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 their IP telephony network.

Using this document

Use this document as a reference point for deploying the Avaya SES solution. The document is 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>Table 1: Related documents</th>
</tr>
</thead>
<tbody>
<tr>
<td>Document Title</td>
</tr>
<tr>
<td>----------------</td>
</tr>
<tr>
<td>4600 Series IP Telephone LAN Administrator Guide</td>
</tr>
<tr>
<td>4610SW SIP Telephone User's Guide</td>
</tr>
<tr>
<td>4610SW SIP Telephone Quick Reference</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—Set up SES servers

- For detailed instructions on the initial setup of Avaya SIP Enablement Services Server, refer to
  *Installing and Administering 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
- *Installing and Administering SIP Enablement Services*, 03-600768
- *Avaya Toll Fraud and Security Handbook*, 555-025-600
- *Security and the Avaya S8700 Media Servers*, Avaya white paper
- SIP Softphone Quick Setup Guide
- SIP Softphone Getting Started
- *SIP Support in Avaya Communication Manager*, 555-245-206
- 4600 Series IP Telephone LAN Administrator Guide, 555-233-507
- *4600 Series IP Telephone Installation Guide*, 555-233-128
- *Avaya 4602/4602SW SIP Telephone User’s Guide*, 16-300035
- *Avaya 4602/4602SW SIP Telephone Quick Reference*, 16-300471
- *Avaya 4610SW SIP IP Telephone User’s Guide*, 16-300472
- *Avaya 4610SW SIP IP Telephone Quick Reference*, 16-300473
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`.

- 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.
About this document

● 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 to 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 IM
- 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 and Administering SIP Enablement Services, 03-600768, Chapter 2: "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 media servers and Communication Manager.
   See Chapter 2: Setting up the media server on page 17

2. Install and configure Avaya SIP Enablement Services on the SES servers
   See Chapter 3: Setting up the servers running SES on page 23

3. Integrate Communication Manager, SIP Enablement Services, and the 4600 series telephones via Extension to Cellular and Off-PBX Station (OPS).
   See Setting up OPS stations on page 26.

4. Establish SIP Trunks in Communication Manager.
   See Chapter 4: Establishing SIP trunks in Communication Manager on page 27.

5. Configure the Release 2.2 Series SIP telephone for use with the SES solution.
   See Chapter 5: Setting up a SIP Endpoint on page 29.
Overview of Avaya SES solution components

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

- **Avaya SIP Enablement Services server**
- One or more [Media servers running Avaya Communication Manager R4.0](#)
- **Gateways**
- SIP endpoints
  - **Avaya 4600 Series IP Telephones supporting SIP**
  - **Avaya softphones**
- **Third-party SIP clients**

Avaya SIP Enablement Services server

The SES application runs on Avaya’s S8500-series hardware platform. 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 media servers, the SES server supports communication between and among the various non-SIP endpoints. 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 might be any from this list:

- Avaya 4600 series telephones
- Avaya SIP Softphone Release 2 or later
- Avaya IP Softphone Release 5 and later

SIP-enabled endpoints must register with the SES server. Although an Avaya IP Softphone is not managed by a media server, it is required to register H.323 with Avaya Communication Manager. Although IP softphones use H.323 for voice, the SIP as a protocol is also used for communications. For example, instant messages on IP softphones use SIP.
Reference point—System requirements, installation and administration for the SES server

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

- System requirements - "Chapter 2: Requirements for the SIP solution"
- Installation - "Chapter 3: Setup and configuration", and "Chapter 4: Installation procedures"
- Administration - "Chapter 3: Setup and configuration—Installation checklist" "Chapter 3: Setup and configuration—Post installation tasks"
- Administration in general - "Chapter 5: Administration web interface"

Media servers running Avaya Communication Manager R4.0

Avaya's SES solution can be deployed with the S8300, S8400, S8500-series, and the S8700-series media 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. These 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 R4.0 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 SIP solution with SES can employ any of the Avaya gateways.
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, first 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 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, 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 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 28 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, the administrator 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 SIP Softphone R2.x

SIP can be deployed with either the Avaya IP or SIP softphone clients. To deploy the SIP Softphone solution, the client software must be installed on the SIP user’s PC by an administrator of that PC.

The softphone must have the current version of firmware, 2.2 at the time of this writing.

