Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Installation and Maintenance Guide Release 2.2

16-603159
Issue 1
September 2008

Notice

While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document may be incorporated in future releases.

For full legal page information, please see the complete document, Avaya Legal Page for Hardware Documentation, Document number 03-600759.

To locate this document on our Web site, simply go to http://www.avaya.com/support and search for the document number in the search box.

Documentation disclaimer

Avaya Inc. is not responsible for any modifications, additions, or deletions to the original published version of this documentation unless such modifications, additions, or deletions were performed by Avaya. Customer and/or End User agree to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation to the extent made by the Customer or End User.

Link disclaimer

Avaya Inc. is not responsible for the contents or reliability of any linked Web sites referenced elsewhere within this documentation, and Avaya does not necessarily endorse the products, services, or information described or offered within them. We cannot guarantee that these links will work all of the time and we have no control over the availability of the linked pages.

Warranty

Avaya Inc. provides a limited warranty on this product. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language, as well as information regarding support for this product, while under warranty, is available through the following Web site:

http://www.avaya.com/support

Copyright

Except where expressly stated otherwise, the Product is protected by copyright and other laws respecting proprietary rights. Unauthorized reproduction, transfer, and or use can be a criminal, as well as a civil, offense under the applicable law.

Avaya support

Avaya provides a telephone number for you to use to report problems or to ask questions about your product. The support telephone number is 1-800-242-2121 in the United States. For additional support telephone numbers, see the Avaya Web site:

http://www.avaya.com/support

Software License

USE OR INSTALLATION OF THE PRODUCT INDICATES THE END USER'S ACCEPTANCE OF THE TERMS SET FORTH HEREIN AND THE GENERAL LICENSE TERMS AVAILABLE ON THE AVAYA WEBSITE AT http://support.avaya.com/LicenseInfo/ ("GENERAL LICENSE TERMS"). IF YOU DO NOT WISH TO BE BOUND BY THESE TERMS, YOU MUST RETURN THE PRODUCT(S) TO THE POINT OF PURCHASE WITHIN TEN (10) DAYS OF DELIVERY FOR A REFUND OR CREDIT.

Avaya grants End User a license within the scope of the license types described below. The applicable number of licenses and units of capacity for which the license is granted will be one (1), unless a different number of licenses or units of capacity is specified in the Documentation or other materials available to End User. "Designated Processor" means a single stand-alone computing device. "Server" means a Designated Processor that hosts a software application to be accessed by multiple users. "Software" means the computer programs in object code, originally licensed by Avaya and ultimately utilized by End User, whether as stand-alone Products or pre-installed on Hardware. "Hardware" means the standard hardware Products, originally sold by Avaya and ultimately utilized by End User.

License Type(s):

Designated System(s) License (DS). End User may install and use each copy of the Software on only one Designated Processor, unless a different number of Designated Processors is indicated in the Documentation or other materials available to End User. Avaya may require the Designated Processor(s) to be identified by type, serial number, feature key, location or other specific designation, or to be provided by End User to Avaya through electronic means established by Avaya specifically for this purpose.

Third-party Components

Certain software programs or portions thereof included in the Product may contain software distributed under third party agreements ("Third Party Components"), which may contain terms that expand or limit rights to use certain portions of the Product ("Third Party Terms"). Information identifying Third Party Components and the Third Party Terms that apply to them is available on Avaya's Web site at:

http://support.avaya.com/ThirdPartyLicense/

Interference

Using a cell, mobile, or GSM telephone, or a two-way radio in close proximity to an Avaya IP Telephone might cause interference.

Patents

T9 Text Input and other products are covered by one or more of the following patents: U.S. Pat. Nos. 5,187,480,5,818,437, 5,945,928, 5,953,541, 6,011,554, 6,286,064, 6,307,548, 6,307,549, and 6,636,162,6,646,573, 6,970,599; Australia Pat. Nos. 727539, 746674, 747901; Austria Pat. Nos.AT225534, AT221222; Brazil P.I. No. 9609807-4; Canada Pat. Nos. 1,331,057, 2,227,904,2,278,549, 2,302,595; Japan Pat. Nos. 3532780, 3492981; United Kingdom Pat. No.2238414B; Hong Kong Standard Pat. No. HK1010924; Republic of Singapore Pat. Nos.51383, 66959, 71979; European Pat. Nos. 1 010 057 (98903671.0), 1 018 069 (98950708.2); Republic of Korea Pat. Nos. KR201211B1, KR226206B1, 402252; People's Republic of China Pat. No. ZL96196739.0; Mexico Pat. Nos. 208141, 216023, 218409; RussianFederation Pat. Nos. 2206118, 2214620, 2221268; and additional patent applications are pending.

Contents

Chapter 1: Introduction

About This Guide

This guide describes how to install and maintain the 9600 Series IP Telephones in a Session Initiation Protocol (SIP) environment.

The 9600 Series IP Telephones product line supports two signaling protocols, the Session Initiation Protocol (SIP) and the H.323 protocol. Both of the following must be installed to use 9600 Series IP Telephones with the SIP protocol:

- Avaya Communication Manager Release 4.0 and greater, and
- SIP Enablement Session (SES) software Release 4.0 and greater.

Note:

Any reference to HTTP in this guide applies equally to HTTPS.

When running the 9600 Series IP Telephones in an H.323 environment see the *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Installation and Maintenance Guide* for installation and maintenance information.

This document does not cover administration for Avaya Distributed Office. Full documentation for Avaya Distributed Office is available on the Avaya support Web site, www.avaya.com/support.

Intended Audience

This document is intended for personnel who install and administer the 9600 Series SIP IP Telephones.

CAUTION:

Avaya does not provide product support for many of the products mentioned in this document. Take care to ensure that there is adequate technical support available for the servers involved, including, but not necessarily limited to, HTTP, HTTPS, and DHCP servers. If the servers are not functioning correctly, the IP telephones might not be able to operate correctly.

Document Organization

The guide contains the following sections:

Chapter 1: Introduction	Provides an overview of this guide.
Chapter 2: 9600 Series SIP IP Telephone Installation	Describes the equipment and resources required to properly install and operate the 9600 Series SIP IP Telephones. Provides instructions on installing the telephones out of the box.
Chapter 3: Local Administrative Options	Describes how to set local administrative options, if requested by the system or LAN administrator.
Chapter 4: Maintaining 9600 Series SIP IP Telephones	Describes maintenance actions like downloading telephone software from the Avaya support Web site and customizing system values.
Chapter 5: Troubleshooting Guidelines	Describes error conditions and messages that might occur during the installation of the 9600 Series SIP IP Telephones.
Appendix A: Restart Scenarios	Explains the different scenarios possible for the sequence of the restart process.
Appendix B: Glossary of Terms	Provides a glossary of terms used in this document or which are generally applicable to 9600 Series SIP IP Telephones.
Appendix C: Related Documentation	Provides references to external documents that relate to telephony in general.

Issue Date

This is the second release of this document, issued in December, 2007. This document was first issued in May, 2007.

What's New in This Release

New material in this issue to support SIP Release 2.2 software includes:

Language Support - 9600 Series SIP IP Telephones now support Hebrew and Arabic. These phones also support Hebrew and Korean for text entry. For more information see the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*

Push Feature: The administrator can push content including text, web pages and audio files to the phone. Additionally the administrator can use this feature to clear the call log and web history, to change the phone language and username name, and to load a customized screen.

New UI Customization Capabilities: The User Interface for the 9620, 9630, and 9640 SIP IP telephones can now be customized via an xml file. You can use this file to link screens together, launch phone applications, and disable hardbuttons, among other things. Detailed information on how to do this can be found in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Developer Guide*.

New Logo Features: You now have the option to change the logo that displays on the 9600 Series SIP IP telephone.

New Skin Features: You now have the option to change the skin that displays on the 9640 SIP IP telephone.

New Configuration Parameters - The following configuration parameters have been added for this release. Find details about these parameters in "Chapter 8" of the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.

- USER_DISPLAY_NAME
- CURRENT_CONTENT
- LANGUAGE_DIRECTION
- TPSLIST
- SUBSCRIBELIST
- PUSHPORT
- PUSHCAP
- SYMMETRIC_RTP
- DISPLAY_NAME_NUMBER
- INGRESS_DTMF_VOL_LEVEL

Online Documentation

See the Avaya support site at http://www.avaya.com/support for 9600 Series SIP IP Telephone technical and end user documentation.

See Appendix C: Related Documentation for Web sites that list related, non-Avaya documents, such as those published by the Internet Engineering Task Force (IETF) and the International Telecommunication Union (ITU).

Customer Support

For 9600 Series SIP IP Telephone support, call the Avaya support number provided to you by your Avaya representative or Avaya reseller.

Information about Avaya products can be obtained at the following URL:

http://www.avaya.com/support

Chapter 2: 9600 Series SIP IP Telephone Installation

Introduction

The 9600 Series SIP IP Telephones use Internet Protocol (IP) technology with Ethernet interfaces. The IP telephones supplement the existing Avaya IP Solutions platform.

The 9600 Series SIP IP Telephones support DHCP and HTTP/HTTPS over IPv4/UDP which enhance the administration and servicing of the telephones. These telephones use DHCP to obtain dynamic IP Addresses and HTTP or HTTPS to download new software versions or customized settings for the telephones.

All 9600 Series SIP IP Telephones provide the ability to have one IP connection on the desktop for both a telephone set and a PC using an Ethernet switch.

In compliance with Australian law, the following information is provided:

> This equipment shall be installed and maintained by trained service personnel. All the input/output ports are classified as Safety Extra Low Voltage (SELV, in the meaning of IEC 60950). To maintain safety compliance when connecting the equipment electrically to other equipment, the interconnecting circuits shall be selected to provide continued conformance of clause 2.3 for SELV circuits (generally, double/reinforced insulation to 240Vac rms to any primary/mains circuitry and 120Vac rms to any telecommunications network circuitry). To ensure that these conditions are adhered to, interconnect the equipment only with the already approved/certified equipment.

IP Telephone Models

There are five telephone set models currently defined in the 9600 Series SIP IP Telephone family:

- 9620 9630G 9640G
- 9630 9640

The telephones have an internal Ethernet switch that allows the telephone and a PC to share the same LAN connection, if appropriate. Thus, 9600 models do not need, or work with, the 30A switched hub interface. Telephone models appended with "G" have a gigabit Ethernet (GigE) interface, which speeds data transmission.

This document describes the installation of these telephones and post-installation maintenance issues. For details about using the features provided by the telephones, see the user documentation for each telephone. For information about desk or wall mounting any of the 9600

Series SIP IP Telephones, see the instructions boxed with the telephone. Wall or desk mount instructions are also available on the Avaya support Web site http://www.avaya.com/support.

Software

The 9600 Series IP Telephones ship from the factory set to the H.323 protocol. To run the telephones in a SIP environment, you must convert the telephone(s) to SIP firmware. Further, a factory-shipped 9600 Series IP Telephone will not contain the most up-to-date software for registration and SIP operation. When the telephone is first plugged in, a software download from an HTTP server might be initiated. The software download gives the telephone upgraded H.323 functionality, however, you must still download the latest SIP software bundle for telephones to be converted to SIP, then convert applicable telephones to run SIP software, as described in Converting Software on 9600 Series IP Telephones on page 12.

For subsequent downloads of software upgrades, SIP Enablement Services (SES) provides the capability for a remote reboot of the IP telephone. As a consequence of restarting, the telephone automatically downloads new software if it is available. Chapter 4: Maintaining 9600 Series SIP IP Telephones covers downloading new software releases.

Pre-Installation Checklist

Before plugging in the 9600 Series IP Telephones, verify that all the following requirements are met. Failure to do so prevents the telephones from working properly and can have a negative impact on the network. Print copies of this checklist for each server and IP telephone.

Verify These Network Requirements

- ☐ **1.** Ensure that the LAN uses Ethernet Category 5e cabling running the IPv4 version of Internet Protocol.
- ☐ **2.** Ensure that the following is installed and/or set up and operative:
  - Avaya Communication Manager (CM) Release 4.0 or greater.
  - SIP Enablement Services 4.0 or greater. 9600 Series SIP IP Telephones with SIP Release 2.0 software registered with SES 4.0 servers have only those features compatible with that server.
  - NTP Time Server.

  Important:
  The above must be configured properly to support SIP. The CM Outboard Proxy SIP (OPS) Station Form must be completed to enable SIP prior to plugging in the telephones. For information, see SIP Support in Avaya Communication Manager Running on Avaya S8XXX Servers (Document Number 555-245-206).

Verify These Network Requirements (continued)

❑ **3.** The following circuit packs are installed on the switch:
- TN2602 IP Media Processor circuit pack. Sites with a TN2302 IP Media Processor circuit pack are strongly encouraged to install a TN2602 circuit pack to benefit from increased capacity.
- TN799B, C, or D Control-LAN (C-LAN) circuit pack.

Important:

IP telephone firmware requires TN799C V3 or greater C-LAN circuit pack(s). For more information, see the *Communication Manager Software and Firmware Compatibility Matrix* on the Avaya support Web site http://www.avaya.com/support.

❑ **4.** The Communication Manager (CM) call server is configured correctly, as described in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide* and Avaya Communication Manager documentation. Both documents are available at http://www.avaya.com/support.

❑ **5.** The DHCP server and application are administered as described in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.

❑ **6.** The HTTP server and application are administered as described in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*. Extra files need to be administered if you want to customize the telephone user interface. Information on this can be found in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Developer Guide*.

