Installing and Administering Avaya J100 Series IP Phone

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Note

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

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Australia Statements

Handset Magnets Statement:

⚠ **Danger:**

The handset receiver contains magnetic devices that can attract small metallic objects. Care should be taken to avoid personal injury.

Industry Canada (IC) Statements

RSS Standards Statement

This device complies with Industry Canada licence-exempt RSS standard(s). Operation is subject to the following two conditions:

1. This device may not cause interference, and
2. This device must accept any interference, including interference that may cause undesired operation of the device.

Le présent appareil est conforme aux CNR d'Industrie Canada applicables aux appareils radio exempts de licence. L'exploitation est autorisée aux deux conditions suivantes:

1. L'appareil ne doit pas produire de brouillage, et
2. L'utilisateur de l'appareil doit accepter tout brouillage radioélectrique subi, même si le brouillage est susceptible d'en compromettre le fonctionnement.

Radio Transmitter Statement

Under Industry Canada regulations, this radio transmitter may only operate using an antenna of a type and maximum (or lesser) gain approved for the transmitter by Industry Canada. To reduce potential radio interference to other users, the antenna type and its gain should be so chosen that the equivalent isotropically radiated power (EIRP) is not more than that necessary for successful communication.

Conformément à la réglementation d'Industrie Canada, le présent émetteur radio peut fonctionner avec une antenne d'un type et d'un gain maximal (ou inférieur) approuvé pour l'émetteur par Industrie Canada. Dans le but de réduire les risques de brouillage radioélectrique à l'intention des autres utilisateurs, il faut choisir le type d'antenne et son gain de sorte que la puissance isotrope rayonnée équivalente ne dépasse pas l'intensité nécessaire à l'établissement d'une communication satisfaisante.

This Class B digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada.

Radiation Exposure Statement

This equipment complies with FCC & IC RSS102 radiation exposure limits set forth for an uncontrolled environment. This equipment should be installed and operated with minimum distance 20cm between the radiator & your body. This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.

Cet équipement est conforme aux limites d'exposition aux rayonnements ISEDétablies pour un environnement non contrôlé. Cet équipement doit être installé et utilisé avec un minimum de 20 cm de distance entre la source de rayonnement et votre corps.

Industry Canada (IC) Statements

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Cet appareil numérique de la classe B est conformeà la norme NMB-003 du Canada.

Japan Statements

Class B Statement

This is a Class B product based on the standard of the VCCI Council. If this is used near a radio or television receiver in a domestic environment, it may cause radio interference. Install and use the equipment according to the instruction manual.

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取扱説明書に従って正しい取り扱いをして下さい。　VCCI－B

Denan Power Cord Statement

Danger:

Please be careful of the following while installing the equipment:

- Please only use the connecting cables, power cord, and AC adapters shipped with the equipment or specified by Avaya to be used with the equipment. If you use any other equipment, it may cause failures, malfunctioning, or fire.
- Power cords shipped with this equipment must not be used with any other equipment. In case the above guidelines are not followed, it may lead to death or severe injury.

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México Statement

The operation of this equipment is subject to the following two conditions:

1. It is possible that this equipment or device may not cause harmful interference, and
2. This equipment or device must accept any interference, including interference that may cause undesired operation.

La operación de este equipo está sujeta a las siguientes dos condiciones:

1. Es posible que este equipo o dispositivo no cause interferencia perjudicial y
2. Este equipo o dispositivo debe aceptar cualquier interferencia, incluyendo la que pueda causar su operación no deseada.

Power over Ethernet (PoE) Statement

This equipment must be connected to PoE networks without routing to the outside plant.

U.S. Federal Communications Commission (FCC) Statements

Compliance Statement

The changes or modifications not expressly approved by the party responsible for compliance could void the user’s authority to operate the equipment.

To comply with the FCC RF exposure compliance requirements, this device and its antenna must not be co-located or operating to conjunction with any other antenna or transmitter.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interferences that may cause undesired operation.

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designated to provide reasonable protection against harmful interferences in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interferences to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.

- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Radiation Exposure Statement

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment . This equipment should be installed and operated with minimum distance of 8 in or 20 cm between the radiator and your body. This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.

