
  One or more of chapters 7-21, depending on your hardware configuration
- Administration - Chapter 4: "Setup and configuration"—"Installation checklist"
  Chapter 4: "Setup and configuration"—"Post installation tasks"
- Administration in general - Chapter 22: "Administration web interface"

Servers running Avaya Communication Manager 5.0

Avaya’s SES solution can be deployed with Communication Manager running on Avaya S8300, S8400, S8500-series, and the S8700-series servers. These hardware elements deliver application-enabling data, voice, fax, video and messaging capabilities to your network. They support both bearer and signaling traffic routed between packet-switched and circuit-switched networks. The gateways are optimized for use with enterprise class telephony and provide a variety of flexible deployment options. Options include both partial environments, such as IP and TDM, and fully IP environments.

Features for SIP in Avaya Communication Manager 5.0 running on an Avaya media server include these:

- SIP trunking
- SIP stations
- Call Detail Record (CDR) support
- Access control
- Routing and dial plan
Gateways

The Avaya SES solution can be deployed with any of the Avaya gateways supporting S8XXX. Note that if the Avaya G250 Gateway is used, it may limit the normal capacity of 100 SIP users on Avaya SES 5.0 running on an Avaya S8300C Server co-resident with Communication Manager 5.0 or later. No other Avaya gateway servers limit SIP-user capacity of SES.

Avaya one-X™ Deskphone supporting SIP for 9600 Series phones

Verify that you have up-to-date firmware for your supported SIP endpoints by going to this web site:

http://www.avaya.com/support/

Then select the link for Downloads under Most Visited Support Areas. See the latest 9600 Series one-X™ Deskphone SIP telephone documentation for more details.

Key benefits of an Avaya 9600 Series one-X™ Deskphone SIP telephone

- The 9600 Series SIP telephone is a fully featured SIP device also utilizing the features of Communication Manager through SIP Enablement Services.
- This telephone employs standards-based SIP telephony call control, allowing integration into multi-vendor telephony environments using SIP.
- The simple H.323 to SIP firmware upgrade provides a migration path to SIP that maximizes investment protection.

Note:
The 9600 Series SIP telephone incorporates a SIP stack. The telephone registers with a server running SIP Enablement Services (SES) software and communicates through the server with Communication Manager for call control.

See Other scenarios for using an SES system on page 30 for more information on multi-vendor SIP telephony deployments.

Converting Avaya 9600 Series telephones to one-X™ Deskphone SIP

SIP can be deployed on the Avaya 9620, 9630/9630G, 9640/9640G telephones. Avaya ships these phones from the factory and they contain the latest load of H.323 software. To use an H.323 phone with SES, you must convert the H.323 phone to a SIP phone.

Reference Point—Enhancing 9600 Series H.323 IP telephones to SIP telephones

- For detailed instructions on enhancing the software on the Avaya 9600 Series IP telephones to one-X™ Deskphone SIP, refer to the document Avaya one-X™ Deskphone
Overview of Avaya SES solution components

Reference point—System requirements, installation and administration for 9600 Series SIP IP telephones

- A separate Avaya one-X™ Deskphone for 9600 Series SIP IP Telephone Quick Reference for each of the 9620, 9630/9630G, and 9640/9640G IP Telephones provides installation instructions and initial configuration tasks.
- A separate Avaya one-X™ Deskphone SIP for 9600 Series SIP IP Telephones R2.x User Guide for each of the 9620, 9630/9630G, and 9640/9640G IP Telephones provides detailed tasks associated with using the phone.

Avaya 4600 Series IP Telephones supporting SIP

Verify that you have up-to-date firmware for your supported SIP endpoints by going to this web site:

http://www.avaya.com/support/

Then select the link for Downloads under Most Visited Support Areas. For the 4602, you may need to migrate your H.323 firmware from 1.8.x to 2.2 before converting the phone to SIP firmware. See the latest 4600 Series IP telephone documentation for more details.

Key benefits of an Avaya 4600 Series SIP telephone

- The 4600 Series SIP telephone is economical and can utilize the features of Communication Manager through SIP Enablement Services.
- This telephone employs standards-based SIP telephony call control and so allows integration into multi-vendor telephony environments using SIP.
- The simple H.323-to-SIP firmware upgrade provides a migration path to SIP that maximizes investment protection.

Note:

The 4600 Series SIP telephone incorporates a SIP stack. The telephone registers with a server running SIP Enablement Services (SES) software and communicates through the server with Communication Manager for call control.
See Other scenarios for using an SES system on page 30 for more information on multi-vendor SIP telephony deployments.

**Avaya 4602/4602SW, 4610SW or 4620SW/4621SW telephones**

SIP can be deployed on the Avaya 4602/4602SW, 4610SW or 4620SW/4621SW telephones. Avaya ships these phones from the factory and they contain the latest load of H.323 software. To use an H.323 phone with SES, the administrator must convert the H.323 phone to a SIP phone.

**Reference Point—Enhancing 4600 Series H.323 IP telephones to SIP telephones**

- For detailed instructions on enhancing the software on the Avaya 4600 Series SIP IP telephones, refer to the document *4600 Series IP Telephone R2.2 Installation Guide*, 555-233-128, Chapter 2: “4600 Series IP Telephone Installation”—“Converting Software on Avaya 4600 Series IP Telephones”.

**Reference point—System requirements, installation and administration for 4602 SIP telephones**

- The *Avaya 4602/4602SW SIP Telephone User’s Guide*, 16-300035 provides detailed tasks associated with using the 4602 or 4602SW SIP telephone.
- The *Avaya 4602/4602SW SIP Telephone Quick Reference*, 16-300471 provides installation instructions and initial configuration tasks for the 4602/4602SW SIP telephone.