❑ **7.** The SIP upgrade script and application files from the Avaya Support Web site, http://www.avaya.com/support, are loaded correctly on the HTTP/HTTPS server.

8. If applicable, the Voice Mail server is administered as described in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.

Notes:

- Any or all of the server applications mentioned in items 5-8 can be co-resident on the same hardware, subject to the specific restrictions of each individual application.
- See the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide* for more information about:
  - administering other network equipment,
  - administering applications like firewalls, and
  - information about topics like port utilization.

Requirements to Verify for Each IP Telephone

❑ **9.** You have an extension number and an Avaya Communication Manager security code (password) for each applicable IP telephone.

❑ **10.** You have an OPTIM extension number and an Avaya Communication Manager security code (password) for each telephone, and have configured SIP Enablement Services for each telephone.

❑ **11.** A Category 5e LAN jack is available at each telephone site.

- ❑ **12.** Electrical power is provided to each telephone by a Telephone Power Module (DC power jack) (must be ordered separately). If the LAN provides PoE (Power over Ethernet) to the telephone, no power module is required.
- ❑ **13.** 1 Category 5e modular line cord is available for the connection between the IP telephone and the PC, if applicable.
- ❑ **14.** Verify that the 9600 Series IP Telephone package includes the following components:
  - 1 telephone set with pre-attached stand.
  - 1 handset capable of transmitting and receiving 7KHz audio.
  - 1 H4DU 9-foot long (when extended) 4-conductor coiled handset cord, plugged into the telephone and the handset.
  - 1 Category 5e modular line cord for the connection from the IP telephone to the Ethernet wall jack.
  - *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Safety Instructions.*
  - *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Stand Instructions*.
  - *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Wall Mount Instructions*.
- ❑ **15.** IP telephones ship from the factory with H.323 software. Existing installations might also have many IP telephones running H.323 software. For instructions on how to convert between H.323 and SIP software, see Converting Software on 9600 Series IP Telephones.

Note:

For sites using headsets, the 9600 Series SIP IP Telephones support only the HIS headset cords.

Converting Software on 9600 Series IP Telephones

9600 Series IP Telephones use either H.323 or SIP software but come from the factory with H.323 software loaded by default. After telephone connection, you should ensure that those telephones that will run under a SIP protocol are set up properly. This section describes how to determine what your telephone "environment" is and then convert applicable telephones from H.323 to SIP software, or from SIP to H.323 software.

There are several H.323 to SIP or SIP to H.323 conversion scenarios, and each scenario depends on whether the majority of your telephones are H.323 or SIP:

- **H.323-Centric** - an environment where the majority of IP telephones are and will remain running H.323 software, but some telephones will become SIP IP telephones. In an H.323-centric environment, the appropriate H.323 telephone binary files must reside on the HTTP server and Communication Manager must be configured with the appropriate H.323 parameters. To convert an individual telephone from H.323 to SIP, both the SIP Enablement Services (SES) server and Avaya Communication Manager (CM) must be configured with the appropriate SIP parameters. Any telephone in use prior to conversion must run Release S1.2 or greater software with a SIG parameter value of “default”

(H.323). See Table 1, the H.323 to SIP and SIP to H.323 Conversion Chart for conversion instructions.

- **SIP-Centric** - an environment where the majority of IP telephones are or will become SIP telephones running SIP software. In a SIP-centric environment, the ***96xxSIP...*** software bundle must reside on the HTTP server and both SES and CM must be configured with the appropriate SIP parameters. To convert an individual telephone from SIP to H.323, Avaya Communication Manager (CM) must be configured with the appropriate H.323 parameters. Any SIP telephone in use prior to conversion must run Release SIP 1.0 or greater software with a SIG parameter value of “default” (SIP). See Table 1, the H.323 to SIP and SIP to H.323 Conversion Chart for conversion instructions.

What makes an environment H.323- or SIP-centric depends on the type of upgrade script files the environment is running (H.323 or SIP, see Downloading Software Upgrades on page 45) and the Signaling Protocol Identifier (SIG) parameter setting. The SIG parameter has three possible values:

- Default - either H.323 or SIP, set automatically for all telephones depending on whether your environment is H.323-centric or SIP-centric as determined by the software bundle downloaded and the changes you make to the ***alternate_96xxupgrade.txt*** file.
- H.323 - manually set to H.323 for a specific telephone by an installer or administrator according to the procedures in this section.
- SIP - manually set to SIP for a specific telephone by an installer or administrator according to the procedures in this section.

Converting 9600 Series IP Telephones

An H.323 IP telephone can be either in use with possible customized settings or out of the box with factory default settings. An out of the box telephone you want to convert to SIP requires accessing the SIG Craft procedures early in the power up and initialization process and setting the Signaling Protocol Identifier (SIG) parameter for that telephone to “SIP.” Converting to SIP early avoids having to first load H.323 software, log in, and then invoke the “in use” process to load the SIP software.

Note:

For information about the SIG parameter, see “Choosing the Right Application File and Upgrade Script File” in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*. For information on setting or changing the SIG parameter, see Signaling Protocol Identifier on page 39.

Table 1: H.323 to SIP and SIP to H.323 Conversion Chart

Environment	To convert this type of telephone	To this type of telephone	Then:
SIP-centric	H.323 factory set	SIP	No action is required because the Signaling Protocol Identifier (SIG) defaults to SIP. Upon power-up & network connection, the telephone automatically downloads the proper SIP files.
SIP-centric	H.323 in use	SIP	Perform the **SIG** Craft procedure to change the SIG parameter value from “**1**" (H323) to “**default**” (SIP). For information, see Signaling Protocol Identifier on page 39. Save the SIG parameter change. Restart the telephone as covered in Restart the Telephone on page 39.
SIP-centric	SIP	H.323	Perform the **SIG** Craft procedure to change the SIG parameter value from “**default**” to “**1**" (H323). For information, see Signaling Protocol Identifier on page 39. Save the SIG parameter change. Restart the telephone as covered in Restart the Telephone on page 39.
H.323-centric	H.323 in use	SIP	Perform the **SIG** Craft procedure to change the SIG parameter value from “**default**” to “**2**" (SIP).” For information, see Signaling Protocol Identifier on page 39. Save the SIG parameter change. Restart the telephone as covered in Restart the Telephone on page 39.
H.323-centric	H.323 factory set	SIP	Connect the telephone to a power source and to the network. Press the **Program** softkey as soon as it displays in the first softkey position to access the **Craft Access Code Entry** screen. Perform the **SIG** Craft procedure and change the value from “**default**” to “**2**" (SIP). Save the SIG parameter change. Restart the telephone as covered in Restart the Telephone on page 39.
H.323-centric	SIP	H.323	Perform the **SIG** Craft procedure to change the SIG parameter value from “**2**" (SIP) to “**default**” (H323). For information, see Signaling Protocol Identifier on page 39. Save the SIG parameter change. Restart the telephone as covered in Restart the Telephone on page 39. Save the change & restart telephone.

Assembling the 9600 Series SIP IP Telephone

CAUTION:

Be careful to use the correct jack when plugging in the telephone. The jacks are located on the back of the telephone housing and are flanked by icons to represent their correct use.

Powering the 9600 Series IP Telephone

All 9600 Series SIP IP Telephones can be locally powered with a Telephone Power Module (DC power jack), available separately. In addition, the telephones support IEEE 802.3af-standard LAN-based power. Before installing a 9600 Series IP Telephone, verify with the LAN administrator whether the LAN supports IEEE 802.3af, and if so, whether the telephone should be powered locally or by means of the LAN.

Note:

The last step in assembling the 9600 Series SIP IP Telephone **must** be applying power. Apply power either by plugging the power cord into the power source (local powering) or plugging the modular line cord into the Ethernet wall jack (power by PoE).

CAUTION:

Failure to connect the proper cables with the proper jacks might result in an outage in parts of your network.

Figures 1 and 2 provide illustrations to connect cords to jacks on 9600 IP Series Telephones. Use the illustrations and associated procedures as appropriate for telephone assembly.