Energy star compliance statement

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WiFi transmitter

- Frequencies for 2412-2472 MHz, transmit power: 17.8 dBm
- Frequencies for 5180-5240 MHz, transmit power: 19.14 dBm

General Safety Warning

- Use only the Avaya approved Limited Power Source power supplies specified for this product.
- Ensure that you:
    - Do not operate the device near water.
    - Do not use the device during a lightning storm.
    - Do not report a gas leak while in the vicinity of the leak.
    - For Accessory Power Supply – Use Only Limited Power Supply Phihong Technology Co. Ltd. Model: PSAC12R-050, Output: 5VDC, 2.4A.

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Contents

Chapter 1: Introduction

Purpose

This document contains information about preparing Avaya J100 Series IP Phones for installation, deployment, initial administration, and administration tasks including data and security.

This document is intended for the deployment engineers or support personnel who install, administer, and maintain Avaya J100 Series IP Phones.

The deployment engineers or the support personnel must have the following knowledge, skills, and tools:

Knowledge

- DHCP
- SIP
- Installing and configuring Avaya Aura® components
- Installing and configuring IP Office components
- Configuring 802.1x and VLAN

Skills

How to administer and configure:

- Avaya Aura® Session Manager
- Avaya Aura® Communication Manager
- Avaya Aura® Presence Services
- Avaya Aura® Session Border Controller
- IP Office
- DHCP server
- HTTP or HTTPS server
- Microsoft Exchange Server

Tools

- Avaya Aura® System Manager
- IP Office Manager
- IP Office Web Manager

Chapter 2: J100 Series IP Phone overview

The J100 Series IP Phone is a series of phones that you can use for unified communication. The series leverages the enterprise IP network and eliminates the need of a separate voice network. It offers superior audio quality and customizability with low power requirements in a Session Initiation Protocol (SIP) environment.

With this phone, you can:

- Make conference calls more efficiently and enhance customer interactions with high-quality audio.
- Gain access to information quickly through easy-to-read and high-resolution displays.
- Create a survivable, scalable infrastructure that delivers reliable performance and flexible growth as business needs change.
- Increase performance by deploying Gigabit Ethernet within your infrastructure.
- Reduce energy costs by using efficient Power-over-Ethernet (PoE) including sleep mode, which lowers energy consumption significantly.

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Related links

J100 Series IP Phone models on page 10
Hardware on page 11
Power specifications on page 12
Supported codecs on page 12

J100 Series IP Phone models

Phone model	Description
J129 IP Phone	A SIP-based phone with a monochrome display that supports single line call appearance.
J169 IP Phone	A SIP-based phone with a grayscale display that supports eight call appearances with four lines of call display. The phone can also support up to three button modules each supporting 24 application lines.

Table continues…

Phone model	Description
J179 IP Phone	A SIP-based phone with a color display that supports eight call appearances with four lines of call display. The phone can also support up to three button modules each supporting 24 application lines.

Related links

J100 Series IP Phone overview on page 10

Hardware

Avaya J100 Series IP Phones supports the following specifications:

Standard	J129	J169	J179	JBM24
Call appearances	1	8	8	N/A
Wall mountable	Yes	Yes	Yes	Yes
Stand	Dual position	Dual position	Dual position	Dual position
Touch screen	N/A	N/A	N/A	N/A
Display type	Monochrome	Grayscale	Colored	Grayscale
Display	2.3”, 128 x 32 pixel	3.5”, 320 x 240 pixel	3.5”, 320 x 240 pixel	
Dual color call indicator	0	8	8	24
Ethernet switch	Dual 10/100	Dual 10/100/1000	Dual 10/100/1000	N/A
WLAN support	No	No	Yes (As an optional module)	N/A
Softkeys call control	3	4	4	N/A
BT support	No	No	Optional module	N/A
Wired Handset	Yes	Yes	Yes	N/A
Wired Headset	No	Yes	Yes	N/A
Expansion module capable	No	Yes (3)	Yes (3)	N/A
Optional DC Power	No	Yes	Yes	N/A
GSPPOE power adapter	Yes	Yes	Yes	

Related links

J100 Series IP Phone overview on page 10

Power specifications

The J100 Series IP Phones can use different power sources like LAN based Power, the Global Single Port PoE Injector (GSPPOE) or power module (DC power jack).