**Reference point—System requirements, installation and administration for 4610SW SIP telephones**

- The *Avaya 4610SW SIP IP Telephone User’s Guide*, 16-300472 provides detailed tasks associated with using the 4610SW SIP Telephone.
- The *Avaya 4610SW SIP IP Telephone Quick Reference*, 16-300473 provides installation instructions and initial configuration tasks for the 4610SW SIP.

**Reference point—System requirements, installation and administration for 4620SW/4621SW SIP telephones**

- The *Avaya 4620SW/4621SW SIP IP Telephone User’s Guide*, 16-300474 provides detailed tasks associated with using the 4620SW or 4621SW SIP telephone.
- The *Avaya 4620SW/4621SW SIP IP Telephone Quick Reference*, 16-300475 provides installation and initial configuration tasks for the 4620SW or 4621SW SIP telephone.
Avaya softphones

Avaya IP Agent R6.x and IP Softphone R5.x with IM client

SIP can be deployed with either the Avaya IP or SIP softphone clients. To deploy the SES solution, you must add the IM client module software to the IP softphone or the IP Agent software. The SIP softphone client software is SIP-enabled with IM capability.

Reference Point—Enhancing H.323 IP softphones to IM-enabled softphones
For detailed instructions on enabling the Avaya IP Softphone client software to include the SIP-enabled instant messaging with presence application, refer to IP softphone online help:
http://www.support.avaya.com > IP Agent > Avaya IP Agent Installation and User Guide

Avaya one-X™ Desktop Edition and SIP Softphone R2.x

SIP can be deployed with either the Avaya IP Softphone 5.x or one-X™ Desktop Edition SIP softphone 2.x clients. To deploy the one-X™ Desktop Edition SIP Softphone solution, the client software must be installed on the SIP user’s PC by an administrator of that PC.
The deskphone must have the current version of SIP firmware for the Avaya telephone.

Reference Point—Installing and configuring the Avaya SIP Softphone
For detailed instructions on installing the Avaya one-X™ Desktop Edition SIP Softphone client software, refer to the SIP softphone application’s online help:
http://www.support.avaya.com > SIP Softphone > Avaya SIP Softphone Administration
Third-party SIP clients

The Toshiba SIP Business Phone, model SP-1020A, is fully supported within the Avaya SES solution. See the documentation for your TSP phone for more information.

Avaya supports and troubleshoots other third-party SIP clients that have been chosen and completed testing by Avaya's Solution and Interoperability Test Lab (SITL), and is documented in Application Notes. However, Avaya does not provide installation, implementation, configuration, or maintenance of any third-party SIP endpoints.

The SIP-enabled products of third parties must be submitted through Avaya’s DevConnect to be tested and ultimately designated as supported applications/devices.

For detailed information about the Avaya DeveloperConnection Program, go to the web site: http://www.avaya.com/gcm/master-usa/en-us/corporate/alliances/devconnect/index.htm

For application notes on which third-party SIP endpoints are supported, go to the web site: http://www.avaya.com and select Do Your Research: Resource Library in the middle of the page.

Then select Application Notes under Resource Type. You may filter your results by the term SIP, or by the specific product name or model of your third-party software or equipment.
Security

Because SES is an Avaya product, and because it operates in conjunction with Avaya Communication Manager, SES is designed and implemented with security as a prime consideration.

Security in general

Avaya provides two documents about security that directly affect SES.

Table 2: Documents that relate to security

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya Toll Fraud and Security Handbook</td>
<td>555-025-600</td>
</tr>
<tr>
<td>Security and the Avaya Communication Manager Media Servers</td>
<td>Technical Support white paper for Avaya Communication Manager software</td>
</tr>
</tbody>
</table>

Security and SES

These points are part of the SES security strategy and design:

- SES administration is done securely over HTTPS, a secure extension to HTTP that encrypts all messages between the web server and a browser. Unique server certificates can be generated and installed on the SES server.
- The SES firewall is configurable to permit only the necessary network services and applications.
- SES runs an intrusion detection system to prevent unauthorized modification of files. Passwords are stored securely.
- SES supports SSH for secure remote login. User logins are denied after successive failed logins. In addition, ASG is used for secure technician login.
- SNMPv3 trap destinations can be configured for secure SNMP.
- SIP signaling can be protected using TLS, depending on endpoint support for TLS. SIP signaling to and from Communication Manager is protected using TLS.
- Telephony end users are authenticated using SIP digest authentication.
Chapter 2: Setting up the Communication Manager server

When setting up one or more server(s) running Communication Manager to accommodate and support the Avaya SES solution, you will need to have these references available:

Table 3: Documents required for setting up Communication Manager servers

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Administration for Network Connectivity for Avaya Communication Manager, 555-233-504</td>
<td>555-233-504</td>
</tr>
<tr>
<td>Getting Started with Avaya SIP Softphone, Administration and Provisioning</td>
<td>Available from your Avaya representative</td>
</tr>
<tr>
<td>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768</td>
<td>03-600768</td>
</tr>
<tr>
<td>Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508</td>
<td>03-602508</td>
</tr>
<tr>
<td>SIP Support in Avaya Communication Manager, 555-245-206</td>
<td>555-245-206</td>
</tr>
</tbody>
</table>

Avaya Communication Manager runs on the following servers:
- Avaya S8300 Server, with or without Avaya SES running co-resident with it
- Avaya S8400 Server
- Avaya S8500-series Servers
- Avaya S8700-series Servers.