Telephone Model:	See:
9620	Figure 1
9630, 9640	Figure 2
9630G, 9640G	Figure 3

Figure 1: Connection Jacks on a 9620 Series SIP IP Telephone

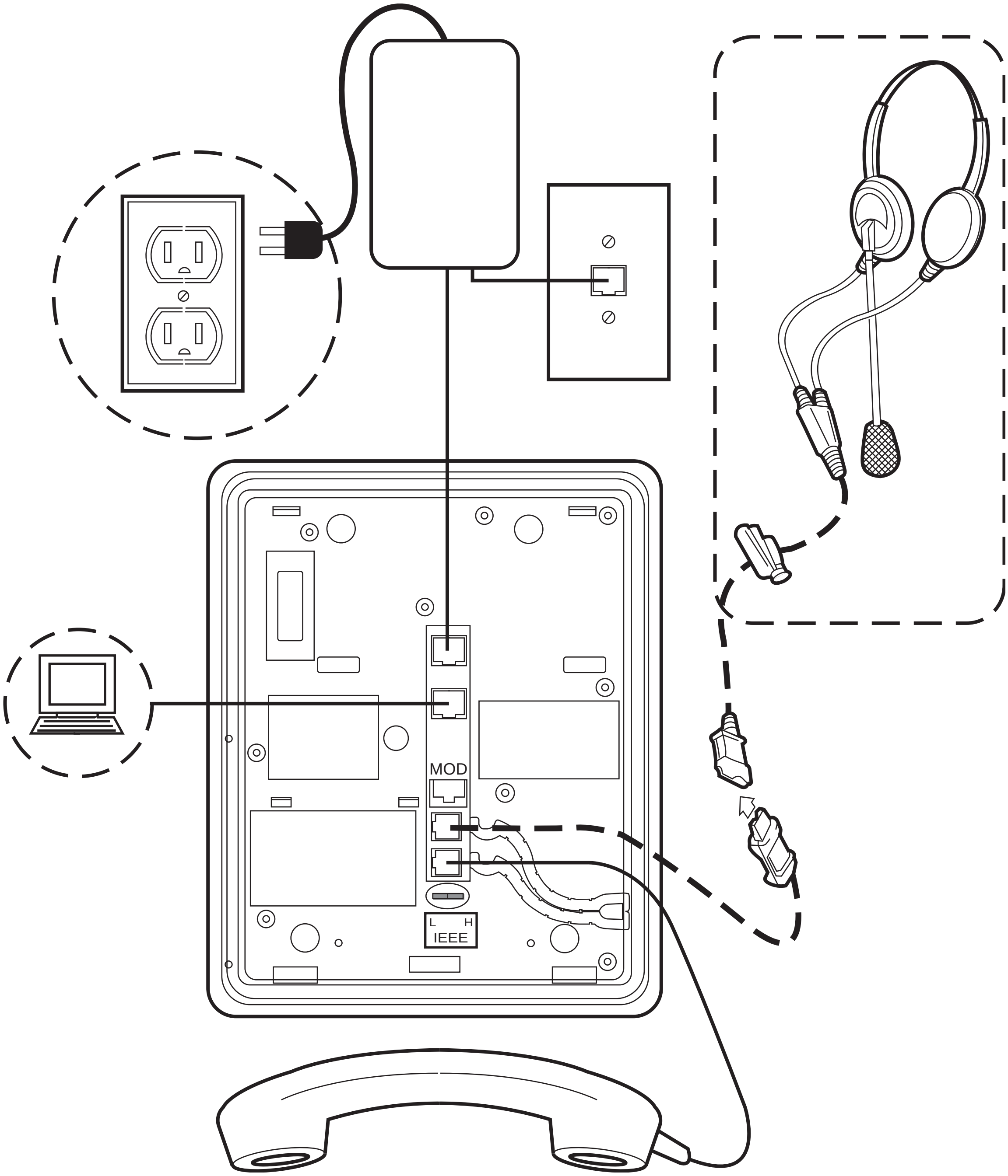


Figure 2: Connection Jacks on a 9630 & 9640 Series SIP IP Telephone

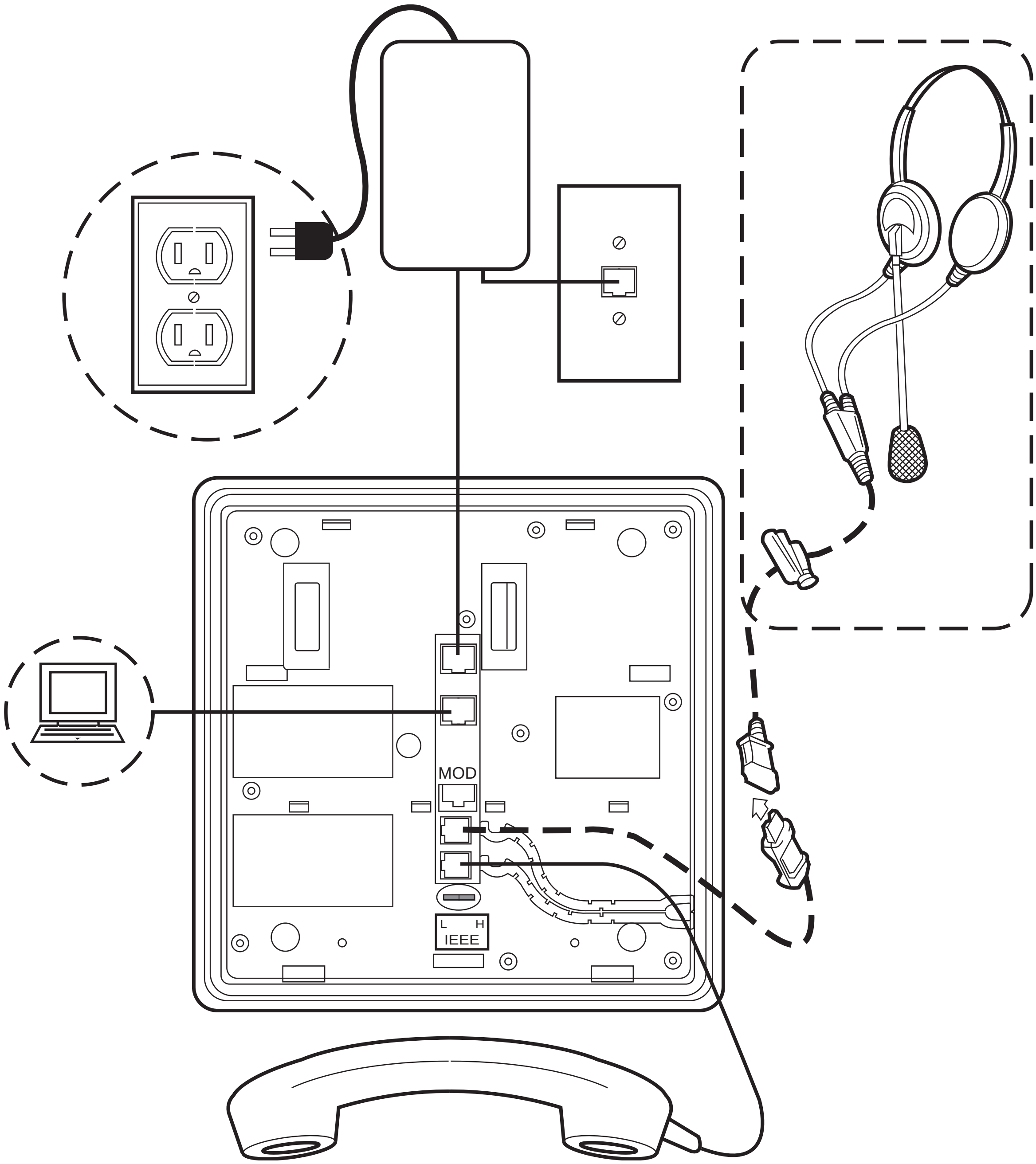
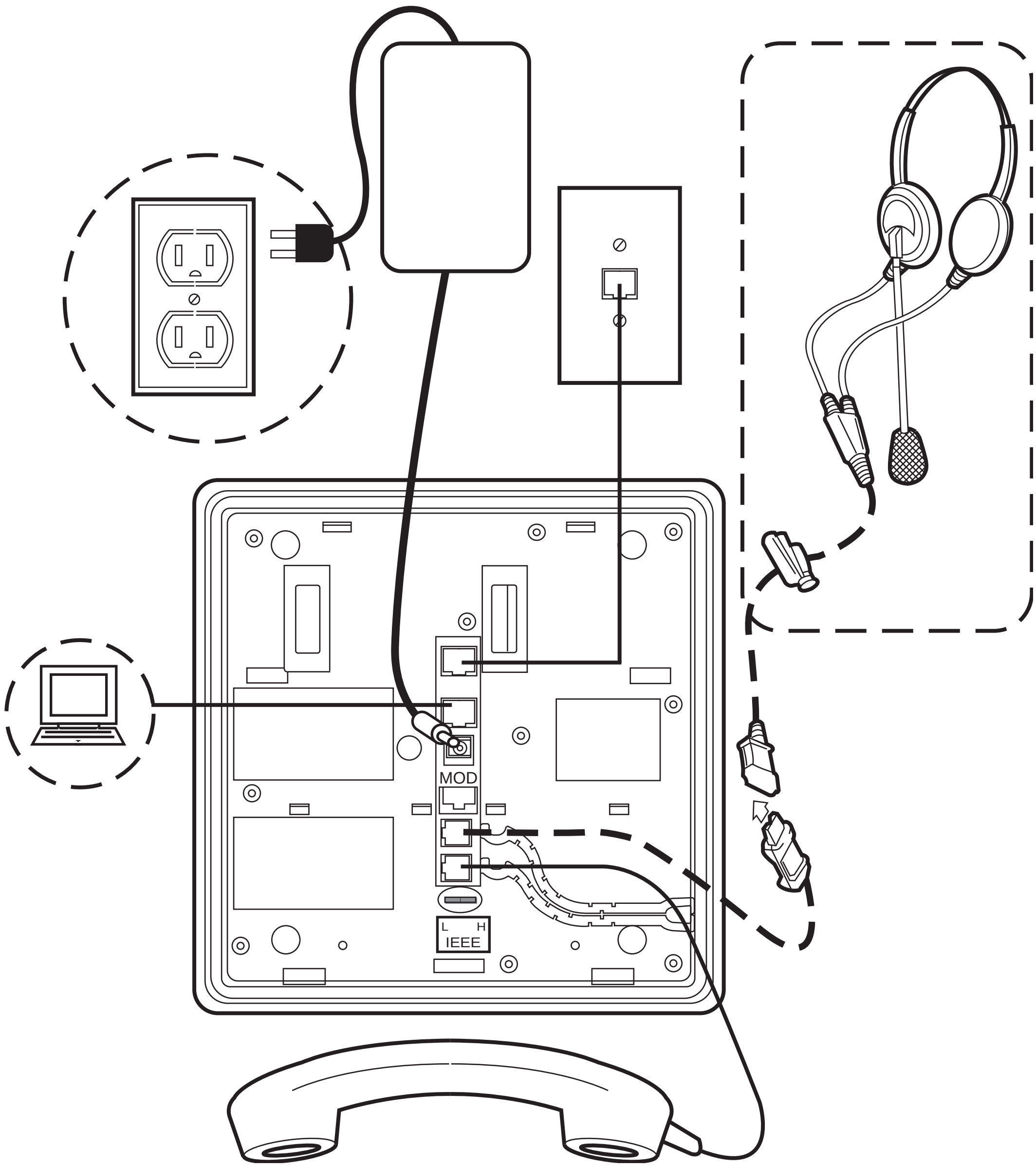


Figure 3: Connection Jacks on a 9630G & 9640G Series SIP IP Telephone



1. Plug one end of the H4DU 4-conductor coiled handset cord into the telephone and the other end into the handset.
2. Plug one end of the first Category 5e modular line cord into the Ethernet jack of the PC and the other end into the secondary Ethernet jack on the 9600 Series IP Telephone, if appropriate.
3. **If the telephone is to be PoE-powered**, plug one end of the second Category 5e modular line cord into the Ethernet jack on the 9600 Series SIP IP Telephone. Plug the other end of this cord into the Ethernet wall jack. If the telephone is to be PoE-powered, you are finished. Do not proceed to Step 4.
4. **If the telephone is to be powered locally**, connect one end of the second Category 5e modular line cord into the Ethernet jack on the 9600 Series SIP IP Telephone. Plug the other end of this cord into the 1151D power brick jack labeled **Phone**. Plug another Category 5e cable into the 1151D power brick jack labeled **Line**. Plug the other end of this cable into the Ethernet wall jack. Finally, connect the 1151D to an AC power source. You are now finished.

Dynamic Addressing Process/Telephone Startup

Important:

Before starting this process, read Converting Software on 9600 Series IP Telephones on page 12 to understand the requirements for converting factory-set H.323 telephones to SIP and make any changes necessary to suit your particular environment. Also, ensure that both Avaya Communication Manager (CM) and SIP Enablement Services (SES) are properly set up for your telephone environment.

Note:

Before starting this process you must have an OPTIM extension number for the SIP IP telephone, the Avaya Communication Manager security code (password), and a login and password on the SES server.

Any reference to the HTTP server applies equally to an HTTPS server.

The following description of the process of installing the SIP IP telephones assumes that the process is executed successfully. For errors that might be encountered during the process and the messages displayed, see Chapter 5: Troubleshooting Guidelines.

When you plug the IP telephone set into the Ethernet wall jack and apply power, if applicable, the following process takes place.

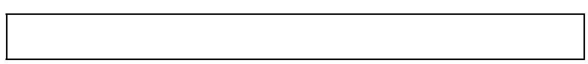
Note:

If the application has already been downloaded, the whole process takes approximately 1 to 2 minutes after the telephone is plugged in. For software upgrades, including the boot file and application file download, the process might take 5 - 10 minutes. The duration is based on LAN loading, how many telephones are being installed at once, and similar factors.

Do not unplug the power cord during the download process.

1. The telephone activates the Ethernet line interface, the PC Ethernet jack, and dial pad input to allow the invocation of procedures. The activation occurs as soon as possible after power-up or a reset.
2. During hardware initialization, configuration parameters are set to default values. The system initialization values for contrast and brightness are checked for non-null values, and set accordingly. The Avaya one-X™ name and logo display.
3. The system initialization value for the language file in use is checked for a non-null value, in which case the text strings in the language file named by that value are used for text display. Otherwise, English text strings are displayed.
4. The boot code checks for a primary software code image, loads it into volatile memory, and transfers control to it. If a primary software code image is not found, the boot code loads and transfers control to the backup software code image. Feedback displays in the form of a moving outline on the black squares below the logo. The outline moves from one square to the next to indicate processing is occurring.

   When storage of a new backup image begins, **Updating:** displays on the Title Line and **DO NOT UNPLUG THE PHONE!** displays on the Prompt Line until replaced by a subsequent message. In addition, a progress bar consisting of an unfilled black rectangle displays, centered on an Application Line below the logo image, as shown below.

   The rectangle fills from left-to-right as storage proceeds, with the filled percentage of the rectangle being approximately the same as the percentage of the file that has been stored.

Important:

Pressing the **Program** softkey at any time during startup invokes the Craft Access entry procedure to allow manual settings, but only if the PROCSTAT (local dialpad procedure status) system value is “0” providing full access to local procedures or if PROCSTAT is “1” in certain instances requiring input. For information, see Chapter 3: Local Administrative Options. If Craft procedures are invoked, the startup process terminates. The **Program** softkey also displays in conjunction with a message describing a processing conflict, for example, when an ARP response indicates a conflict in obtaining the IP Address.

5. The telephone displays the speed of the Ethernet interface in Mbps, that is, 10, 100, or 1000. The message No Ethernet displays until the software determines whether the interface is 10 Mbps, 100 Mbps, or 1000Mbps.

   Note:

   The Ethernet speed indicated is the LAN interface speed for both the telephone and any attached PC, assuming the administrator has not disabled the latter interface by a PHY2STAT setting.

6. The IP telephone sends a request to the DHCP server and invokes the DHCP process.

   The following message displays:

   ```
   DHCP: s secs
   ```

   where ***s*** is the number of seconds that have elapsed since DHCP was invoked.

7. VLAN verification and tagging occur. The following message displays:

   ```
   VLAN ID = n
   ```

   where ***n*** is the VLAN ID being used.

8. The DHCP server provides IP Addresses for the following hardware:
   - The IP telephone
   - The HTTP/HTTPS server
   - The SIP Proxy server

9. Using the list of IP Addresses provided by the DHCP server, the telephone performs a router check and verifies that the router is on the same subnet as the IP Address. The telephone cycles through the gateway IP Addresses with ARPs or pings until it receives a response.Using the list of gateway IP Addresses provided by the DHCP server, the telephone. When the router is located, received LLDP TLVs are processed. Then the HTTP process starts.

   Note:

   Any change in VLAN-related configuration parameters resulting from LLDP triggers a telephone reset.

10. The HTTP process starts with an `HTTP GET` command, which displays on the telephone's Title Line.

    Note:

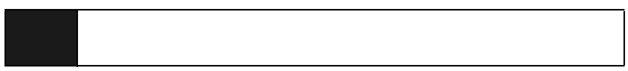
    Pressing the Program softkey at any time during startup invokes the Craft Access entry procedure to allow manual settings, but only if the PROCSTAT (local dialpad procedure status) system value is "0" providing full access to local procedures or if PROCSTAT is "1" in certain instances requiring input. For information, see Chapter 3: Local Administrative Options. If Craft procedures are invoked, the startup process terminates. The Program softkey also displays in conjunction with a message describing a processing conflict, for example, when an ARP response indicates a conflict in obtaining the IP Address.

11. When connected, the telephone looks for an upgrade script file.
12. The HTTP server sends and identifies an upgrade script, gets the settings file, the language file, and any firmware updates.

    The GET message might have to be sent several times. Each time the GET message is sent, the URI for the current HTTP request displays.
13. When the telephone determines that the application file received is valid, the following message displays:

    ```
    File Obtained;please wait...
    s secs
    ```

    where ***s*** is the number of elapsed seconds while non-volatile memory is erased.
14. While the application file is saved in flash memory, a progress bar shows the status:

15. The telephone checks for LLDP messages and re-checks VLAN status and tagging. If LLDP causes a change in the values of L2Q or L2QVLAN, a reset occurs to obtain a new IP Address.
16. If applicable, the telephone attempts to download a valid device certificate using simple certificate enrollment protocol (SCEP).

Simple Certificate Enrollment Protocol (SCEP)

1. When SCEP is initiated the telephone attempts to contact an SCEP server via HTTP, using the value of the configuration parameter MYCERTURL as the URI. The HTTP connection is established to the transport address specified by the value of the configuration parameter HTTPPROXY if HTTPPROXY is not null and if the configuration parameter HTTPEXCEPTIONDOMAINS is null, or• if HTTPEXCEPTIONDOMAINS is not null and the rightmost part of the domain portion of MYCERTURL does not match one of the values of HTTPEXCEPTIONDOMAINS. The values of the configuration parameters MYCERTKEYLEN, MYCERTCN, MYCERTDN are used in the certificate request.
2. While the telephone is attempting to contact the SCEP server and to obtain a certificate, the Title Line displays:

    ```
    SCEP: In progress...
    s secs
    ```

    where ***s*** is the number of seconds since SCEP was initiated/
3. If the initial attempt to contact the SCEP server is not successful the telephone continues with start-up, and will not try to contact the SCEP server again unless it is reset or power-cycled.