Reference Point—Installing and configuring the Avaya SIP Softphone

For detailed instructions on installing the Avaya SIP Softphone client software, refer to SIP softphone 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), as 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 in order to be tested and ultimately documented as supported applications/devices.

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

http://www1.avaya.com/enterprise/sig/devconnect/

For application notes on which third-party SIP endpoints are supported, go to the web site: http://www.avaya.com and select Research By: Resource Type in the middle of the page. Then select Resource Type >> Application Notes.
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>white paper</td>
</tr>
</tbody>
</table>

Security and SES

These points are part of the SES R4.0 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 of TLS. SIP signaling to CM is always protected using TLS.
- Telephony end users are authenticated using SIP digest authentication.
Chapter 2: Setting up the media server

When setting up the media server to accommodate SES, you will need these references available:

Table 3: Documents required for setting up the media 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 Avaya representatives</td>
</tr>
<tr>
<td>Installing and Administering SIP Enablement Services, 03-600768</td>
<td>03-600768</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 media servers:

- Avaya S8300 Media Server
- Avaya S8400 Media Server
- Avaya S8500-series Media Server
- Avaya S8700-series Media Server

Note:
Prior to setting up the media servers to enable SIP, IP connectivity must be configured correctly. For details on this topic, refer to Administration for Network Connectivity for Avaya Communication Manager, 555-233-504.

Setting up the Avaya media server

As noted above, prior to setting up the Avaya media server for SIP, you must configure IP connectivity correctly. After IP connectivity is configured, complete the following tasks:

- Setting up SIP support in Avaya Communication Manager on the media servers on page 18
- Setting up the servers running SES on page 23
- Setting up Click to Conference on page 20
Setting up the media server

- Setting up Advanced SIP Telephony on page 20
- Setting up Modular Messaging as an adjunct system on page 21

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 Avaya SES server(s), refer to Installing and Administering SIP Enablement Services, 03-600768:
  - Chapter 3: "Setup and configuration—Pre installation checklist"
  - Chapter 3: "Setup and configuration—Post installation tasks"
  - Chapter 4: "Installation procedures—Administer Communication Manager and endpoints"
- For information on setting up Avaya Communication Manager, refer to SIP Support in Avaya Communication Manager, 555-245-206, Chapter 2: "SIP Support in Communication Manager".

Setting up SIP support in Avaya Communication Manager on the media servers

Avaya’s SES solution can be deployed with the S8300, S8400, S8500-series, and the S8700-series media 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. These gateways are optimized for use with enterprise class telephony and to provide a variety of flexible deployment options. Options include partial environments, such as IP and TDM, and full IP environments.

With Avaya Communication Manager, SIP features include these:

- SIP trunking
- SIP stations
- Call Detail Record (CDR)
- Access control
- Routing
- Dial plan
SIP Trunking

Support for SIP trunks allows an enterprise to connect its media servers to a SIP-enabled proxy server, the Avaya SIP Enablement Services (SES) server, and through the proxy to an external SIP service provider, if desired. The trunk support in 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 or S8710 media servers, the signaling-group properties in Communication Manager must be administered to match in certain 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. 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 Off-PBX Station (OPS) Installation and Administration Guide, 210-100-500

Reference point—Modular Messaging

- For using Modular Messaging as an adjunct system to SES, see the SES installation and administration guide: Installing and Administering SIP Enablement Services, 03-600768, Chapter 5: "Administration Web Interface—Adjunct Systems".

Access control

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

Reference point—Setting up the media servers

All necessary information for setting up the media 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, SIP Support in Avaya Communication Manager—Requirements for SIP".
- For setting up and configuring SIP trunks and Communication Manager, refer to Chapter 3: "Administering Communication Manager for SIP".

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 media servers. This involves several screens, including for example the **Locations**, **Numbering**, and **Route Pattern** screens.

Reference point—Administering call routing on the media servers

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".

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 an Avaya representative.

Setting up Advanced SIP Telephony

For application notes on how to set up SES for use with Advanced SIP Telephony, go to the website:

http://www.avaya.com and select Research By: Resource Type in the middle of the page. Then select the link for Application Notes under Resource Type. Filter the results for AST.
Setting up Modular Messaging as an adjunct system

For administration guidance on how to implement Avaya’s Modular Messaging product as an adjunct system to SES, see this material:

*Installing and Administering SIP Enablement Services*, 03-600768, Chapter 5: "Administration Web Interface—Adjunct Systems".