Note:
Prior to enabling SIP/SES on the server(s) running Communication Manager, IP connectivity must be configured correctly. For details, refer to Administration for Network Connectivity for Avaya Communication Manager, 555-233-504.

Setting up the Avaya server

As noted above, prior to setting up an Avaya server for SIP, you must configure IP connectivity correctly. After IP connectivity has been configured, complete the following tasks:
Setting up the Communication Manager server

- Setting up SIP support in Avaya Communication Manager on page 20
- Setting up the server(s) running SES on page 25
- If applicable for your SIP clients, Setting up Click to Conference on page 22
- If applicable for your SIP endpoints, Setting up Advanced SIP Telephony on page 22
- And, if applicable, Setting up Modular Messaging as an adjunct system on page 23.

Reference point—List of related documents

- For information on correctly configuring IP connectivity on your media server, refer to Administration for Network Connectivity for Avaya Communication Manager, 555-233-504. Many sections in this document discuss this topic.
- For information on setting up standalone Avaya SES server(s), refer to Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768:
  - Chapter 4: "Setup and configuration"—"Pre installation checklist", and "Post installation tasks"
  - One or more of chapters 7-22: "Installation procedures"—"Administer Communication Manager and endpoints"
- For information on setting up co-resident Avaya S8300C server(s) for SES, refer to Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508.
- For information on setting up Avaya Communication Manager, refer to SIP Support in Avaya Communication Manager, 555-245-206, Chapter 2: "Administering SIP in Avaya Communication Manager".

Setting up SIP support in Avaya Communication Manager

Avaya’s SES solution can be deployed with the S8300, S8400, S8500-series, and the S8700-series servers running Communication Manager. These hardware elements deliver application-enabling data, voice, fax, video and messaging capabilities to your network. They support both bearer and signaling traffic routed between packet-switched and circuit-switched networks. The gateways are optimized for use with enterprise class telephony and to provide a variety of flexible deployment options. Options include partial environments, such as both IP and TDM, as well as full IP environments with H.323 and SIP endpoints.

With Avaya Communication Manager, SIP features include these:

- SIP trunking
- SIP stations
- Call Detail Recording (CDR)
- Profile-based access control
- Call routing
Enterprise dial plan.

**SIP trunking**

Support for SIP trunks allows an enterprise to connect its Avaya Communication Manager servers to a SIP-enabled proxy server running Avaya SIP Enablement Services (SES), and through that proxy to an external SIP service provider, if desired. The trunk support in Avaya Communication Manager complies with SIP standards IETF RFC 3261 and interoperates with any SIP-enabled endpoint that also complies with the standard.

In complex configurations with Avaya S8700-series servers, SIP signaling-group properties in Communication Manager must be administered to match each other in certain, specific ways.

**SIP stations**

Support for SIP stations using SIP trunks allows any fully compliant SIP phone to interoperate with Avaya phones. This means any SIP phone, from Avaya or a third party, that complies with the appropriate RFC or draft service standards can do the following:

- Dial and be dialed as an extension in the enterprise dial plan. Outboard Proxy SIP (OPS) stations support additional features as well, like bridging.
- Put calls on hold and participate in transfers and conference calls.

**Reference point—OPS**

- For more details on OPS installation, refer to these documents:
  - *Avaya Extension to Cellular User’s Guide*, 210-100-700
  - *Avaya Extension to Cellular and OPS Installation and Administration Guide*, 210-100-500

**Reference point—Modular Messaging**

- For using Modular Messaging as an adjunct system to SES, see the applicable SES installation and administration guide: *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, 03-600768, Chapter 22: "Administration Web Interface"—"Adjunct Systems", or see the document *Administering SIP Enablement Services on the Avaya S8300 Server*, 03-602508.

**Access control**

Support is provided for the full control of access to external trunks from any phone. Note that some other means of access control, such as a network firewall, typically would be required to control access to SIP trunks from SIP endpoints that are outside of the enterprise.
Setting up the Communication Manager server

Reference point—Setting up Communication Manager servers

All necessary information for setting up the servers with Avaya Communication Manager is detailed in *SIP Support in Avaya Communication Manager*, 555-245-206. Specific topics in this document are as follows:

- For hardware and software requirements, refer to Chapter 2: "Administering SIP in Avaya Communication Manager"—"Requirements for SIP".
- For setting up and configuring SIP trunks in Communication Manager, refer to Chapter 3: "Administering Communication Manager for SIP Enablement Services".

Routing and Dial Plan

The final step before you make SIP calls from endpoints connected to Communication Manager is to administer call routing properly on the Avaya CM servers. This involves several screens, including for example the Locations, Numbering, and Route Pattern screens.

Reference point—Administering SIP call routing in Communication Manager

Information for administering call routing on the media servers with Avaya Communication Manager is detailed in *SIP Support in Avaya Communication Manager*, 555-245-206. Refer to Chapter 3: "Administering Communication Manager for SIP Enablement Services".

Setting up Click to Conference

For information on how to administer the click-to-conference feature of SES, obtain this Avaya document:

*Getting Started with Avaya SIP Softphone, Administration And Provisioning*, available from your Avaya representative.

Setting up Advanced SIP Telephony

For application notes on how to set up SES for use with Advanced SIP Telephony, go to Avaya’s web site:

http://www.avaya.com and select Do Your Research: Resource Library in the middle of the page.