4. If a connection to the SCEP server is successfully established, if the value of the configuration parameter MYCERTWAIT is 1, SCEP remains in progress until the request for a certificate is granted or rejected.
5. If the request for a certificate is granted, ***SCEP: Successful*** displays on the Title line for at least one second, and remains until it is replaced by a subsequent display.
6. The SCEP server connection terminates, and the telephone continues with start-up. If the request for a certificate is rejected ***SCEP: Failed*** displays on the Title line for at least one second, and remains until it is replaced by a subsequent display. In this case, the SCEP server connection terminates and the telephone continues with startup.
7. if the value of the configuration parameter MYCERTWAIT is 0 (zero), SCEP remains in progress until the request for a certificate is granted or rejected or until a response is received indicating that the request is pending for manual approval. If the request for a certificate is granted or rejected, the same text will be displayed as specified above.
8. If a response is received indicating that the request is pending for manual approval, ***SCEP: Pending*** displays on the Title line for at least one second and remains until it is replaced by a subsequent display. The connection to the SCEP server is terminated, and the telephone continues with startup. The telephone periodically attempts to contact MYCERTURL as specified above (but in the background without displaying any message) until the request is granted or rejected.
9. If a device certificate and private key are successfully downloaded, they are saved in non-volatile memory along with the MYCERTURL value used to obtain them.
10. When the point in time is reached at which the percentage of the interval of time specified in the device certificate's Validity object corresponding to the value of the configuration parameter MYCERTRENEW has elapsed, the telephone periodically attempts to contact MYCERTURL as specified above (but in the background without displaying any message) to renew the certificate, until the renewal request is granted or rejected.

Registration and Login

1. Upon successful initialization and power-up, SIP IP telephones display the Login screen with the following prompt:

   ```
   Enter Username and Password
   ```

2. Enter the User Name/ID assigned to this telephone.
3. Enter the password and press **Login**.

   The extension is visible during entry but the password displays as asterisks. The system determines whether the extension is in use.
4. The telephone contacts PPM, logs in, and downloads the configuration file while displaying:

   ```
   Downloading configuration
   ```

5. SIP Software Release 2.0 and up supports visiting user capabilities. When the value of the VU_MODE configuration parameter is “1” (Optional), the Primary Phone field on the Login screen requires indication of whether this is the user’s primary telephone. Selecting “yes” causes the telephone to operate as a non-VU phone and the inactivity timer is not applied. Selecting “no” causes the telephone to operate in the Visiting User mode, where an inactivity timer will log the user off after a predetermined time. For more information about this feature, see the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.
6. Successful completion of this process produces a dial tone when the **Speaker** button is pressed or the handset is lifted.

   The SIP IP telephone was installed successfully.

Chapter 3: Local Administrative Options

Introduction

During installation or after you have successfully installed an IP telephone, you might be instructed to administer one of the manual procedures described in this chapter. These local administrative procedures are also referred to as Craft Procedures.

Note:

You can modify the settings file to set parameters for IP telephones that download their upgrade script and application files from the same HTTP server. See Chapter 4: Maintaining 9600 Series SIP IP Telephones and “9600 Series IP Telephone Scripts and Application Files” in Chapter 4 of the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.

CAUTION:

Only trained installers or technicians should perform local (craft) procedures. Perform these procedures **only** if instructed to do so by the system or LAN administrator.

Static administration of these options causes upgrades to work differently than if they are administered dynamically. Values assigned to options in static administration are not changed by upgrade scripts. These values remain stored in the telephone until either:

- a new boot file is downloaded, or
- the IP telephone is reset, as indicated in Reset System Values on page 38.

Use these option-setting procedures **only** with static addressing and, as always, only if instructed by the system or LAN administrator. Do **not** use these option-setting procedures if you are using DHCP. DHCP is the Dynamic Addressing Process, as indicated in Dynamic Addressing Process/Telephone Startup on page 19.

Accessing Local (Craft) Procedures

Note:

Local procedures can only be invoked when the local (dialpad) procedure status system value (PROCSTAT) is “0”, giving full access to local procedures.

Note:

The factory-set default Craft Access Code is **27238**.

During Telephone Startup:

1. During startup, invoke local procedures by pressing **Program** to display the Craft Access Command Entry screen:

```
Startup interrupted
Enter command
```

2. Press **Mute** and enter the local dialpad procedure password (0 to 7 numeric digits), as specified by the system administrator in the system value PROCPSWD. For security purposes, the telephone displays an asterisk for each numeric dialpad press.
3. Press **Enter** when password entry is complete.

   The entry is compared to the PROCPSWD value. If they match, the telephone displays the Admin Procedures screen, with the prompt "Select procedure and press Start."
4. Use the navigation arrows to scroll to and highlight the local procedure you want, then press **Select** or **OK**. Or scroll to the procedure you want and press the corresponding line button.

During Normal Telephone Operation:

1. Invoke all local procedures by pressing the **Mute** button, entering the local (dialpad) procedure password (0 to 7 numeric digits), then pressing the # button.

   A 6-second timeout is in effect between button presses after pressing the **Mute** button. If you do not press a valid button within 6 seconds of pressing the previous button, the collected digits are discarded. In this case, no administrative option is invoked.

   The entry is compared to the PROCPSWD value. If they match, the telephone displays the Admin Procedures screen, and prompts "Select procedure and press Start."
2. Use the navigation arrows to scroll to and highlight the local procedure you want, then press **Select** or **OK**. Or scroll to the procedure you want and press the corresponding line button.

Entering Data for Administrative Options

This section applies to all 9600 Series SIP IP Telephones and describes how to enter data for administrative options.

1. The first application line on any screen is automatically highlighted (selected) when the telephone displays the screen. To select the item on that line, press the appropriate softkey at the bottom of the screen, for example, **Change** or **Save**, or the **OK** button. To select a different line, use the down or up navigation arrows to change the line focus. When the desired line is highlighted, then press a softkey or **OK** to select that line.
2. Attempts to enter invalid data are rejected and the telephone emits an error beep.
3. If you enter a numeric digit that causes the IP Address or subnet mask value to exceed 255, or any value to exceed its maximum field value, an error beep tone sounds, the digit is ignored, and the cursor does not move forward.
4. If you enter a numeric digit for a value or for an IP Address or subnet mask field after entering only a zero, the new digit replaces the zero.
5. When you press the **Bksp** softkey to backspace, the most recently entered digit or period is erased from the display. The cursor remains in the erased character's former position.
6. The **More** softkey provides data entry options like symbols, all capital letters for text, numerals, etc.
7. Pressing **Back** or **Exit** exits the local procedures. If any changes were made using the ADDR procedure or if the Crafts Entry screen was invoked during startup, the telephone immediately resets. If no ADDR changes were made or if the local procedures were invoked post-startup, the telephone redisplays the screen (or other display) that was effective when the craft options was invoked.

   Note:

   If **PROCSTAT** has been administered to **1**, you will not be able to invoke any administrative options other than **VIEW**.

About Local Administrative Procedures

Craft procedures allow you to customize the 9600 Series IP Telephone installation for your specific operating environment on a telephone-by-telephone basis. This section provides a description of each local administrative option covered in this guide, with references to the pages on which the option appears.

Note:

Unless otherwise prohibited using administration, a user can view but not change most of the parameters associated with Craft procedures. For more information, see the applicable users guide(s).

Shown As	Craft Procedure Purpose	See
802.1X	Set the 802.1X operational mode	Setting the 802.1X Operational Mode on page 29.
ADDR...	Network Address information programming	Static Addressing Installation on page 30.
AGC...	Enable/disable Automatic Gain Control	Disable/Enable Automatic Gain Control on page 32.
CLEAR...	Clear all values to factory defaults	Clear Procedure on page 32.
DEBUG...	Enable/disable Debug Mode	Disable/Enable Debug Mode on page 34.
GROUP...	Set the Group Identifier	Group Identifier on page 35.
INT...	Network Interface Control	Interface Control on page 36.
LOG...	Enable/disable Event Logging	Disable/Enable Event Logging on page 37.
LOGOUT	Log off the telephone	Logout on page 38.
RESET VALUES	Reset system initialization values to defaults	Reset System Values on page 38.
RESTART PHONE	Restart the telephone	Restart the Telephone on page 39.
SIG...	Set the signaling protocol download flag	Signaling Protocol Identifier on page 39.
SIP...	Configure SIP call settings	Configuring SIP Settings on page 41.
SNTP...	Configure the time server settings	Configuring Time Server Settings on page 42.
SSON...	Set the Site-Specific Option Number	Site-Specific Option Number Setting on page 42.
VIEW	View current parameter values and file names	The View Administrative Option on page 43.

Setting the 802.1X Operational Mode

Important:

The DOT1X configuration parameter must be set to “0” or “1” for the telephone to to support 802.1X pass-thru and the DOT1XSTAT configuration parameter must be set to “1” or “2” for the telephone to support supplicant operation. For more information, see the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.

Use the following procedure to set or change the operational mode.

1. When you select 802.1X from the Craft Local Procedure Screen, the two settings shown represent the text strings associated with the current configuration parameter values of DOTIX (802.1X Pass-Thru Mode) and DOT1XSTAT (802.1X Supplicant Mode), defined as follows:

   For the Pass-thru mode:

   - “On” if DOT1X = 0
   - “Pass-thru mode + proxy logoff” if DOT1X = 1
   - “Off” if DOT1X = 2

   For the Supplicant:

   - “Off” if DOT1XSTAT = 0
   - “On” if DOT1XSTAT = 1
   - “On with multicast” if DOT1XSTAT = 2
2. To change the setting, select the line you want to change and press the **Change** softkey or the Right (or Left) navigation arrow to cycle through the settings.
3. Press **Save** to store the new setting and redisplay the Craft Local Procedure screen.

Pre-Installation Checklist for Static Addressing

Before performing static programming of address information, verify that all the requirements listed in the Verify These Network Requirements section of the Pre-Installation Checklist are met. You do not have to consider item 4 on page 12, as it refers to the DHCP server. In addition, you must have the values for the following parameters. Failure to do so can cause data entry errors that prevent the telephone from working. Such errors can also have a negative impact on your network. Print copies of this checklist for each subnet.

- ❑ **1.** The IP Address of the telephone.
- ❑ **2.** The IP Address of the router.
- ❑ **3.** The IP subnet mask.
- ❑ **4.** The IP Address of the HTTP and/or /HTTPS server.
- ❑ **5.** The IP Address of the DNS server.
- ❑ **6.** The VLAN ID (the L2QVLAN value).
- ❑ **7.** The VLANTEST value.

Static Addressing Installation

The usual way to assign IP Addresses to IP telephones is the automatic method described in Dynamic Addressing Process/Telephone Startup on page 19. There might be times, however, when manual assignment of IP Addresses is desired.

CAUTION:

Static addressing is necessary when a DHCP server is unavailable.

Because of the increased opportunities for text entry errors associated with static addressing, Avaya strongly recommends that a DHCP server be installed and static addressing avoided.

Use the following procedure to invoke manual address information programming.

1. When you select **ADDR** from the Admin Procedure Screen, the Static Addressing Local Procedure screen displays as follows with the prompt “Select address to change.”:

Static Addressing screen		Line Description and (Configuration Parameter Value)
Use DHCP	***Yes***	Yes or No (USE_DHCP)
Phone	***nnn.nnn.nnn.nnn***	Telephone IP Address (IPADD)
Router	***nnn.nnn.nnn.nnn***	Router in use; gateway/router IP Address(es) (ROUTER)
Mask	***nnn.nnn.nnn.nnn***	IP network mask (NETMASK)
HTTPS File Server	***nnn.nnn.nnn.nnn***	IP Address of HTTPS File Server (TLSSRVR)
HTTP File Server	***nnn.nnn.nnn.nnn***	IP Address of the HTTP File Server (HTTPSSRVR)
DNS Server	***nnn.nnn.nnn.nnn***	DNS server IP Address(es) (DNSSRVR)
802.1Q	***0=auto, 1=on, 2=off***	(L2Q)
VLAN ID	***dddd***	(L2QVLAN)
VLANTEST	***ddd***	Number of seconds to wait for a DHCP offer (VLANTEST)

   where:

   - ***nnn.nnn.nnn.nnn*** is the current IP Address associated with the specific address information to its left, which could be either a value previously set by a technician, or the original IP Address value if no previous change was made,
   - ***dddd*** is the current value of L2QVLAN and ***ddd*** is the current value of VLANTEST, respectively.

2. Use the navigation arrows to scroll to and highlight the address/item you want to change, then use the appropriate softkey(s) and the dialpad to change the value as described in Step 3.

3. Depending on the item you selected, choose one of the following:

If you want to	Then
Change any of the IP Address values (File, Phone, Router, Subnet Mask, &/or DNS Server)	Use the dialpad to enter the new IP Address. IP Addresses have three sets of three digits followed by a period. Pressing ***** following entry of three digits causes a period to be placed in the next position, and the cursor to advance one position to the right. For example, to enter the IP Address 111.222.333.444, press the 1 on the dialpad three times then press *****, press the **2** on the dialpad three times then press *****, press the 3 on the dialpad three times then press *****, then press the 4 on the dialpad three times. Proceed to the next step.