The following table lists the various power requirements with or without peripherals.

Device	Power requirement
J129 IP Phone	• IEEE 802.3af • GSPPOE - Avaya 48V PoE power inserter (Optional Component)
J169 IP Phone	• IEEE 802.3af POE (Class 1) without JBM 24 button module • 802.3af PoE (Class 2) if using any JBM24 button module • GSPPOE - Avaya 48V PoE power inserter (Optional Component) • 5V DC Power adapter with barrel jack (Optional Component)
J179 IP Phone	• IEEE 802.3af POE (Class 1) without JBM 24 button or wireless module • 802.3af PoE (Class 2) if using any JBM24 button module or wireless module • GSPPOE - Avaya 48V PoE power inserter (Optional Component) • 5V DC Power adapter with barrel jack (Optional Component) **Note:** Power the phone with GSPPOE or 5V DC power adapter if the JBM 24 button module and the wireless module are in use simultaneously.

Related links

J100 Series IP Phone overview on page 10

Supported codecs

Avaya J100 Series IP Phones supports the following codecs and call control protocol:

Codecs	J129	J169	J179
Call control protocol	SIP	SIP	SIP
Codecs	• G.711a • G.711µ • G.729 • G.729a • G.729ab • G.726 • Opus	• G.711a • G.711µ • G.729 • G.729a • G.729ab • G.726 • Opus	• G.711a • G.711µ • G.729 • G.729a • G.729ab • G.726 • Opus

Table continues…

Codecs	J129	J169	J179
	• G722	• G722	• G722

Related links

J100 Series IP Phone overview on page 10

Chapter 3: Phone installation

Hardware setup

Wi-Fi overview

The Wi-Fi module enables the phone to connect to a network through a wireless network. If the phone loses connection to one Wi-Fi network, it continues to operate with another redundantly configured wireless network or Ethernet network. A Wi-Fi status icon displays when Wi-Fi is in use. If the phone is connected to Ethernet switch and the Ethernet link goes down, a pop-up message displays to change network connectivity to Wi-Fi.

You can configure Wi-Fi network by :

- Setting Wi-Fi parameters by using the Settings file
- Configuring Wi-Fi from the phone UI
- Configuring Wi-Fi parameters from the web UI

Note that VLAN and LLDP functionalities are not supported over a wireless network.

J100 wireless module

Avaya J129 IP Phone and Avaya J179 IP Phone support wireless module. The wireless module is an optional component and you can order this module separately.

✱ **Note:**

Avaya J169 IP Phone do not support the J100 wireless module.