Then select the link for Application Notes under Resource Type. Filter the results for AST, or the specific product supporting advanced SIP telephony in which you are interested.
Setting up Modular Messaging as an adjunct system

For administration guidance on how to implement Avaya’s Modular Messaging product as an adjunct system in an SES solution, refer to the following resources:

* Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 22: "Administration Web Interface”—"Adjunct Systems".

* Configuration Note 88010—This note will be published on the Avaya support web site. Search the Configuration Notes Master Index, or for PSN #1186U on SIP and MM Adjunct integration.
Setting up the Communication Manager server
Chapter 3: Setting up the server(s) running SES

When working with the server(s) with SES installed, you might use these references:

Table 4: Documents required for setting up an Avaya SES server

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768</td>
<td>03-600768</td>
</tr>
<tr>
<td>Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508</td>
<td>03-602508</td>
</tr>
<tr>
<td>Avaya Extension to Cellular and OPS Installation and Administration Guide</td>
<td>210-100-500</td>
</tr>
</tbody>
</table>

Configuring the server running SES

All servers running SIP Enablement Services, that is, the SIP proxy servers, must be properly connected to, and configured on, an enterprise's IP network. Appendix C: Worksheet for installation on page 41 provides a complete list of the information you need to answer questions when prompted by the install script.

**Note for SES X.0 to SES X.1 upgrades:** - If you are upgrading from SES R3.0 to SES R3.1, or R4.0 to 4.0.1, use the three Server Upgrade screens in the maintenance interface of SES. These screens may not be used to upgrade and migrate directly from SES R3.x to R5.0 or later.

Trunking

For endpoint clients on Avaya Communication Manager to interoperate through the Avaya SES server, SIP trunking must be administered properly on the server running the Communication Manager software. Refer to SIP Support in Avaya Communication Manager, 555-245-206, Chapter 3: "Administering Communication Manager for SIP Enablement Services" for more details. This administration requires that you specify several values, including:

- Network names of the SES hosts on the **IP Node Names** screen
- Authoritative domain assigned to server(s) on the **IP Network Region** screen.

Additional administration of call routing may be required, depending on current server values.
Setting up the server(s) running SES

Note:
For 4600 Series and 9600 Series SIP IP Telephones, the same authoritative domain must be specified in the phone’s settings file(s) as on the IP Network Region screen. Also, unlike the H.323 versions of these IP telephones, the SIP versions must have working IP network access to their respective TFTP or HTTP servers upon startup. SES access is required for phone registration.

Task list for configuring server(s) for SES

Table 5 lists the tasks necessary to configure a standalone server for SES. For greater detail about each task, refer to the documents that correspond with each task listed in the table.

Note:
For greater detail about the tasks necessary to configure a co-resident server running Communication Manager to run SES as well, refer to Administering SIP Enablement Services on the Avaya S8300 Server, 03-602508.

Table 5: Configuration task list for standalone server(s) running SES

<table>
<thead>
<tr>
<th>✔</th>
<th>Task</th>
<th>Reference document</th>
</tr>
</thead>
</table>
| | Perform initial assembly and setup. | Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768  
  - Chapter 4: "Setup and configuration”—"Configuring a new server"  
  - Chapter 4: "Setup and Configuration”—"Pre-installation tasks" |
| | Install all server software. | Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768  
  - Chapters 7-21: "Installation Procedures" |
| | Perform administrative tasks on the server by way of a Point to Point Protocol (PPP) session. | Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768  
  - Chapter 4: "Setup and configuration”—"Installation checklist"  
  - Chapter 4: "Setup and configuration”—"Post installation checklist" |

1 of 2
Creating SIP users in the SES system

To create users in the SES system, you must access the web-based user interface and perform the necessary tasks in the following reference point.

Reference point—Creating users via the user screens in SES

- For creating a new SIP user in your SES solution, refer to Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 5: "Administration web interface"—"User Screens"—"Add User screen".

- For server administration of users, refer to Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 5: "Administration web interface"—"User screens"—"Extension Task"—"List Media Server Extensions for this user".

For loading users in bulk with ProVision, refer to Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768, Chapter 5: "Administration web interface"—"Export/Import with ProVision"—"Import".

Also refer to ProVision’s help subsystem and also to the Notepad newsletter, available from the ProVision web site's Document Center. Log in to ProVision at this web site:

http://www.avaya.com/support/iProVision/
Setting up OPS stations

Outboard Proxy SIP (OPS)

Beginning with Avaya Communication Manager Release 2.1, the Outboard Proxy SIP (OPS) application type is used to administer a SIP phone. OPS cannot be disabled using the Extension to Cellular enable/disable feature button.

Note:
Any SIP phone or endpoint must have access to and register with the SIP proxy, regardless of whether OPS is administered in Communication Manager.

The Extension to Cellular/OPS application allows for many of the parameters used for the original Extension to Cellular application to be ported onto one of several DCP and IP station types. From a call processing perspective, Extension to Cellular/OPS is in fact dealing with a multi-function phone, whereas the previous Extension to Cellular implementation utilized one or two XMOBILE stations that behaved like analog station types.

Extension to Cellular/OPS supports these features:

● Support for Session Initiation Protocol (SIP) phones
● Administrative ability to map certain Communication Manager features to phone extensions

Reference Point—Configuring Extension to Cellular and OPS for SIP

● Configuring Extension to Cellular and OPS for SIP is detailed in the Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide, 210-100-500.