Change the VLAN ID value	Use the dialpad to enter the new static VLAN ID of from 0 to 4094, inclusive. Proceed to the next step.
Change the VLANTEST value	Use the dialpad to enter the new value of the **DHCPOFFER** wait period of from 0 to 999, inclusive. Proceed to the next step.

4. Press **Save** to store the new setting and redisplay the Admin Procedures screen or **Cancel** to return to the Admin Procedures screen without saving the value entered.

   Once the new values are stored, the telephone is reset.

   If a new boot program is downloaded from the HTTP/HTTPS server after you enter static addressing information, you must reenter your static addressing information.

Disable/Enable Automatic Gain Control

Use the following procedure to turn automatic gain control for the handset, headset, and/or the Speaker on or off.

1. When you select **AGC** from the Admin Procedures Screen, the following text displays:

Handset Auto Gain Control	***On***
Headset Auto Gain Control	***On***
Speaker Auto Gain Control	***On***

   where, the setting shown is the text string associated with the current system value of AGCHAND, AGCHEAD, or AGCSPKR, defined as:

   - “On” if the respective AGC*XXXX* system value is “1”.
   - Off if the respective AGC*XXXX* system value is “0”.

2. To change the setting, select (highlight) the appropriate line and press the **Change** softkey or the **Right** or **Left** navigation arrow to toggle the selected setting from On to Off or vice versa.
3. Press **Save** to store the new setting(s), update the associated system value(s), and redisplay the Admin Procedures screen.

Clear Procedure

Sometimes, you might want to remove *all* administered values, user-specified data, and option settings. Essentially, you want to return a telephone to its initial “clean slate” or out of the box condition. This is usually done following telephone repair or when passing a telephone to a new, dedicated user when the **LOGOUT** option is not sufficient. For example, a new user is assigned the same extension, but requires different permissions than the previous user.

The **Clear** option erases all administered data — static programming, file server and call server programming, and user settings, and restores all such data to default values. This option does not affect the software load itself. If you have upgraded the telephone, the telephone retains the latest software. Once you have cleared a telephone, you can administer it normally.

CAUTION:
This procedure erases all administered data, without any possibility of recovering the data.

Use the following procedure to clear the telephone of its administrative, user-assigned, and options values.

1. When you select **CLEAR** from the Admin Procedures Screen, the telephone displays a confirmation screen.
2. If you do not want to clear all values, press **No** to terminate the procedure and retain the current values. Press **Yes** to clear all values to their initial default values.

   The telephone displays the following text:

   ```
   Clearing values.
   ```

   The telephone is cleared to its “out of the box” state, resetting the following values to their factory defaults:

   - The 802.1X identity and password.
   - All system values and system initialization values.
   - User options, parameter settings, identifiers and password.
   - Any user data like Contact Lists or Call Logs are deleted.

   After clearing the values, the telephone resets.

Disable/Enable Debug Mode

Use the following procedure to turn the debug mode for the button module serial port on or off. 9600 Series IP Telephones running SIP Release 2.0 software do not support a button module.

CAUTION:
A DEBUG setting of “On” disables any adapter plugged into the MOD jack on the underside of the telephone.

1. When you select **DEBUG** from the Admin Procedures Screen, the following text displays:

Debug Mode	**On**

   where the setting shown is the text string associated with the current system value of DEBUG, defined as:

   - “On” if the DEBUG_ENABLED system value is “1”.
   - “Off” if the DEBUG_ENABLED system value is “0”.

2. Use the navigation arrows or press **Change** to toggle the selected setting from On to Off or vice versa.
3. Press **Save** to store the new setting.

   If the value changed from "On" to "Off", the telephone initiates a reset. If the value changed from "Off" to "On" the Admin Procedures screen redisplays.

   The telephone saves the new value.

Group Identifier

Use the following procedure to set or change the Group Identifier.

Note:

Perform this procedure only if the LAN Administrator instructs you to do so. For more information about groups, see The GROUP System Value on page 48.

1. When you select **GROUP** from the Admin Procedures Screen, the following text displays:

```
Setting:
```

   where the ***setting*** is the current system value of GROUP.

2. Enter a valid **Group** value (0-999).
3. Press **Save** to store the new setting and redisplay the Admin Procedures screen.

Interface Control

Use the following procedure to set or change the interface control value.

1. When you select **INT...** from the Admin Procedures Screen, the following text displays with a prompt to use the Right and Left navigation arrows to select a setting:

```
Ethernet:                     Choice Selector
PC Ethernet:                  Choice Selector
```

The values shown are the text strings associated with the current PHY1STAT on the Ethernet line and the current PHY2STAT system value on the PC Ethernet line.

The PHY1STAT text strings are:

- "Auto" when PHY1STAT = 1
- "10Mbps half" when PHY1STAT = 2
- "10Mbps full" when PHY1STAT = 3
- "100Mbps half" when PHY1STAT = 4
- “100Mbps full” when PHY1STAT = 5
- “1000 Mbps full” when PHY1STAT = 6

Note:

A PHY1STAT value of 6 applies only to telephone models that support Gigabit Ethernet (GigE), otherwise this value/choice does not display.

The PHY2STAT text strings are:

- "Disabled" when PHY2STAT = 0
- "Auto" when PHY2STAT = 1
- "10Mbps half" when PHY2STAT = 2
- "10Mbps full" when PHY2STAT = 3
- "100Mbps half" when PHY2STAT = 4
- “100Mbps full” when PHY2STAT = 5
- “1000Mbps full” when PHY2STAT = 6

Note:

A PHY2STAT value of 6 applies only to telephone models that support Gigabit Ethernet (GigE), otherwise this value/choice does not display.

2. To change the Ethernet setting, press the **Right** navigation arrow or the **Change** softkey to cycle through the possible settings.

   Depending on the current value, the next sequential text string is selected and displayed as the setting. For example, if the current value is 10Mbps half (2), pressing the Right navigation arrow changes the value to 10Mbps full (3). If the current value is 1000Mbps full (6), pressing the Right navigation arrow changes the value to Auto (1).

3. To change the PC Ethernet setting, select that line and press the Right navigation arrow or **Change** to cycle through the possible settings.
4. Press **Save** to store the new setting(s) and redisplay the Admin Procedures screen.

Disable/Enable Event Logging

Use the following procedure to enable or disable logging of system events.

1. When you select **LOG** from the Admin Procedures Screen, the telephone prompts you to use the Right and Left navigation arrows to select and change a setting and displays the following text:

Log:	**Choice Selector Bar for the SYSLOG_LEVEL value defined below.**
Log Categories...	**Use only when directed to do so by Avaya Services.**
Remote Logging	**on/off**
Remote Log Server	**nnn.nnn.nnn.nnn**

   where the ***text string*** is the wording associated with the current system value of SYSLOG_ENABLED (1 = Enabled; 0 = Disabled) and SYSLOG_LEVEL, defined as:

   - “Emergencies” when SYSLOG_LEVEL = 0
   - “Alerts” when SYSLOG_LEVEL = 1
   - “Critical” when SYSLOG_LEVEL = 2
   - “Errors” when SYSLOG_LEVEL = 3
   - “Warning” when SYSLOG_LEVEL = 4
   - “Notices” when SYSLOG_LEVEL = 5
   - “Information” when SYSLOG_LEVEL= 6
   - “Debug” when SYSLOG_LEVEL = 7

2. To change the **Log** or **Remote Logging Enabled** setting, press the Right (or Left) navigation arrow to cycle through the valid settings. When changing the Remote Log Server value, enter the IP Address to which syslog messages should be sent.

   When changing the **Log** value, depending on the current value, the next sequential text string or value is selected and displayed as the setting. For example, if the current value is Alerts (1), pressing the Right navigation arrow changes the value to Critical (2). If the current value is Debug (7), pressing the Right navigation arrow changes the value to Emergencies (0).

3. Press **Save** to store the new setting and redisplay the Admin Procedures screen.

Logout

Use the following procedure to log off a telephone.

CAUTION:
Once a telephone is logged off, a password and extension might be needed to log back on.

1. When you select **LOGOUT** from the Admin Procedures Screen, the telephone displays a confirmation screen asking if you are sure you want to log out.
2. Press **No** to return to the Admin Procedures screen without logging off the telephone. Press **Yes** to unregister the telephone from the call server.

Reset System Values

Use the following procedure to reset all system initialization values to the application software default values.

CAUTION:
This procedure erases all static information, without any possibility of recovering the data.

1. When you select **RESET VALUES** from the Admin Procedures Screen, the telephone displays a confirmation screen asking if you are sure you want to reset the telephone.
2. Press **No** to return to the Admin Procedures screen without resetting the telephone. Press **Yes** to start the telephone reset.

The telephone resets from the beginning of registration, which might take a few minutes. A reset:

- Resets all system values and system initialization values except AUTH and AUTH_ONLY to default values.
- Resets call server values to their defaults.
- Resets the 802.1X identity and password to their default values.
- Deletes any entries in the Redial buffer.
- Does not affect user-specified data and settings like Contacts data or the telephone login and password. To remove this type of data, see the Clear Procedure on page 32.

Restart the Telephone

Use the following procedure to restart the telephone.

1. When you select **RESTART PHONE** from the Admin Procedures Screen, the telephone displays a confirmation screen asking if you are sure you want to restart the telephone.
2. Press **No** to return to the Admin Procedures screen without restarting the telephone. Press **Yes** to proceed with the registration steps covered in the Dynamic Addressing Process/ Telephone Startup on page 19.

   A restart does not affect user-specified data and settings like Contacts data or the telephone login and password.

   The remainder of the restart procedure depends on the status of the boot and application files. For information, see Appendix A: Restart Scenarios.

Signaling Protocol Identifier

Use the following procedure to set or change the Signaling Protocol Identifier when your environment has more than one protocol on a subnet. A valid SIG Protocol Identifier is either **0** (default), **1** (H.323), or **2** (SIP).

Note:

Perform this procedure only if the LAN Administrator instructs you to do so.

1. When you select **SIG...** from the Admin Procedure Screen, the telephone prompts you to use the Right and Left navigation arrows to select a setting and displays the following text:

   ```
   Setting: text string          Choice Selector
   ```

   where the ***text string*** is the wording associated with the current system value of SIG, defined as:

   - "Default" when SIG = 0
   - "H.323" when SIG = 1
   - "SIP" when SIG = 2

   Note:

   The SIG value "Default" can represent either SIP or H.323 depending on the upgrade file used for the telephone.

2. To change the setting, press the **Change** softkey until you see the setting you want or use the **Right** or **Left** navigation arrow to cycle through the settings.

   Depending on the current value, the next sequential text string is selected and displayed as the setting. For example, if the current value is SIP (2), pressing the Right arrow changes the value to 0 (default). If the current value is H.323 (1), pressing Right arrow changes the value to 2 (SIP).

3. Press **Save** to store the new setting and redisplay the Admin Procedures screen.

   The remainder of this procedure depends on the status of the boot and application files. For information, see Appendix A: Restart Scenarios.

Configuring SIP Settings

Use this procedure to set up SIP-related settings like identifying the SIP Proxy Server.

1. When you select **SIP...** from the Admin Procedures Screen, the telephone prompts you to use the Right and Left navigation arrows to change a setting and displays the following settings and their active values:

Setting	Description/Example	Changes this Configuration Parameter
`SIP Mode:`	**Proxied** or **Peer-to-Peer**.	SIP_MODE
`SIP Domain:`	e.g., avaya.com	SIP_DOMAIN
`Avaya Environment:`	**Yes** or **No -** indicates whether only an Avaya environment (CM & SES) is in effect.	ENABLE_AVAYA_ ENVIRONMENT
`Transport Type:`	**TCP** or **TLS** or **UDP**.	SIPSIGNAL
`SIP Proxy Server:`	*IP Address*	SIPPROXYSRVR or SIPPROXYSRVR_IN_USE
`Avaya Config Server:`	IP Address of Avaya configuration server - only if PPM is not on the same server as the SIP Proxy server.	CONFIGURATION_SERVER or CONFIGURATION_SERVER_IN_USE
`User ID Field:`	Activates “1” /deactivates “0” User ID field on Login screen.	ENABLE_SIP_USER_ID

2. To change a setting:
   - Use the **Down** or **Up** Arrow indicator to move to the line you want to change.
   - If the field shows Right/Left Arrow indicators, press the **Change** softkey or the **Right** or **Left** navigation arrow to cycle through the settings. Depending on the current value, the next sequential value displays.
   - If the field does not show the Choice indicator, press the appropriate softkey(s) and/or use the dialpad to enter the new value.
3. Press **Save** to store the new setting(s) and redisplay the Admin Procedures screen.

Configuring Time Server Settings

Use this procedure to designate a server for Simple Network Time Protocol (SNTP) and to set corresponding values.

1. When you select **SNTP...** from the Admin Procedures Screen, the telephone displays the following settings and prompts you to enter the IP Address of the SNTP server:

	Description/Example	Changes this Parameter
`SNTP Server:`	IP address of the network time server.	SNTPSRVR or SNTPSRVR_IN_USE
`SNTP GMT Offset:`	Local time difference in hours from Greenwich Mean Time, e.g., NJ is **-5**.	GMTOFFSET
`SNTP Daylight Savings Time:`	Indicates whether the telephone should recognize Daylight Savings Time (DST)(0=no DST, 1=DST activated as per DSTOFFSET, 2=automatic based on DSTSTART and DSTSTOP values.	DAYLIGHT_SAVING_SETTING_MODE

2. To change a setting:
   - Use the **Down** or **Up** Arrow indicator to move to the line you want to change.
   - If the field shows Right/Left Arrow indicators, press the **Right** or **Left** navigation arrow to cycle through the available settings.
   - If the field does not show Arrow indicators, press the appropriate softkey(s) and/or use the dialpad to enter the new value.
3. Press **Save** to store the new setting and redisplay the Admin Procedures screen.