*Configuration Note 88010*—This note will be published on the Avaya support web site soon.
Setting up the media server
Chapter 3: Setting up the servers running SES

When working with the servers that have SES installed on them, you will need 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 and Administering SIP Enablement Services, 03-600768</td>
<td>03-600768</td>
</tr>
<tr>
<td>Avaya Extension to Cellular and off-PBX Station (OPS) Installation and Administration Guide</td>
<td>210-100-500</td>
</tr>
<tr>
<td>Avaya Server Availability Management Processor User Guide (SAMP) (for S8500B)</td>
<td>03-300322</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 and configured on an enterprise’s IP network. Appendix C: Worksheet for installation on page 37 provides a complete list of the information you will need to answer the install script questions.

**Note for CCS 2.x to SES 3.x remasters:** - For all existing CCS systems comprising S8500 servers with the Avaya Remote Supervisor Adapter (RSA) module, the RSA loader watchdog must be disabled before the installation process is performed. (By default, a timer of 5 minutes is enabled.) After all the software has been installed and verified on the server (or on both the A and B servers in a duplex pair), then the RSA loader watchdog should be re-enabled.

**Note for SES 3.0 to SES 3.1 upgrades:** - If you are upgrading from SES R3.0 to SES R3.1, use the three Server Upgrade screens in the maintenance interface of SES.
Trunking

For endpoint clients on Avaya Communication Manager to interoperate through the Avaya SES server, SIP trunking must be administered properly on the media server running the Communication Manager software. Refer to *SIP Support in Avaya Communication Manager, 555-245-206, Chapter 3: Setup and Configuration* for more details. This administration requires that you specify these values:

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

**Note:**
For 4600 Series SIP IP Telephones, the same authoritative domain must be specified in the phone’s settings file. Also, unlike the H.323 versions of these phones, the SIP versions must have network access to their TFTP or HTTP servers upon startup.

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Task list for configuring the server for SES

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

**Table 5: Configuration task list for the server running SES**

<table>
<thead>
<tr>
<th>✔</th>
<th>Task</th>
<th>Reference document</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Perform initial assembly and setup.</td>
<td><em>Installing and Administering SIP Enablement Services, 03-600768</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td>● Chapter 3, Setup and configuration—Configuring a new server</td>
</tr>
<tr>
<td></td>
<td></td>
<td>● Chapter 3, Setup and Configuration—Pre-installation tasks</td>
</tr>
<tr>
<td></td>
<td>Install all server software.</td>
<td><em>Installing and Administering SIP Enablement Services, 03-600768</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td>● Chapter 4: Installation Procedures</td>
</tr>
</tbody>
</table>
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 user, refer to *Installing and Administering SIP Enablement Services*, 03-600768, Chapter 5: "Administration web interface—User Screens—Add User screen".

- For user media server administration, refer to *Installing and Administering 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 by means of ProVision, refer to *Installing and Administering 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/

Table 5: Configuration task list for the server running SES (continued)

<table>
<thead>
<tr>
<th>✔</th>
<th>Task</th>
<th>Reference document</th>
</tr>
</thead>
</table>
|  | Perform administrative tasks on the server by way of a Point to Point Protocol (PPP) session. | *Installing and Administering SIP Enablement Services*, 03-600768  
- Chapter 3, Setup and configuration—Installation checklist  
- Chapter 3, Setup and configuration—Post installation checklist. |
|  | Install the server license by way of WebLM. | *Installing and Administering SIP Enablement Services*, 03-600768  
Chapter 4: " Installation Procedures—Server license installation" |
|  | Deliver the authentication file by way of the Automatic Registration Tool (ART). | *Installing and Administering SIP Enablement Services*, 03-600768  
Chapter 3: "Setup and Configuration—Load the authentication file" |

2 of 2
Setting up the servers running SES

Setting up OPS stations

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 Off-PBX Station (OPS) Installation and Administration Guide</td>
<td>210-100-500</td>
</tr>
</tbody>
</table>

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.