Table 6: Documents required for setting up Extension to Cellular and OPS

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya Extension to Cellular and OPS Installation and Administration Guide</td>
<td>210-100-500</td>
</tr>
</tbody>
</table>
Chapter 4: Establishing SIP trunks in Communication Manager

When you set up the SIP trunks on the server running Communication Manager, use these references to help you:

Table 7: Documents required for establishing SIP trunks in Communication Manager

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Support in Avaya Communication Manager, 555-245-206</td>
<td>555-245-206</td>
</tr>
<tr>
<td>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services, 03-600768</td>
<td>03-600768</td>
</tr>
</tbody>
</table>

Session Initiated Protocol (SIP) is an endpoint-oriented messaging standard defined by the Internet Engineering Task Force (IETF). SIP trunking enables the Avaya servers running Communication Manager to function as a POTS gateway and to support the delivery of name and number data between legacy endpoints (for example, DCP telephones) and SIP-enabled endpoints (for example, the new 9600 series IP telephones with one-X™ Deskphone SIP).

For more information on SIP administration and usage, see the overview and introduction chapters of the documents listed in the preceding table.

Advanced SIP trunk administration

Avaya Communication Manager provides the capability to administer IP trunks as either H.323 or SIP. These trunks are administered as a trunk type and associated with the appropriate type of signaling group on the server running Communication Manager. Then, the server(s) running SES enable(s) these SIP trunks to be connected to SIP endpoints.
Establishing SIP trunks in Communication Manager

Reference Point—SIP Trunk Administration

- To administer SIP trunks in Avaya Communication Manager, see SIP Support in Avaya Communication Manager, 555-245-206, Chapter 3: "Administering Communication Manager for SIP"—"Administer SIP signaling and trunks". Capacities are as follows:

<table>
<thead>
<tr>
<th>Media Server</th>
<th>Maximum Administered SIP Trunks</th>
</tr>
</thead>
<tbody>
<tr>
<td>S8700 series</td>
<td>5000</td>
</tr>
<tr>
<td>S8500 series</td>
<td>800</td>
</tr>
<tr>
<td>S8400</td>
<td>400</td>
</tr>
<tr>
<td>S8300</td>
<td>450</td>
</tr>
</tbody>
</table>

Note: There is a maximum of 255 SIP trunks in a SIP signaling group.

Other scenarios for using an SES system

There are other scenarios in which SIP trunking can link an enterprise’s Avaya IP telephony network with a third-party SIP service provider’s network. The service provider may employ different proxy servers, PSTN gateways and other equipment as part of these scenarios.

Avaya’s Solution and Interoperability Test Lab (SITL) tests other scenarios, and then documents them in Application Notes. However, Avaya does not provide installation, implementation, configuration, or maintenance of any third-party SIP IP equipment.

The SIP-enabled products of third parties must be submitted through the Avaya DevConnect program in order to be tested and ultimately designated as supported applications or devices.

For detailed information about the Avaya DeveloperConnection Program, go to the web site:

For application notes on which third-party SIP endpoints are supported, go to the web site: http://www.avaya.com and select Do Your Research: Resource Library in the middle of the page.

Then select the link for Application Notes under Resource Type. You may filter results using SIP or the specific product name or model of third-party equipment in which you are interested.
Chapter 5: Setting up a SIP Endpoint

When you configure the endpoints for your SES system, use document tables Table 11 and Table 12 in this section to guide you.

Compatibility Matrix

The following table summarizes the minimum version of SIP-enabled firmware required for each of the supported Avaya telephones to operate properly with SES Release 5.0 software.

Table 9: Firmware versions required for using Avaya 9600 Series and 4600 Series SIP IP Telephones

<table>
<thead>
<tr>
<th>Telephone</th>
<th>Min. SIP Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>9620 IP telephone, one-X™ Deskphone SIP</td>
<td>Release 2.0.x</td>
</tr>
<tr>
<td>9630/9630G IP telephone, one-X™ Deskphone SIP</td>
<td>Release 2.0.x</td>
</tr>
<tr>
<td>9640/9640G IP telephone, one-X™ Deskphone SIP</td>
<td>Release 2.0.x</td>
</tr>
<tr>
<td>4602 IP telephone</td>
<td>SIP not supported</td>
</tr>
<tr>
<td>4602/4602SW SIP IP telephone</td>
<td>Release 2.2</td>
</tr>
<tr>
<td>4610SW SIP IP telephone</td>
<td>Release 2.2</td>
</tr>
<tr>
<td>4620SW/4621SW SIP IP telephone</td>
<td>Release 2.2</td>
</tr>
</tbody>
</table>

Confer with your Avaya representative and check the software downloads section of the Avaya Support web site to see if compatible versions other than the ones listed here are released.
Avaya one-X™ Deskphone SIP for 9600 series telephones

Table 10: Documents required for setting up the Avaya 9600 Series SIP Telephones

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones R2.x Administrator Guide</td>
<td>16-601944</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones R2.x Installation and Maintenance Guide</td>
<td>16-601943</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9620 IP Telephone Quick Reference</td>
<td>16-601947</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9620 IP Telephone User Guide</td>
<td>16-601945</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9630/9630G IP Telephone Quick Reference</td>
<td>16-601948</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9630/9630G IP Telephone User Guide</td>
<td>16-601946</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9640/9640G IP Telephone Quick Reference</td>
<td>16-602408</td>
</tr>
<tr>
<td>Avaya one-X™ Deskphone SIP for 9640/9640G IP Telephone User Guide</td>
<td>16-602403</td>
</tr>
</tbody>
</table>

The 9600 Series SIP IP telephones cannot boot and then register with the SES server and in the appropriate SIP domain unless they have network access to the TFTP or HTTP server.

If an SES solution includes 9600 Series IP telephones, the IP phones must be converted from H.323 protocol to SIP. Then, their station records/extensions must be administered both in Avaya Communication Manager and as SIP users (with the associated media server extension) in SES software. Finally, after parameters for the phones have been set properly, SIP calls may be placed from the phones.