Site-Specific Option Number Setting

CAUTION:

Do **not** perform this procedure if you are using static addressing. Perform this procedure **only** if you are using DHCP **and** the LAN administrator instructs you to do this.

Use the following procedure to set the Site-Specific Option Number (SSON).

1. When you select **SSON...** from the Admin Procedures Screen, the following text displays:

   ```
   Setting:
   ```

   where the ***setting*** is the current system value of DHCP_SSON.
2. To change the setting, press the appropriate softkey(s) and use the dialpad to enter a valid SSON value between 128 and 255.
3. Press **Save** to store the new setting and redisplay the Admin Procedures screen.

The View Administrative Option

If you are using static addressing and encounter problems, use the following procedure to verify the current values of system parameters and file versions.

Note:

Unless otherwise prevented using administration, the user can view but not change most of the parameters associated with Craft Local Procedures. For more information about this option, see the applicable user guide(s).

Note:

If the View Network Information option is not available due to being disabled by administration, use the **ADDR** option to view IP Addresses. See Static Addressing Installation on page 30. The IP Addresses might have been entered incorrectly. Verify whether you were provided with correct IP Addresses.

1. When you select **VIEW** from the Admin Procedures Screen, the following text displays:

Setting	Description	Associated Configuration Parameter
`Model`	Telephone Model, e.g., 9630	MODEL
`Application File`	The “big app” filename	
`Boot File`	The “little app” filename	
`Group`	One to three digit GROUP value	GROUP
`MAC`	MAC Address	MACADDR
`SIP Proxy Server`		SIPPROXYSRVR_IN_USE
`Router`		ROUTER_IN_USE
`HTTPS file server`		TLSSRVR
`HTTP file server`		HTTPSRVR_IN_USE
`DNS server`		DNSSRVR_IN_USE
`SNTP server`		SNTPSRVR_IN_USE
`Protocol`	Signaling protocol in effect, e.g., SIP	
`Phone SN`	Telephone Serial Number	
`PWB SN`	Printed Wiring Board (circuit board) Serial Number (may not apply to all phones)	
`PWB Comcode`	Software-readable PWB serial number and comcode (may not apply to all phones)	PHONECC

2. Use the navigation arrows to scroll through the viewable information.
3. Press **Back** at any time to return to the Admin Procedures screen.

Chapter 4: Maintaining 9600 Series SIP IP Telephones

Introduction

This chapter covers maintaining the 9600 Series SIP IP Telephones, for example, downloading a new telephone software version from the Avaya support Web site. The recommended configuration is the latest call server software and the latest IP telephone firmware.

Important:

You can convert a 9600 Series IP Telephone from H.323 to SIP software, or from SIP to H.323 software. When converting from one protocol type to another on a given telephone, see Converting Software on 9600 Series IP Telephones on page 12. Note that, depending on the telephone model and the software version you start from, additional steps may be required from those mentioned in this section.

Downloading Software Upgrades

The files necessary to operate the 9600 Series IP Telephones are available on the Avaya Web site at: http://www.avaya.com/support. You must select one of two software “bundles” to download the latest software, depending on whether your telephone environment is primarily SIP-centric or H.323-centric.

Each bundle contains:

- An upgrade script file, **96xxupgrade.txt**, which allows you to upgrade to new software releases and new functionality without having to replace SIP IP telephones. The upgrade script tells the telephone whether a software upgrade is needed. All Avaya IP Telephones attempt to read this file whenever they reset. The upgrade script file is also used to point to the settings file. An "alternate" upgrade script is also included, designed for environments that will support both the H323 and SIP modes of operation. For such environments, the file needs to be edited in those sections having headings of “H.323 EDIT INSTRUCTIONS.” Specific instructions are provided in the Readme file that accompanies each software bundle. Once these changes are made, the alternate file should be renamed to “96xxupgrade.txt” and placed in the HTTP download directory. The HTTP download directory holds the telephone backup and application binaries the telephone will download. Renaming the alternate file causes any “96xxupgrade.txt” files residing in that directory to be overwritten.

Note:

The alternate 96xxupgrade.txt script file relies on the SIG setting as set in the local administrative (Craft) Signaling Protocol Identifier procedure being set to “SIP” for telephones destined to have SIP firmware loads, and “Default” or “H.323” for those telephones destined to have H.323 firmware loads.

- parameter settings and values that customize the telephones for your enterprise. One settings file is used for all your Avaya IP Telephones.
- Application files for all current 9600 Series SIP IP Telephones.
- Other useful information such as a ReadMe file.

In addition to the upgrade script, application files and Read Me file you need the latest binary code the Avaya SIP IP Telephones use, which is part of the software bundle you choose for your site. All these files are in a self-extracting executable file that comes in both zipped and unzipped format.

When the majority of your IP telephones are SIP-based, select the software bundle identified as “SIP” from the Web site. The application files in this SIP software bundle are the same as in the H.323 bundle. The difference is a modified upgrade script file that assumes SIP is the default protocol for your 9600 Series IP Telephones, and that H.323 is the exception. For more information on SIP-centric environments, see Converting Software on 9600 Series IP Telephones on page 12.

Download Procedure

The Avaya-provided upgrade script files and the binaries included in the zip files upgrade the Avaya IP Telephones. You should not need to modify them. It is essential that all the binary files be together on the file server. When downloading a new release onto a file server with an existing release already on it, we recommend that you:

- Stop the file server.
- Back up all the current file server directories as applicable.
- Copy your ***46xxsettings.txt*** file to a backup location.
- Remove all the files in the download directory. This ensures that you do not have an inappropriate binary or configuration file on the server. The only system values that can be used in the Conditional statement are: BOOTNAME, GROUP, and SIG.
- Download the self-extracting executable file, or the corresponding zip file.
- Extract all the files.
- Copy your ***46xxsettings.txt*** file back into the download directory.
- Check the Readme files for release-specific information.
- Modify the ***46xxsettings.txt*** file as desired.
- Restart the HTTP/HTTPS server.
- Reset your Avaya IP Telephones.

Updating the Settings File

After checking the application software, the 9600 Series IP Telephone looks for a 46xxsettings file. Another important maintenance activity might be to update the settings file for any changes to your customized settings. Checking the Read Me file can provide an indication of the impact of a software upgrade on your current settings.

Note:

You use one settings file for all your Avaya IP Telephones including the 9600 Series SIP IP Telephones covered in this document, 9600 Series IP Telephones (H.323 protocol, as covered in the *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Administrator Guide*), and 4600 Series IP Telephones, as covered in the *4600 Series IP Telephone LAN Administrator Guide* (Document 555-233-507).

The Avaya-provided upgrade script file includes lines that tell the telephone to **GET *46xxsettings.txt***. These lines cause the telephone to use HTTP/HTTPS to attempt to download the file specified in the `GET` command. If the file is obtained, its contents are interpreted as an additional script file. That is how your settings are changed from the default settings. If the file cannot be obtained, the telephone continues processing the upgrade script file. If the settings file is successfully obtained but does not include any setting changes the telephone stops using HTTP. This happens when you initially download the script file template from the Avaya support Web site, before you make any changes. When the settings file contains no setting changes, the telephone does not go back to the upgrade script file.

You can change the settings file name, if desired, as long as you also edit the corresponding `GET` command in the upgrade script file. However, we encourage you **not** to alter the Avaya-provided upgrade script file. If Avaya changes the upgrade script file in the future, any changes you have made will be lost. We strongly encourage you to use the ***46xxsettings*** file to customize your settings instead.

For detailed information about modifying the settings file, see Chapter 4 in the *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Administrator Guide*.

See Chapter 7 in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide* for details about specific values. You need only specify settings that vary from defaults, although specifying defaults is harmless.

Downloading Language Files

Language files must be stored in the same location as the 46xxsettings file or under the HTTP Server directory, defined using the **SET HTTPDIR [*HTTP server directory path*]** command.

You can download a new language file version only if the filename differs from the language file previously downloaded. Alternately, you can remove the old language file using an empty **SET**

LANGUAGES command in the 46xxsettings file before downloading a language file with the same filename.

Note:

Language files for 9600 Series SIP IP Telephones have a **.xml** filename extension whereas language files for 9600 Series IP Telephones set to H.323 have a .txt filename extension.

The GROUP System Value

You might have different communities of end users, all of which have the same model telephone, but which require different administered settings. For example, you might want to restrict Call Center agents from being able to log off, which might be an essential capability for "hot-desking" associates. We provide examples of the group settings for each of these situations later in this section.

The simplest way to separate groups of users is to associate each of them with a number. Use the GROUP system value for this purpose. The GROUP system value **cannot** be set in the 46xxsettings file. The GROUP system value can only be set on a telephone-by-telephone basis using a Craft procedure. To set up groups, first identify which telephones are associated with which group and designate a number for each group. The number can be any integer from 0 to 999, with 0 as the default, meaning your largest group would be assigned as Group 0.

Then, at each non-default telephone, invoke the **GROUP...** Local (dialpad) Administrative procedure as specified in Chapter 3: Local Administrative Options and specify which GROUP number to use. Once the GROUP assignments are in place, edit the configuration file to allow each telephone of the appropriate group to download its proper settings.

Here is an illustration of a possible settings file for the example of a Call Center with hot-desking associates at the same location:

```
IF $GROUP SEQ 1 goto GROUP1
IF $GROUP SEQ 2 goto GROUP2
{specify settings unique to Group 0}
goto END

# GROUP1
{specify settings unique to Group 1}
goto END

# GROUP2
{specify settings unique to Group 2}

# END
{specify settings common to all Groups}
```

Chapter 5: Troubleshooting Guidelines

Introduction

This chapter describes problems that might occur during both installation and normal operation of the 9600 Series SIP IP Telephones and possible ways of resolving these problems.

This chapter contains the following sections:

- Descriptions of error conditions and methods for resolving them.
- Error and status messages, and methods for resolving them.

Error Conditions

There are three areas where installers can troubleshoot problems before seeking assistance from the system or LAN administrator:

1. Check both the power and Ethernet wiring for the following conditions:
   - Whether all components are plugged in correctly.
   - Check LAN connectivity in both directions to all servers - DHCP, HTTP, HTTPS, Avaya Communication Manager, and/or SIP Proxy server.
   - If the telephone is supposed to be powered from the LAN, ensure that the LAN is properly administered and is compliant with IEEE 802.3af.
2. If you are using static addressing:
   - Use the **VIEW** Craft procedure to find the names of the files being used and verify that these filenames match those on the HTTP/HTTPS server. See The View Administrative Option on page 43 for more information. Check the Avaya Web site to verify whether the correct files are being used.
   - Use the **ADDR** Craft procedure to verify IP Addresses. See Static Addressing Installation on page 30 for information.
3. If the 9600 Series SIP IP Telephone is not communicating with the system (DHCP, HTTP, or CM call server), make a note of the last message displayed, as described in Table 2 and/or Table 3. Consult the system administrator.
4. If you expect the telephone to be PoE-powered, verify with the LAN administrator that PoE power is indeed supported on the LAN.

DTMF Tones

SIP telephones send DTMF tones according to the SEND_DTMF_TYPE parameter setting. The default setting of this parameter sends DTMF "tones" as "telephone event" RTP packets per RFC 2833. Whether a non-SIP telephone hears these DTMF tones depends on whether the Avaya Communication Manager media resource converts the "telephone event" RTP packets into audio RTP packets.

Power Interruption

If power to a 9600 Series SIP IP Telephone is interrupted while the telephone is saving the application file, the HTTP/HTTPS application can stop responding. If this occurs, restart the HTTP/HTTPS server.

Installation Error and Status Messages

The 9600 Series SIP IP Telephones issue messages in the currently selected language, or if the telephone is logged off, in the language specified by the SYSTEM_LANGUAGE parameter value. If English is not the selected language, the telephone displays messages in English only when they are associated with local procedures, for example, the **VIEW** Craft local procedure.

Most of the messages in Table 2 display only for about 30 seconds or less, and then the telephone resets. The most common exception is `Extension in Use`, which requires manual intervention.

Table 2: Possible Error and Status Messages During Installation of 9600 Series IP Telephones

Message	Cause/Resolution
`Address Conflict`	**CAUSE:** The telephone has detected an IP Address conflict. **RESOLUTION:** Verify administration to identify duplicate IP Address(es).
`Bad Router`	**CAUSE:** The telephone cannot find a router based on the information in the DHCP file. **RESOLUTION:** Use static addressing to specify a router address, or change administration on DHCP, as indicated in the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.
	1 of 2

Table 2: Possible Error and Status Messages During Installation of 9600 Series IP Telephones (continued)

Message	Cause/Resolution
`DHCP: CONFLICT`	**CAUSE:** At least one of the IP Addresses offered by the DHCP server conflicts with another address. **RESOLUTION:** Review DHCP server administration to identify duplicate IP Address(es).
`Finding router...`	**CAUSE:** The telephone is proceeding through boot-up. **RESOLUTION:** Allow the telephone to continue.
`No Ethernet`	**CAUSE:** When first plugged in (or during operation), the SIP IP telephone is unable to communicate with the Ethernet. **RESOLUTION:** Verify the connection to the Ethernet jack, verify the jack is Category 5, verify power is applied on the LAN to that jack, etc.
`Restarting...`	**CAUSE:** The telephone is in the initial stage of rebooting. **RESOLUTION:** Allow the telephone to continue.
`SCEP: Failed`	**CAUSE:** Simple Certificate Enrollment Protocol (SCEP) has rejected a request for a certificate. **RESOLUTION:** Although the SCEP server connection is terminated, startup continues. No action required.
`Subnet conflict`	**CAUSE:** The telephone is not on the same VLAN subnet as the router. **RESOLUTION:** Administer an IP Address on the telephone using Static Addressing Installation, or administer network equipment to administer the telephone appropriately.
`Updating: DO NOT UNPLUG THE TELEPHONE`	**CAUSE:** The telephone is updating its software image. **RESOLUTION:** Allow the telephone to continue.