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.
Chapter 4: Establishing SIP trunks in Communication Manager

When you set up the SIP trunks on the media server, 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 and Administering SIP Enablement Services, 03-600768</td>
<td>03-600768</td>
</tr>
</tbody>
</table>

The Session Initiated Protocol (SIP) is an endpoint-oriented messaging standard defined by the Internet Engineering Task Force (IETF). SIP trunking enables the Avaya media servers running Communication Manager R4.0 to function as a POTS gateway and support the delivery of name and number between legacy endpoints and SIP-enabled endpoints.

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

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 on the media server. The server running SES enables 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—Administer SIP trunks".

Table 8: SIP Trunk capacity

<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 equipment.

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

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

http://www1.avaya.com/enterprise/sig/devconnect/

For application notes on which third-party SIP endpoints are supported, go to the web site:

http://www.avaya.com and select Research By: Resource Type in the middle of the page.

Then select the link for Application Notes under Resource Type. You may filter results using SIP.
Chapter 5: Setting up a SIP Endpoint

When you configure the endpoints for your SES system, use document tables Table 10 and Table 11 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 R4.0 software.

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

<table>
<thead>
<tr>
<th>Telephone</th>
<th>Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>4602 IP telephone</td>
<td>not supported</td>
</tr>
<tr>
<td>4602/4602SW SIP telephone</td>
<td>Release 2.2</td>
</tr>
<tr>
<td>4610SW SIP telephone</td>
<td>Release 2.2</td>
</tr>
<tr>
<td>4620SW/4621SW SIP telephone</td>
<td>Release 2.2</td>
</tr>
</tbody>
</table>

Confer with your Avaya representative to see if versions other than the ones listed here are released.
Avaya 4602/4602SW, 4610SW and 4620SW/4621SW SIP telephones

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 Research By: Resource Type in the middle of the page.

Then select the link for Application Notes under Resource Type. You may filter results using SIP or the model number of your Avaya telephone.
Table 11: Documents required for setting up the Avaya softphones

<table>
<thead>
<tr>
<th>Document Title</th>
<th>Document ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Softphone R2.1 Quick Setup Guide</td>
<td><a href="http://www.support.avaya.com">www.support.avaya.com</a> SIP Support area</td>
</tr>
<tr>
<td>SIP Softphone R2.1 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 with IM client</td>
<td>online help</td>
</tr>
<tr>
<td>Avaya SIP Softphone Release 2.x</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 R2 before installing 2.1 SIP softphone.

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

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

Reference Point—Installing and configuring the Avaya SIP Softphone

For detailed instructions on installing the Avaya SIP softphone client software, refer to SIP softphone online help:

http://www.support.avaya.com > 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 quickly refer to a document for a specific task.

About this document

- Reference point—Set up SES servers on page 6
- Reference point—Defined on page 7

Chapter 1: SES Solution Overview

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

Chapter 2: Setting up the media server

- Reference point—List of related documents on page 18
- Reference point—OPS on page 19
- Reference point—Modular Messaging on page 19
- Reference point—Administering call routing on the media servers on page 20
- Reference point—Setting up the media servers on page 20

Chapter 3: Setting up the servers running SES

- Reference point—Creating users via the user screens in SES on page 25
- Reference Point—Configuring Extension to Cellular and OPS for SIP on page 26

Chapter 4: Establishing SIP trunks in Communication Manager

- Reference Point—SIP Trunk Administration on page 28
Reference Points

Chapter 5: Setting up a SIP Endpoint

- Reference Point—Enhancing 4600 Series H.323 IP telephones to SIP telephones on page 30
- Reference Point—Enhancing H.323 IP softphones to IM-enabled softphones on page 32
- Reference Point—Installing and configuring the Avaya SIP Softphone on page 32
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 RSA loader watchdog and begin installation again. (The default timer interval of 5 minutes for the loader watchdog is insufficient to allow for full 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 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 Release Notes for using the trustedhost command.</td>
</tr>
<tr>
<td>Server authentication errors occur.</td>
<td>Refer to Installing and Administering SIP Enablement Services, 03-600768, Chapter 3: Setup and Configuration—Load the authentication file.</td>
</tr>
<tr>
<td>Server licensing errors occur.</td>
<td>Refer to the Installing and Administering SIP Enablement Services, 03-600768, Chapter 4: &quot;Installation Procedures—Server license installation&quot;.</td>
</tr>
</tbody>
</table>
## Troubleshooting the SES solution