Reference Point—Enhancing 9600 Series H.323 IP telephones to SIP telephones

- For detailed instructions on enhancing the software on the Avaya 9600 Series IP telephones to one-X™ Deskphone SIP, refer to the document Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones R2.x Installation and Maintenance Guide, 16-601943, Chapter 2: "9600 Series SIP IP Telephone Installation"—"Converting Software on 9600 Series IP Telephones".
Setting parameters

The 9600 Series SIP telephone has these basic tools and capabilities to assist administrators with managing its settings and features:

- The 96XXsettings.txt (or 46XXsettings.txt) file, annotated with information for setting most parameters
- DHCP for setting some parameters
- Downloadable configuration files for setting telephone parameters on startup
- Manual programming of critical parameters from the telephone’s dial pad
- Downloadable firmware updates (manual and automatic)

You can set a common set of parameters using DHCP, manual programming, and configuration files. Setting these parameters establishes the telephone’s operating parameters.

Avaya SIP telephones will not be able to register in an SES system unless they can obtain the correct SIPDOMAIN setting. Avaya SIP telephones include the 4600-series, the 9600-series, and the 16CC. The 46XX phones obtain the domain setting from the 46XXsettings.txt file; you may set up a separate 96XXsettings.txt file for the 96XX phones, but the SIPDOMAIN setting must be the same for your enterprise.

Always configure the SIPDOMAIN setting for the phones in the appropriate text file(s) and then ensure that the phones transfer settings from the file(s) via TFTP or HTTP during boot up.

The line in the file(s) for this setting is:

    SET SIPDOMAIN = yourSIPdomainName.com

For application notes on configuring specific SIP endpoints, go to the web site:

http://www.avaya.com and select Do Your Research: Resource Library in the middle of the page.

Then select the link for Application Notes under Resource Type. You may filter the results using the term SIP or the specific model number of your Avaya telephone.
Avaya 4602/4602SW, 4610SW and 4620SW/4621SW SIP telephones

Table 11: Documents required for setting up the Avaya 4600 Series SIP Telephones

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>4600 Series IP Telephone LAN Administrator Guide</td>
<td>555-233-507</td>
</tr>
<tr>
<td>4600 Series IP Telephone Installation Guide</td>
<td>555-233-128</td>
</tr>
<tr>
<td>4602/4602SW SIP Telephone Quick Reference</td>
<td>16-300471</td>
</tr>
<tr>
<td>4602/4602SW SIP Telephone User’s Guide</td>
<td>16-300035</td>
</tr>
<tr>
<td>4610SW SIP IP Telephone Quick Reference</td>
<td>16-300473</td>
</tr>
<tr>
<td>4610SW SIP IP Telephone User Guide</td>
<td>16-300472</td>
</tr>
<tr>
<td>4620SW/4621SW SIP IP Telephone Quick Reference</td>
<td>16-300475</td>
</tr>
<tr>
<td>4620SW/4621SW SIP IP Telephone User Guide</td>
<td>16-300474</td>
</tr>
</tbody>
</table>

The 4600 Series SIP IP telephones cannot boot and then register with the SES server and in the appropriate SIP domain unless they have network access to the TFTP or HTTP server.

If an SES solution includes 4600 Series IP telephones, the IP phones must be converted from H.323 protocol to SIP. Then their station records must be administered as OPS in Communication Manager and as SIP users with a media server extension in SES software. Finally, after parameters for the phones have been set properly, SIP calls may be placed from the phones.

Reference Point—Enhancing 4600 Series H.323 IP telephones to SIP telephones

- For detailed instructions on enabling the software on the 4600 Series IP Telephones, refer to 4600 Series IP Telephone Installation Guide, 555-233-128, Chapter 2: "4600 Series IP Telephone Installation,"—"Converting Software on Avaya 4600 Series IP Telephones".
Setting parameters

The 4600 Series SIP telephone has these basic tools and capabilities to assist administrators with managing its settings and features:

- The 46XXsettings.txt file, annotated with information for setting most parameters
- DHCP for setting some parameters
- Downloadable configuration files for setting telephone parameters on startup
- Manual programming of critical parameters from the telephone’s dial pad
- Downloadable firmware updates (manual and automatic)

You can set a common set of parameters using DHCP, manual programming, and configuration files. Setting these parameters establishes the telephone’s operating parameters.

For application notes on configuring SIP endpoints, go to the web site:

http://www.avaya.com and select Do Your Research: Resource Library in the middle of the page.

Then select the link for Application Notes under Resource Type. You may filter the results using the term SIP or the specific model number of your Avaya telephone.
Setting up Avaya softphone clients

Table 12: Documents required for setting up the Avaya softphones

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>one-X™ Desktop Edition R2.x Quick Setup Guide</td>
<td><a href="http://www.support.avaya.com">www.support.avaya.com</a> SIP Support area</td>
</tr>
<tr>
<td>one-X™ Desktop Edition R2.x Getting Started</td>
<td><a href="http://www.support.avaya.com">www.support.avaya.com</a> SIP Support area</td>
</tr>
<tr>
<td>Avaya IP Agent R6.x/IP Softphone R5.x (or later) with IM client</td>
<td>online help</td>
</tr>
<tr>
<td>Avaya SIP Softphone Release 2</td>
<td>online help</td>
</tr>
</tbody>
</table>

If an SES solution includes Avaya IP Softphone clients, the version of IP Agent or IP Softphone including the SIP-enabled instant messaging and presence module must be used. Note that these clients manage presence information in a peer-to-peer fashion.