2 of 2

Operational Errors and Status Messages

Table 3 identifies some of the possible operational problems that might be encountered after successful 9600 Series IP Telephone installation. The user guide for a specific telephone model also contains troubleshooting for users having problems with specific IP telephone applications. Most of the problems reported by 9600 Series IP Telephone users are not likely to be problems with the telephone itself. Problems are more likely LAN-based, where Quality of Service, server administration, and other issues can impact end-user perception of IP telephone performance.

Table 3: Operational Error Conditions for 9600 Series SIP IP Telephones

Condition		Cause/Resolution
During Craft procedure access, display freezes at prompt “Press # to program”		**CAUSE:** Craft access has failed; telephone cannot operate. **RESOLUTION:** Unplug the telephone, then plug it in again to reset.
After Login, the progress bar shows just a few completed bars and stops moving.		**CAUSE:** Login has failed. **RESOLUTION:** Check that the LAN and File servers are operating correctly. Re-attempt login.
The telephone continually reboots, or reboots continuously about every 15 minutes.		**CAUSE:** The telephone cannot find the call server. **RESOLUTION:** Ensure that SIPPROXYSRVR is administered either manually or through DHCP or HTTP, as appropriate. **CAUSE:** This might be a firmware fault because the MAC address in memory is corrupted. **RESOLUTION:** Return the telephone to Avaya for repair.
The message light on the telephone turns on and off intermittently, but the telephone never registers.		**CAUSE:** This is a hardware fault. **RESOLUTION:** The telephone must be returned to Avaya for repair.
The telephone stops working in the middle of a call,	**AND** no lights are lit on the telephone and the display is not lit.	**CAUSE:** Loss of power. **RESOLUTION:** Check the connections between the telephone, the power supply, and the power jack. For example, verify that either static addressing was not used or that any changes to static addresses were entered correctly.
	AND power to the telephone is fine (and the telephone might have gone through the restarting sequence).	**CAUSE:** Loss of path to the call server or the other party’s telephone, DHCP Lease expired, or DHCP server not available when telephone attempts to renegotiate DHCP lease. **RESOLUTION:** As above.

1 of 5

Table 3: Operational Error Conditions for 9600 Series SIP IP Telephones (continued)

Condition		Cause/Resolution
The telephone was working, but does not work now,	**AND** no lights are lit on the telephone and the display is not lit.	**CAUSE:** Loss of power. **RESOLUTION:** Check the connections between the telephone, the power supply, and the power jack.
	AND power to the telephone is fine, but there is no dial tone or the call appearances or feature buttons do not work.	**CAUSE:** Loss of communication with the call server. **RESOLUTION:** Check LAN continuity from the call server to the telephone using ARP or trace-route and from the telephone to the call server by invoking a Feature button. Verify that administration has not changed for the LAN equipment (routers, servers, etc.) between the call server and the telephone. Verify no one changed the telephone settings locally using the **VIEW** and **ADDR** craft procedures, as described earlier in this guide. Verify the telephone volume is set high enough.
	AND the telephone was recently moved.	**CAUSE:** Loss of communication with the call server. **RESOLUTION:** As above, but pay particular attention to the possibility that the telephone is being routed to a different DHCP server, or even a different proxy server. If so, the new server might need to be administered to support the telephone.
	AND the network was recently changed to upgrade or replace servers, re-administer the CM call server, SIP, add or change NAT, etc.	**CAUSE:** Loss of communication with SES. **RESOLUTION:** As above.

2 of 5

Table 3: Operational Error Conditions for 9600 Series SIP IP Telephones (continued)

Condition		Cause/Resolution
The telephone works, but the audio quality is poor, specifically:		
	the user hears echo when speaking on a handset.	**CAUSE:** Echo from digital-to-analog conversion on your CM call server trunk. **RESOLUTION:** Verify which trunk is causing the echo, and swap the trunk's Trunk Termination parameter on the call server.
	the user hears echo on a headset, but not on a handset.	**CAUSE:** Improper headset adapter. **RESOLUTION:** Replace adapter with Avaya's M12LU or 3412-HIC adapters. We recommend the M12LU, since it supports Automatic Gain Control.
	the user is on Speaker and hears no echo, but the far-end hears echo.	**CAUSE:** Room acoustics. **RESOLUTION:** Ensure that there are six inches or so of blank space to the right of the telephone. If that is insufficient, use the handset.
	the user experiences sudden silences such as gaps in speech, or static, clipped or garbled speech, etc.	**CAUSE:** Jitter, delay, dropped packets, etc. **RESOLUTION:** You can have the user provide diagnostic data by invoking the Network Information feature under the **A** (Avaya) button on the telephone. One or more Quality of Service (QoS) features should be implemented in the network as covered in Chapter 3: Local Administrative Options. **CAUSE:** Improper non-Category 5 wiring. **RESOLUTION:** Replace non-Category 5 wiring with Category 5 wiring.
	the user hears fluctuations in the volume level which are worse when the Speaker is on, or at the beginning of a call, or when a call goes from no one talking abruptly to a loud voice.	**CAUSE:** The user has changed the Automatic Gain Control (AGC) or environmental acoustics are not consistent with the current audio settings. **RESOLUTION:** Try different on/off settings for the AGCHAND, AGCHEAD, and AGCSPKR parameters.
The telephone works properly except for the Speaker.		**CAUSE:** The Speaker was disabled in the settings file. **RESOLUTION:** Check the settings file and re-enable the Speaker if appropriate.
The telephone works properly, except incoming DTMF tones are not received.		**CAUSE:** The TN2302AP board does not pass in-band DTMF tones. **RESOLUTION:** None; the board is operating as designed.

3 of 5

Table 3: Operational Error Conditions for 9600 Series SIP IP Telephones (continued)

<table>
<tr><th colspan="2">Condition</th><th>Cause/Resolution</th></tr>
<tr><td colspan="2">When a line is selected, a short dial tone burst sounds followed by a reorder/fast busy tone.</td><td>CAUSE: The extension is provisioned on SES and some CM forms, but not on the off-pbx-telephone station-mapping form. CM does not know to map back to SES and rejects the line reservation.
RESOLUTION: Map the extension on the off-pbx-telephone station-mapping form, a sample of which appears in Appendix C of the Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide.

CAUSE: Possible error in SIG group configuration on CM, which indicates the default region for the SIP trunk to Communication Manager.
RESOLUTION: On the IP-network-region form, ensure that the region pointed to is configured with an authoritative domain that is the same as the SES SIP domain. also verify that the station in question has not been redirected to a different network region on the ip-network map.</td></tr>
<tr><td colspan="2">The HTTP/HTTPS script file and settings file are ignored (not being used by the telephone).</td><td>CAUSE: The system value AUTH is set to 1 (HTTPS required) but no valid address is specified in TLSSRVR.
RESOLUTION: Change AUTH to 0 (zero), or enter a valid address for TLSSRVR.</td></tr>
<tr><td rowspan="2">The HTTP/HTTPS script file is ignored or not used by the telephone,</td><td>AND the HTTP/ HTTPS server is a LINUX or UNIX system.</td><td>CAUSE: UNIX and LINUX systems use case-sensitive addressing and file labels.
RESOLUTION: Verify the file names and path in the script file are accurately specified.</td></tr>
<tr><td>AND telephone administration recently changed.</td><td>CAUSE: The 96xxupgrade.txt file was edited incorrectly, renamed, etc.
RESOLUTION: Download a clean copy of the 96xxupgrade.txt file from the Avaya support Web site at http://www.avaya.com/support, and do not edit or rename it. Customize or change only the 46xxsettings file, as discussed in Chapter 4: Maintaining 9600 Series SIP IP Telephones.</td></tr>
<tr><td colspan="2">Some settings in the settings file are being ignored while other settings are being used properly.</td><td>CAUSE: Improper settings file administration.
RESOLUTION: Verify that customized settings are correctly spelled and formatted.</td></tr>
</table>

4 of 5

Table 3: Operational Error Conditions for 9600 Series SIP IP Telephones (continued)

Condition		Cause/Resolution
Some settings in the settings file are being ignored while other settings are being used properly,	**AND** the setting being ignored is one or more of the AGC settings.	**CAUSE:** The user changed the AGC setting(s). **RESOLUTION:** Have the user reset the AGC value(s) back to the desired setting(s).
	AND the setting being ignored is the TIMEFORMAT setting.	**CAUSE:** The user changed the time format using the Avaya Menu Options & Settings. **RESOLUTION:** None; the user wants the time set that way.
Telephone power is interrupted while the telephone is saving the application file **and** the HTTP/HTTPS application stops responding.		**CAUSE:** The HTTP/HTTPS server stops responding if power is interrupted while a telephone is saving the application file. **RESOLUTION:** Restart the HTTP/HTTPS server, as applicable.
The user indicates an application or option is not available.		**CAUSE:** The ***46xxsettings*** script file is not pointed to accurately, or is not properly administered to allow the application. **RESOLUTION:** Assuming the user is meant to have that application, verify the 46xxsettings script file is properly specified for your system, including case if your file server is UNIX or LINUX, and extension. Then, verify all the relevant parameters indicated in Table 7 of the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*, are accurately specified in the 46xxsettings file.
User data disappeared when the user logged off one telephone and logged into another telephone.		**CAUSE:** Possible PPM problem. **RESOLUTION:** Contact the SES administrator.

5 of 5

Appendix A: Restart Scenarios

Scenarios for the Restart Process

The sequence of the restart process depends on the status of the boot and application files. This appendix explains the different scenarios possible.

Note:

The file names used in this appendix are examples only. Your particular file names are likely to be different.

The procedures described in this appendix assume restarts occur during normal telephone operation rather than during telephone startup.

Restart the Telephone

Use the following procedure to restart the telephone.

1. Invoke RESTART by pressing the **Mute** button.
2. Enter the local (dialpad) procedure password (0 to 7 numeric digits), then press the # button.

   Note:

   A 6-second timeout is in effect between button presses. If you do not press a valid button within 6 seconds of pressing the previous button, the collected digits are discarded. In this case, no administrative option is invoked.

   The entry is compared to the PROCPSWD value. If they match, the telephone displays the Craft Local Procedure screen and prompts “Select procedure and press Start.”
3. Use the navigation arrows to scroll to and highlight **RESTART PHONE**, then press **Select** or **OK**. Or on a 9630 SIP IP Telephone, scroll to **RESTART PHONE** and press the corresponding line button.
4. When you select RESTART PHONE from the Admin Procedure Screen, the telephone displays a confirmation screen asking if you are sure you want to perform a restart.

5. Press **No** to return to the Craft Local Procedure screen without restarting the telephone. Press **Yes** to proceed with the registration steps covered in the Dynamic Addressing Process/Telephone Startup on page 19.

   A restart does not affect user-specified data and settings like Contacts data or the telephone login and password.

   The remainder of the restart procedure depends on the status of the boot and application files:

If this condition applies:	See:
Boot File Needs to be Upgraded	Boot File Needs to be Upgraded on page 58.
Latest Boot File Loaded/No Application File or Application File Needs to be Upgraded	Latest Boot File Loaded/No Application File or Application File Needs to be Upgraded on page 62.
Latest Boot File and System-Specific Application File Already Loaded	Latest Boot File and System-Specific Application File Already Loaded on page 63.

Boot File Needs to be Upgraded

The telephone automatically downloads an upgraded boot file if applicable.

1. The telephone activates the Ethernet line interface, the PC Ethernet jack, and dial pad input to allow the invocation of procedures. The activation occurs as soon as possible after power-up or a reset.
2. During hardware initialization, the system initialization values for contrast and brightness are checked for non-null values, and set accordingly. The Avaya one-X™ name and logo display on sets with bit-mapped displays.
3. The system initialization value for the language file in use is checked for a non-null value, in which case the text strings in the language file named by that value are used for text display. Otherwise, English text strings are displayed.
4. The boot code checks for a primary software code image, loads it into volatile memory, and transfers control to it. If a primary software code image is not found, the boot code loads and transfers control to the backup software code image. Feedback displays in the form of a moving outline on the black squares below the logo. The outline moves from one square to the next to indicate processing is occurring.
   When storage of a new backup image begins, **Updating:** displays on the Title Line and **DO NOT UNPLUG THE PHONE!** displays on the Prompt Line until replaced by a subsequent

message. In addition, a progress bar consisting of an unfilled black rectangle displays, centered on an Application Line below the logo image, as shown below.

The rectangle fills from left-to-right as storage proceeds, with the filled percentage of the rectangle being approximately the same as the percentage of the file that has been stored.

Note:

Holding down the **OK** button during Step 4 displays the software image filename.

5. When control is passed to the software that was just loaded, the following messages display:

```
Starting...
```

```
Updating boot code...
DO NOT UNPLUG THE PHONE!
```

This message continues while the new boot code is being written into RAM.