Installing the Wireless Module

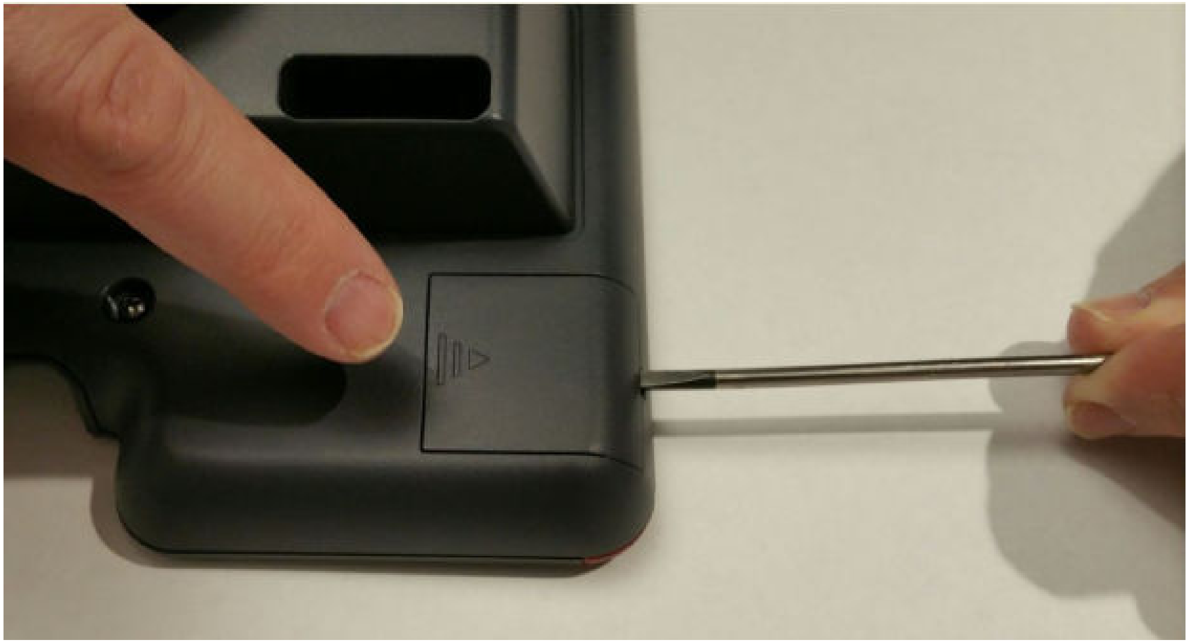
Before you begin

Get the following items:

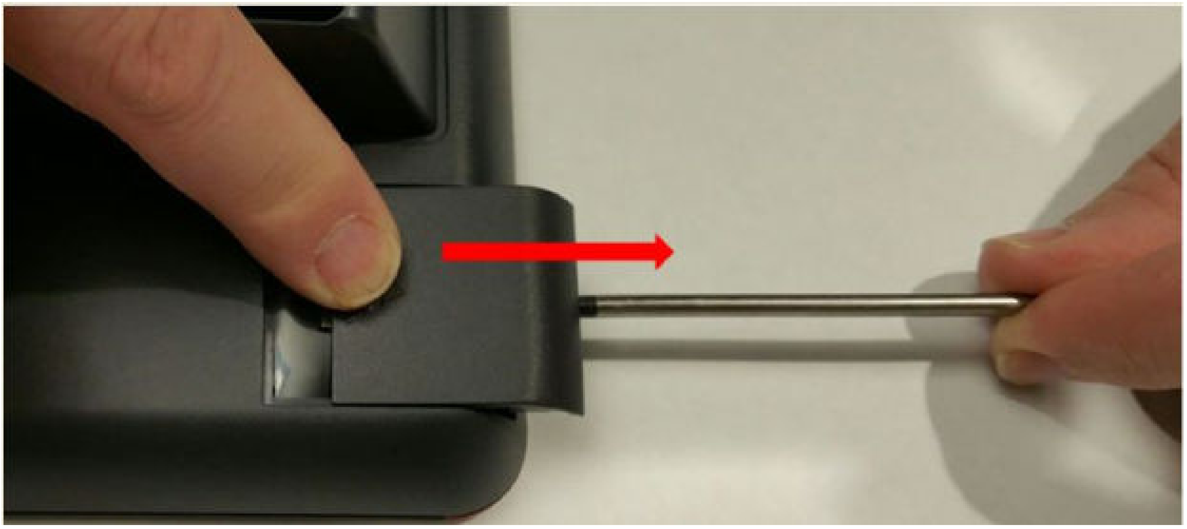
- Phillips #1 screw driver to install the screw of the J100 Wireless Module.
- A flat screw driver that fits in the opening of the module panel.