You must uninstall SIP softphone clients before installing one-X™ Desktop Edition Release 2.

Reference Point—Enhancing H.323 IP softphones to IM-enabled softphones

For detailed instructions on how to include and enable the SIP-based instant messaging (with presence) application in Avaya softphone client software, refer to IP softphone online help. http://www.avaya.com/support > IP Agent > Avaya IP Agent Installation and User Guide


If an SES solution includes Avaya one-X™ Desktop Edition or SIP softphone clients, then the software must be installed and configured to use. Note that these clients manage their presence information by means of the SIP Personal Information Manager running on an SES server.

Reference Point—Installing and configuring the Avaya one-X™ Desktop Edition

For detailed instructions on installing one-X™ Desktop Edition or uninstalling the Avaya SIP softphone client software, refer to SIP softphone online help, or one of these locations:

http://www.avaya.com/support/ > Find Documentation > Avaya one-X™ Desktop Edition

http://www.avaya.com/support > SIP Softphone > Avaya SIP Softphone Administration

Appendix A: Reference Points

This appendix lists all of the reference points contained within this document. This appendix can serve as a quick reference for those who need to refer quickly to a document for a specific task.

About this document

- Reference point—Setting up SES server(s) on page 6
- Reference point—Defined on page 8

Chapter 1: SES Solution Overview

- Reference point—System architecture, server requirements, and local failover on page 9
- Reference point—Co-resident system architecture and server requirements on page 10
- Reference point—System requirements, installation and administration for standalone SES servers on page 11
- Reference Point—Enhancing 9600 Series H.323 IP telephones to SIP telephones on page 12
- Reference point—System requirements, installation and administration for 9600 Series SIP IP telephones on page 13
- Reference Point—Enhancing 4600 Series H.323 IP telephones to SIP telephones on page 14
- Reference point—System requirements, installation and administration for 4602 SIP telephones on page 14
- Reference point—System requirements, installation and administration for 4610SW SIP telephones on page 14
- Reference point—System requirements, installation and administration for 4620SW/4621SW SIP telephones on page 14
- Reference Point—Enhancing H.323 IP softphones to IM-enabled softphones on page 15
- Reference Point—Installing and configuring the Avaya SIP Softphone on page 15

Chapter 2: Setting up the Communication Manager server

- Reference point—List of related documents on page 20
- Reference point—OPS on page 21
- Reference point—Modular Messaging on page 21
- Reference point—Setting up Communication Manager servers on page 22
- Reference point—Administering SIP call routing in Communication Manager on page 22

Chapter 3: Setting up the server(s) running SES

- Reference point—Creating users via the user screens in SES on page 27
Reference Points

- Reference Point—Configuring Extension to Cellular and OPS for SIP on page 28

Chapter 4: Establishing SIP trunks in Communication Manager
- Reference Point—SIP Trunk Administration on page 30

Chapter 5: Setting up a SIP Endpoint
- Reference Point—Enhancing 9600 Series H.323 IP telephones to SIP telephones on page 32
- Reference Point—Enhancing 4600 Series H.323 IP telephones to SIP telephones on page 34
- Reference Point—Enhancing H.323 IP softphones to IM-enabled softphones on page 36
- Reference Point—Installing and configuring the Avaya one-X™ Desktop Edition on page 36
## Appendix B: Troubleshooting the SES solution

This Appendix lists common problems or symptoms and some suggested solutions to them.

<table>
<thead>
<tr>
<th>Problem/Symptom</th>
<th>Suggested Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phantom calls</td>
<td>If there is an active, ringing call to a SIP softphone, and there is a network outage, a phantom call to that extension occurs after network connectivity is restored. That is, when the network comes back up, the phone will ring and no one will be there. The customer receiving the call should simply hang up.</td>
</tr>
<tr>
<td>The server does not respond after a laptop is connected to the services port.</td>
<td>Use a serially connected keyboard and monitor to disable console redirection in the server BIOS, save, reboot and begin again.</td>
</tr>
<tr>
<td>The server cannot find files needed to boot.</td>
<td>Ensure that the boot order specified in the server BIOS indicates CD as first before attempting to boot from the Avaya CD.</td>
</tr>
<tr>
<td>An S8500 server keeps rebooting before the Installer script completes.</td>
<td>Disable the loader watchdog timer and begin installation again. (The default timer interval of 5 minutes for the RSA loader watchdog is insufficient to allow for complete installation.)</td>
</tr>
<tr>
<td>Link bounces, CRC errors, collisions, and other aberrations</td>
<td>Verify the network interface cards (NICs) in all Avaya servers (SES, media servers, etc.) and in enterprise LAN/WAN data equipment have been set to the 100Mbps full-duplex mode.</td>
</tr>
<tr>
<td>DNS errors occur and SIP calls cannot be made or received within the enterprise domain.</td>
<td>Do not use capital letters, spaces, underscores, or special characters (a hyphen is OK) in the names of domains.</td>
</tr>
<tr>
<td>External SIP calls via third-party servers are automatically dropped after a period of time.</td>
<td>Refer to instructions in SES 5.x Release Notes for using the <code>trustedhost</code> command.</td>
</tr>
<tr>
<td>Server authentication errors occur.</td>
<td>Refer to <em>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services</em>, 03-600768, Chapter 3: &quot;Setup and Configuration&quot;—&quot;Load the authentication file&quot;.</td>
</tr>
<tr>
<td>Server licensing errors occur.</td>
<td>Refer to the <em>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services</em>, 03-600768, Chapter 4: &quot;Installation Procedures&quot;—&quot;Server license installation&quot;.</td>
</tr>
<tr>
<td>Problem/Symptom</td>
<td>Suggested Solution</td>
</tr>
<tr>
<td>------------------------------------------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Alerts and events occur.</td>
<td>Refer to the <em>Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services</em>, 03-600768, Appendix B: SNMP Alerts.</td>
</tr>
<tr>
<td>SIP calls route to the wrong server.</td>
<td>Check the host address maps on the server(s) running SES software.</td>
</tr>
<tr>
<td>Presence information for Avaya IP Softphone 5.2 users cannot be seen in the IM client.</td>
<td>On the Add Host or Edit Host screen for the SES home serving these users, ensure that the Presence Access Policy (Default) field has been set to “Allow All”</td>
</tr>
<tr>
<td>4600 Series SIP IP telephones produce many alarms in the alarm log on the media server.</td>
<td>If a 46XX or another IP-based station type is administered in Communication Manager, many maintenance alarms are generated. If a DCP station type such as 6408D+ or 6424D+ is used, with X-port specified, then these maintenance alarms are not generated, but then undesired interactions with both PSA and TTI on the Communication Manager server may occur. See SES 5.x release notes for details. Note that OPS is required for any SIP-enabled extension.</td>
</tr>
</tbody>
</table>