Important:

Pressing **Program** at this time invokes the Craft Access entry procedure to allow manual settings, but only if the PROCSTAT (local dialpad procedure status) system value is “0” providing full access to local procedures. For information, see Chapter 3: Local Administrative Options.

6. The telephone displays the speed of the Ethernet interface in Mbps, that is, 0, 10, 100, or 1000. The message `No Ethernet` displays until the software determines whether the interface is 10 Mbps, 100 Mbps, or 1000Mbps.

Note:

The Ethernet speed indicated is the LAN interface speed for both the telephone and any attached PC, assuming the administrator has not disabled the latter interface by a PHY2STAT setting.

7. The IP telephone sends a request to the DHCP server and invokes the DHCP process.

The following message displays:

```
DHCP: s secs
```

where ***s*** is the number of seconds that have elapsed since DHCP was invoked. The number of elapsed seconds is incremented once per second, until DHCP successfully completes.

8. VLAN verification and tagging occur. The following message displays:

```
VLAN ID = n
```

where ***n*** is the VLAN ID being used.

9. The DHCP server provides IP Addresses for the following hardware:
   - The IP telephone
   - The HTTP/HTTPS server
   - The SIP Proxy server
10. Using the list of IP Addresses provided by the DHCP server, the telephone performs a router check and verifies that the router is on the same subnet as the IP Address. The telephone cycles through the gateway IP Addresses with ARPs or pings until it receives a response.
11. The HTTP process starts with an HTTP GET command to obtain a URI for the HTTP server, which displays on the telephones' Title Line.

Important:
Pressing **Program** at this time invokes the Craft Access entry procedure to allow manual settings, but only if the PROCSTAT (local dialpad procedure status) system value is "0" giving full access to local procedures). For information, see Chapter 3: Local Administrative Options.

12. When connected, the telephone looks for an upgrade script file.

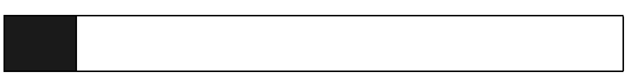
The HTTP server sends and identifies an upgrade script. Each time the GET message is sent, the URI for the current HTTP request displays.

13. When the telephone determines that the application file received is valid, the following message displays:

```
File Obtained;please wait...
s secs
```

where ***s*** is the number of elapsed seconds while non-volatile memory is erased.

14. While the application file is saved in flash memory, a progress bar shows the status:



15. If applicable, the telephone attempts to download a valid device certificate using simple certificate enrollment protocol (SCEP).

16. While the telephone is attempting to contact the SCEP server and to obtain a certificate, the Title Line displays:

```
SCEP: In progress...
s secs
```

17. where ***s*** is the number of seconds since SCEP was initiated. If the request for a certificate is granted, ***SCEP: Successful*** displays on the Title line for at least one second, and remains until it is replaced by a subsequent display.
18. If a device certificate and private key are successfully downloaded, they are saved in non-volatile memory along with the MYCERTURL value used to obtain them.
19. Upon successful initialization and power-up, SIP IP telephones display the Login screen with the following prompt:

```
Enter Username and Password
```

20. Enter the User Name/ID assigned to this telephone.
21. Enter the password and press **Login**.

    The extension is visible during entry but the password displays as asterisks. The system determines whether the extension is in use.
22. The telephone contacts PPM, logs in, and downloads the configuration file while displaying:

```
Downloading configuration
```

23. SIP Software Release 2.0 and up supports visiting user capabilities. When the value of the VU_MODE configuration parameter is “1” (Optional), the Primary Phone field on the Login screen requires indication of whether this is the user’s primary telephone. Selecting “yes” causes the telephone to operate as a non-VU phone and the inactivity timer is not applied. Selecting “no” causes the telephone to operate in the Visiting User mode, where an inactivity timer will log the user off after a predetermined time. For more information about this feature, see the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide*.
24. Successful completion of this process produces a dial tone when the **Speaker** button is pressed or the handset is lifted.
25. Continue with the next procedure.

Latest Boot File Loaded/No Application File or Application File Needs to be Upgraded

This procedure occurs with normal application file upgrades.

1. The telephone displays:

```
Restarting...
```

2. The telephone detects and displays the speed of the Ethernet interface in Mbps, that is, 0, 10, or 100. The message `No Ethernet` displays until the software determines whether the interface is 10 Mbps or 100 Mbps.

   Note:

   The Ethernet speed indicated is the LAN interface speed for both the telephone and any attached PC.

3. The software determines whether sufficient IP Address information was downloaded. In this scenario, it is discovered that sufficient information has **not** been downloaded. The following message displays while the DHCP process is invoked:

```
DHCP: 0 secs
* to program
```

   The number of elapsed seconds is incremented once per second, until DHCP successfully completes.

4. While the IP telephone establishes a TCP connection to the HTTP server, the telephone displays the following message:

```
HTTP:n ipadd
```

   where ***n*** is the number of the IP Address obtained from the HTTP server and ***ipadd*** is the IP Address.

5. The following message displays while the HTTP process is invoked:

```
HTTP: n uri
```

   The number increments once per second, until the HTTP server responds.

6. While the upgrade script file is being downloaded from the HTTP server, all IP telephones display the following message:

```
HTTP:n sc etag
```

   where ***n*** is the number of the IP Address obtained from the HTTP server, *sc* is the status code of the HTTP response and *etag* is the value of the ETag header.

7. The script file is processed. The software determines that the name of the boot code file in the telephone (BOOTNAME) is not the latest version. APPNAME is set to the name of an application file to replace the boot code. The following message displays while the application file is downloaded into RAM:

```
app_filename
n KB received
```

   where ***n*** is the number of KBs downloaded.

8. When the telephone determines that the application file received is valid, the following message displays:

```
File Obtained;please wait...
s secs
```

   where ***s*** is the number of seconds that elapse while non-volatile memory is erased.

9. The following message displays while the application file is stored in flash memory:

```
Saving to flash
n%, x secs
```

   where ***n*** is the percentage of the file that was stored, and ***x*** is the number of elapsed seconds. This usually takes longer than the file's download.

10. The telephone is reset so the new system-specific application file can be executed.
11. Continue with the next procedure.

Latest Boot File and System-Specific Application File Already Loaded

This happens with normal resets.

1. The telephone displays:

```
Restarting...
```

2. The telephone detects and displays the speed of the Ethernet interface in Mbps, that is, 0, 10, or 100. The message `No Ethernet` displays until the software determines whether the interface is 10 Mbps or 100 Mbps.

   Note:

   The Ethernet speed indicated is the LAN interface speed for both the telephone and any attached PC.

3. The software determines whether sufficient IP Address information was downloaded. In this scenario, it is discovered that sufficient information has **not** been downloaded. The following message displays while the DHCP process is invoked:

   ```
   DHCP: s secs
   ```

   where ***s*** is the number of elapsed seconds until DHCP successfully completes.

4. VLAN verification and tagging occur. The following message displays:

   ```
   VLAN ID = n
   ```

   where ***n*** is the VLAN ID being used.

5. The HTTP process starts with an HTTP GET command to obtain a URI for the HTTP server, which displays on the telephones' Title Line. When connected, the HTTP server sends and identifies an upgrade script.

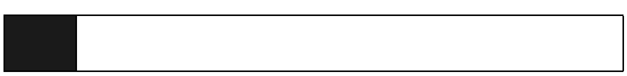
   The GET message might have to be sent several times. Each time the GET message is sent, the URI for the current HTTP request displays.

6. When the telephone determines that the application file received is valid, the following message displays:

   ```
   File Obtained;please wait...
   s secs
   ```

   where ***s*** is the number of elapsed seconds while non-volatile memory is erased.

7. While the application file is saved in flash memory, a progress bar shows the status:

8. The script file is processed. The software determines that the name of the boot code file in the telephone (BOOTNAME) is the latest version, and the name of the application file in the telephone is the same as APPNAME.

9. System-specific registration with the call server is invoked.

10. When registration finishes, a dial tone is available on the telephone.

Appendix B: Glossary of Terms

Terms Used in This Guide

802.1D 802.1Q	802.1Q defines a layer 2 frame structure that supports VLAN identification and a QoS mechanism usually referred to as 802.1D.
802.1X	Authentication method for a protocol requiring a network device to authenticate with a back-end Authentication Server before gaining network access. SIP Software Release 2.0 and up supports IEEE 802.1X using EAP-MD5 and EAP-TLS authentication methods.
Application - specific	Specific to a particular "application" running inside the telephone. For example, configuration file downloading, HTTP push, and the Web browser are all internal applications that use the HTTP protocol. Similarly, the RTCP and CNA clients are internal applications that can invoke traceroute. This term does not include Web-page-based "applications" rendered in the Web browser.
ARP	Address Resolution Protocol, used, for example, to verify that the IP Address provided by the DHCP server is not in use by another IP telephone.
Call Server	In an Avaya SIP environment, the "call server" is the combination of SIP Enablement Services (SES) and Avaya Communication Manager.
CLAN	Control LAN, type of Gatekeeper circuit pack.
CNA	Converged Network Analyzer.
DHCP	Dynamic Host Configuration Protocol, an IETF protocol used to automate IP Address allocation and management.
DiffServ	Differentiated Services, an IP-based QoS mechanism.
DNS	Domain Name System, an IETF standard for ASCII strings to represent IP addresses. The Domain Name System (DNS) is a distributed Internet directory service. DNS is used mostly to translate between domain names and IP Addresses. Avaya 9600 Series SIP IP Telephones can use DNS to resolve names into IP Addresses. In DHCP, TFTP, and HTTP files, DNS names can be used wherever IP Addresses were available as long as a valid DNS server is identified first.

EAP	Extensible Authentication Protocol, or EAP, a universal authentication framework frequently used in wireless networks and Point-to-Point connections defined by RFC 3748. EAP provides some common functions and a negotiation of the desired authentication methods, two of which are EAP-MD5 and EAP-TLS. When EAP is invoked by an 802.1X enabled NAS (Network Access Server) device such as an 802.11 a/b/g Wireless Access Point, modern EAP methods provide a secure authentication mechanism and negotiate a secure PMK (Pair-wise Master Key) between the client and the NAS.
Gatekeeper	H.323 application that performs essential control, administrative, and managerial functions in the media server. Sometimes called CLAN in Avaya documents.
H.323	A TCP/IP-based protocol for VoIP signaling.
HIS	An Avaya headset cord. This is the only headset cord supported by the Avaya 9600 SIP IP telephones.
HTTP	Hypertext Transfer Protocol, used to request and transmit pages on the World Wide Web.
HTTPS	A secure version of HTTP.
IEEE 802.3af	A Local Area Network protocol suite commonly known as Ethernet. PoE is usually implemented through this protocol.
IETF	Internet Engineering Task Force, the organization that produces standards for communications on the internet.
LAN	Local Area Network.
LLDP	Link Layer Discovery Protocol. All IP telephones with an Ethernet interface support the transmission and reception of LLDP frames on the Ethernet line interface in accordance with IEEE standard 802.1AB. SIP Software Release 2.0 and up supports LLDP.
MAC	Media Access Control, ID of an endpoint.
PoE	Power Over Ethernet. A system for transmitting transmit electrical power and data to remote devices over standard twisted-pair cable in an Ethernet network.
PPM	Personal Profile Manager, part of the SIP Enablement Services (SES) platform. PPM is responsible for maintaining and managing end users' personal information in the system.
QoS	Quality of Service, used to refer to several mechanisms intended to improve audio quality over packet-based networks.
RTCP	Real-time Transport Control Protocol.
RTP	Real-time Transport Protocol.
SCEP	Simple Certificate Enrollment Protocol, used to obtain a digital certificate.
SES	SIP Enablement Services.

SIP	Session Initiation Protocol, an open standard defined initially by IETF RFC 3261. SIP is an alternative to H.323 for VoIP signaling, both of which 9600 Series IP Telephones support.
SRTCP	Secure Real-time Transport Control Protocol.
SRTP	Secure Real-time Transport Protocol.
System - specific	Specific to a particular type of call server, for example, Avaya Communication Manager (CM). “System-specific signaling” refers to messages specific to the signaling protocol used by the system, for example, H.323 and/or CCMS messages used by CM and IP Office. “System-specific procedures” refers to telephone software procedures that are specific to the call server with which the software is intended to be used.
TCP	Transmission Control Protocol, a connection-oriented transport-layer protocol.
TLS	Transport Layer Security, an enhancement of Secure Sockets Layer (SSL). TLS is compatible with SSL 3.0 and allows for privacy and data integrity between two communicating applications.
UDP	User Datagram Protocol, a connectionless transport-layer protocol.
Unnamed Registration	Registration with Avaya Communication Manager by an IP telephone with no extension. Allows limited outgoing calling.
URI & URL	Uniform Resource Identifier and Uniform Resource Locator. Names for the strings used to reference resources on the Internet (for example, HTTP://....). URI is the newer term.
VLAN	Virtual LAN.
VoIP	Voice over IP, a class of technology for sending audio data and signaling over LANs.

Appendix C: Related Documentation

IETF Documents

IETF documents provide standards relevant to IP Telephony and are available for free from the IETF Web site: http://www.ietf.org/rfc.html.

ITU Documents

Access the ITU Web site for more information about ITU guidelines and documents, available for a fee from the ITU Web site: http://www.itu.int.

ISO/IEC, ANSI/IEEE Documents

Access the ISO/IEC standards Web site for more information about IP Telephony standards, guidelines, and published documents: http://www.iec.ch.

Index

Numerical

A

B

C

D

E

G

H

I

L