Procedure

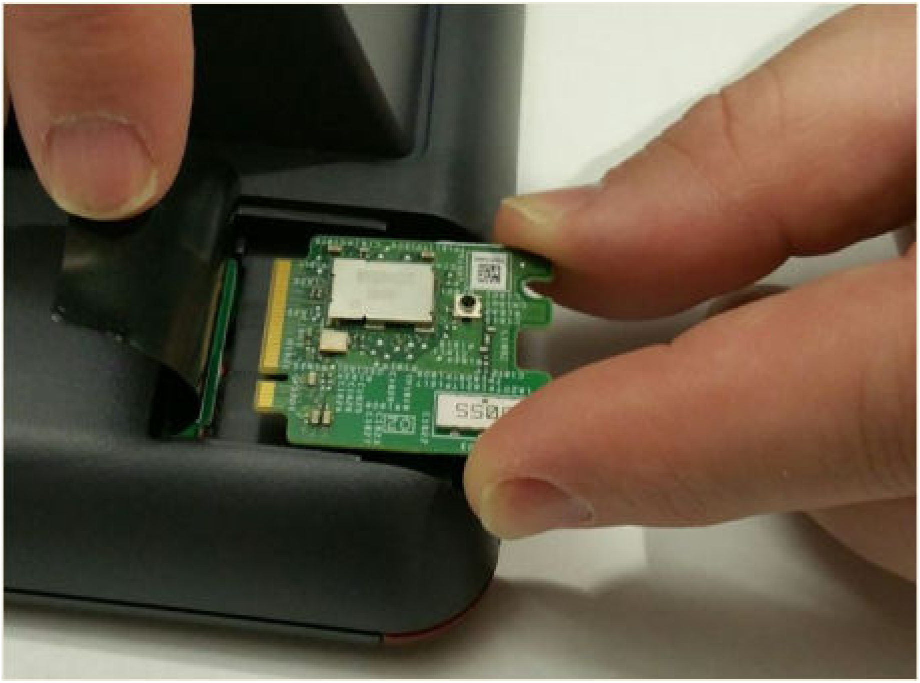
1. Insert the screw driver in the opening of the module panel to release the latch. Do not pry open the panel.



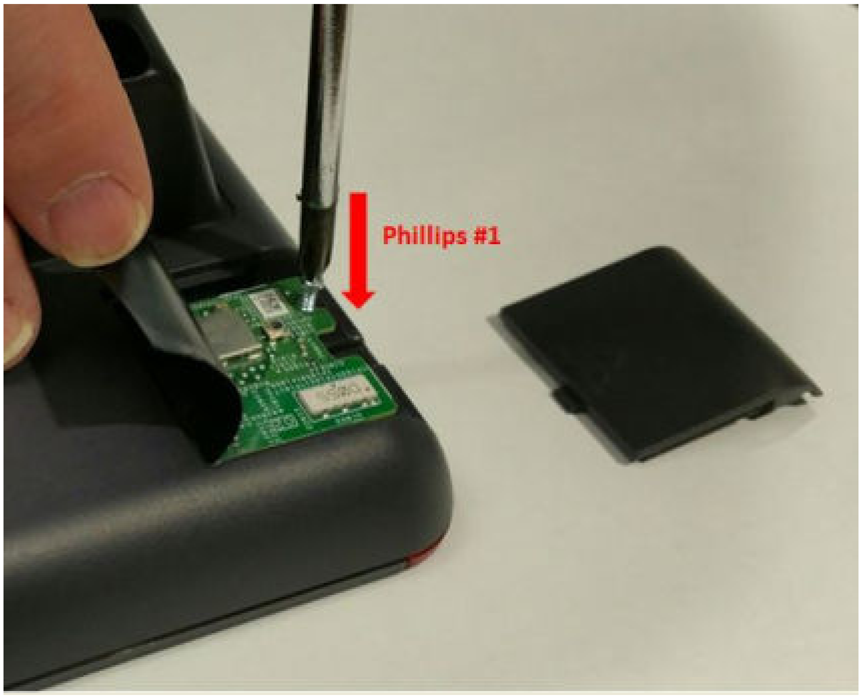
2. To remove the module panel, slide the panel out in the direction of the arrow.



3. Insert the J100 Wireless Module to the edge connector.



4. Use the Phillips #1 screwdriver to fasten the module.



5. Slide the module panel inward to close.

Configuring Wi–Fi using phone UI

About this task

Use this procedure to configure a Wi-Fi network by using phone UI. Note that switching networks causes a reboot of the phone.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the administration password.
3. Press **Enter**.
4. Select **Network Interfaces**.
5. Use the right arrow key to change **Network mode** to **Wi-Fi**.
6. Configure the following fields:
    - **Network config**: Specifies if the WLAN is connected automatically or manually.
    - **SSID**: Specifies the network name for the WLAN you are using. Use the navigation key to select another SSID.
    - **Wi-Fi networks**: Displays available WLAN.
7. Use the navigation key to select a WLAN and press **Connect**.
8. Press one of the following:
    - **Save**
    - **Cancel**
    - **Change**