2 of 2
Appendix C: Worksheet for installation

Use copies of this worksheet to answer all the questions asked by the install script. Use one worksheet for each server in your SES network.

**Table 13: SES Server Planning Form**
Complete with the customer for each server being installed.

<table>
<thead>
<tr>
<th>SES Server Type (Home, Edge or Combo)</th>
<th>Communication Manager HOME Communication Manager Edge</th>
</tr>
</thead>
<tbody>
<tr>
<td>Is this server duplicated? (High Availability - Yes or No)</td>
<td>This Server’s Role if Duplicated (A or B)</td>
</tr>
</tbody>
</table>

**Entered during ‘initial_setup’ sequence:**

<table>
<thead>
<tr>
<th>Host Name</th>
<th>(do not use special characters, spaces, or capital letters.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Domain Name</td>
<td>(do not use special characters, spaces, or capital letters.)</td>
</tr>
<tr>
<td>IP Address</td>
<td></td>
</tr>
<tr>
<td>Netmask</td>
<td></td>
</tr>
<tr>
<td>Gateway</td>
<td></td>
</tr>
<tr>
<td>Primary DNS IP Address</td>
<td></td>
</tr>
<tr>
<td>Secondary DNS IP Address</td>
<td></td>
</tr>
<tr>
<td>Tertiary DNS IP Address</td>
<td></td>
</tr>
<tr>
<td>Logical name of Redundant system</td>
<td>(only required on a duplex server pair)</td>
</tr>
<tr>
<td>Logical IP address of redundant system</td>
<td></td>
</tr>
<tr>
<td>Is this a Master Administrator?</td>
<td>(not prompted on B Server in Redundant configuration)</td>
</tr>
</tbody>
</table>

**Entered from the Web Interface:**

| SIP License Host IP Address | (typically the edge server, and is the physical address, not the logical one, if duplicated. Should only be enabled on one box.) |

1 of 2
Table 13: SES Server Planning Form (continued)
Complete with the customer for each server being installed.

<table>
<thead>
<tr>
<th><strong>Profile-Service Password</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(must be unique for each configured host)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Username WebLM Login</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(entered on the license host)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Password WebLM Login</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(customer owned)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>SIP Trunk - IP Address and Node Name</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(IP address of the CLAN or procr that the SIP trunk uses on Communication Manager)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Media Server Admin Login</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(new customer level super-user login that also needs adding on Communication Manager)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Media Server Admin Password</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(must be a minimum of 7 digits)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Media Server Admin Address</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(IP address of the Communication Manager Active server or procr for SAT access)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>NTP Server IP Address</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(for time synchronization)</td>
</tr>
</tbody>
</table>

Entered in DHCP/46xxsettings.txt File or 96xxsettings.txt file:

<table>
<thead>
<tr>
<th><strong>FTP Server IP Address</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(for backup of station profiles)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>TFTP/HTTP Server IP Address</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(for station firmware and 46xx or 96xx settings file)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>SIP Stations Dial Plan</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(dial string pattern as entered in 46xx or 96xx settings file to allow station dialing without using the 'send' key)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>DHCP Option 176 or 242 Option</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>(see LAN Admin Guide for more details)</td>
</tr>
</tbody>
</table>

Important Notes

<table>
<thead>
<tr>
<th><strong>INADS Line for remote connectivity required</strong></th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th><strong>Passwords must be maintained by the customer. If lost, it is a Time and Materials (T&amp;M) escalation.</strong></th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th><strong>Please make sure that this planning form is part of the permanent customer record (Maestro, EPROJECT, Rover, and so on.</strong></th>
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</tr>
<tr>
<td>4610SW phone</td>
<td>14</td>
</tr>
<tr>
<td>4620SW phone</td>
<td>14</td>
</tr>
<tr>
<td>4621SW phone</td>
<td>14</td>
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<td>4602 SIP phone</td>
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<tr>
<td>4602SW SIP phone</td>
<td>14</td>
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<tr>
<td>4610SW phone</td>
<td>14</td>
</tr>
<tr>
<td>4620 phone</td>
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<td>9640/9640G phone</td>
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<td>IP Agent R6</td>
<td>15</td>
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<td>IP softphone R5</td>
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<table>
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