<table>
<thead>
<tr>
<th>Problem/Symptom</th>
<th>Suggested Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Alerts and events occur.</td>
<td>Refer to the <em>Installing and Administering SIP Enablement Services</em>, 03-600768, &quot;Appendix B: SNMP Alerts&quot;.</td>
</tr>
<tr>
<td>SIP calls route to the wrong server.</td>
<td>Check the host address maps on servers 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 other IP station type is administered in Communication Manager, many maintenance alarms are generated. If a DCP set type such as 6408D+ or 6424D+ is used, with X port specified, then these maintenance alarms are not generated, but undesired interactions with PSA and TTI on the media server may occur. See SES release notes for details. Note that OPS is required for any SIP-enabled extension.</td>
</tr>
</tbody>
</table>
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 12: 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>
</tr>
</thead>
<tbody>
<tr>
<td>Is this server duplicated? (High Availability - Yes or No)</td>
</tr>
<tr>
<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>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Domain Name</td>
</tr>
<tr>
<td>IP Address</td>
</tr>
<tr>
<td>Netmask</td>
</tr>
<tr>
<td>Gateway</td>
</tr>
<tr>
<td>Primary DNS IP Address</td>
</tr>
<tr>
<td>Secondary DNS IP Address</td>
</tr>
<tr>
<td>Tertiary DNS IP Address</td>
</tr>
<tr>
<td>Admin password</td>
</tr>
<tr>
<td>Logical name of Redundant system (only required on a duplex server pair)</td>
</tr>
<tr>
<td>Logical IP address of redundant system</td>
</tr>
<tr>
<td>Is this a Master Administrator? (only resides on an Edge server, Yes or No)</td>
</tr>
<tr>
<td>IP Address of Master Administrator for THIS machine (only asked on a Home server - on a Duplex pair this is the LOGICAL IP Address)</td>
</tr>
</tbody>
</table>

| mvss database password |

1 of 3
### Worksheet for installation

**Table 12: SES Server Planning Form (continued)**

Complete with the customer for each server being installed.

**Only applies to RSA board on S8500 (x305) platform:**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Host name of RMB card</td>
<td></td>
</tr>
<tr>
<td>IP Address of RMB eth port</td>
<td></td>
</tr>
<tr>
<td>Netmask of RMB eth port</td>
<td></td>
</tr>
<tr>
<td>Gateway of RMB eth port</td>
<td></td>
</tr>
<tr>
<td>User Name RMB Login</td>
<td></td>
</tr>
<tr>
<td>Password RMB Login</td>
<td></td>
</tr>
</tbody>
</table>

**Entered from the Web Interface:**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>License Host IP Address (typically the edge server, and is the physical address, not the logical one, if duplicated)</td>
<td></td>
</tr>
<tr>
<td>Profile-Service Password (must be unique for each configured host)</td>
<td></td>
</tr>
<tr>
<td>Username WebLM Login (entered on the license host)</td>
<td></td>
</tr>
<tr>
<td>Password WebLM Login</td>
<td></td>
</tr>
<tr>
<td>SIP Trunk - IP Address and Node Name (IP address of the CLAN or proc that the SIP trunk uses on CM)</td>
<td></td>
</tr>
<tr>
<td>Media Server Admin Login (new customer level super-user login that also needs adding on Communication Manager)</td>
<td></td>
</tr>
<tr>
<td>Media Server Admin Password (must be a minimum of 7 digits)</td>
<td></td>
</tr>
<tr>
<td>Media Server Admin Address (IP address of the Communication Manager Active server or proc for SAT access)</td>
<td></td>
</tr>
<tr>
<td>NTP Server IP Address (for time synchronization)</td>
<td></td>
</tr>
</tbody>
</table>
Table 12: SES Server Planning Form (continued)
Complete with the customer for each server being installed.

Entered in DHCP/46xx Settings.txt File:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FTP Server IP Address</td>
<td>(for backup of station profiles)</td>
</tr>
<tr>
<td>TFTP/HTTP Server IP Address</td>
<td>(for station firmware and 46xx settings file)</td>
</tr>
<tr>
<td>SIP Stations Dial Plan</td>
<td>(dial string pattern as entered in 46xx settings file to allow station dialing without using the 'send' key)</td>
</tr>
<tr>
<td>DHCP Option 176 String</td>
<td>(see LAN Admin Guide for more details)</td>
</tr>
</tbody>
</table>
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