List of Wi-Fi configuration parameters

Parameter Name	Default Value	Description
WIFISTAT	1	Specifies the network interface to be used for network connectivity. Value operation: • 0: Phone connects to only Ethernet network. • 1: Phone connects to Ethernet network, unless manually switched to Wi—Fi • 2: Phone connects to the Wi—Fi network with the SSID defined in the 46xxsettings.txt parameter WLAN_ESSID

Table continues…

Parameter Name	Default Value	Description
ENABLE_NETWORK_CONFIG_BY_USER	1	Enables network configuration to be modified by the user. Value operation: • 0: Disabled • 1: Enabled
WLAN_ESSID	N/A	Specifies the wireless network to be used. The name of the SSID ranges up to 32 characters.
WLAN_SECURITY	none	Specifies the security standard to be used for the wireless network. Value operation: • none: No security standard is defined. • wep: WEP security standard is defined. • wpa2psk: WPA2 security standard with pre-shared key is defined. • wpapsk: WPA security standard with pre-shared key is defined. • wpa2e: WPA enterprise security standard is defined.
WEP_DEFAULT_KEY	N/A	Specifies the index of WEP default key. Value operation: • 1 • 2 • 3 • 4
WLAN_COUNTRY	US	Specifies the ISO country code representing the Wi-Fi regulatory domain.
WLAN_ENABLE_80211D	0	Enables the phone to configure its Wi-Fi regulatory domain to match the 802.11d. Value operation: • 0: Disable

Table continues…

Parameter Name	Default Value	Description
		• 1: Enable
WEP_KEY_LEN	128 bit	Specifies the length of the WEP key. Value operation: • 40 bit • 64 bit • 128 bit
WLAN_PASSWORD	N/A	Specifies the pre-configured Wi-Fi network password. This parameter is applicable if the WIFISTAT is enabled and WLAN_SECURITY is wpa2psk, or WLAN_SECURITY is wpa2e, WLAN_WPA2E_EAP_METHOD is PEAP and WLAN_WPA2E_EAP_PHASE2 is MSCHAPV2. The password must be from 8-63 characters. Note that the space and ASCII 0x20, are not supported.
WEP_KEY_1 to WEP_KEY_4	N/A	Specifies the name of the WEP key. The name of the 40 bit key and 128 bit key are of 10 hex digits and 26 hex digits respectively.
WLAN_WPA2E_EAP_METHOD	PEAP	Specifies the pre-configured 802.1x EAP method. This parameter is applicable if WIFISTAT parameter is enabled and WLAN_SECURITY is set as wpa2e. Value operation: • PEAP • TLS
WLAN_WPA2E_IDENTITY	N/A	Specifies the 802.1x name of pre-configured Wi-Fi network. This parameter is applicable if WIFISTAT parameter is enabled and WLAN_SECURITY is set as wpa2e.

Table continues…

Parameter Name	Default Value	Description
		The name must be from one to 32 characters. Note that the space character and ASCII 0x20, are not supported.
WLAN_WPA2E_ANONYMOUS_IDENTITY	N/A	Specifies the 802.1x anonymous name of pre-configured Wi-Fi network. This parameter is applicable if WIFISTAT parameter is enabled, WLAN_WPA2E_EAP_METHOD is set to PEAP and WLAN_SECURITY is set as wpa2e. The name must be from one to 32 characters. Note that the space character and ASCII 0x20, are not supported.
WLAN_L2QUAD	6	Specifies the layer 2 priority value for audio frames generated by the telephone. Valid value is from zero to seven.
WLAN_DSCPAUD	46	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for audio frames generated by the telephone. Valid value is from zero to 63.
WLAN_L2QSIG	3	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for audio frames generated by the telephone. Valid value is from zero to 63.
SET WLAN_DSCPSIG	34	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for signaling frames generated by the telephone. Valid value is from zero to 63.

Wall mounting Avaya J100 Series IP Phones

About this task

Wall mounting kit and procedure of Avaya J100 Series IP Phones are similar except the wall mounting bracket. Wall mounting brackets look different for Avaya J169/J179 IP Phone and Avaya

J129 IP Phone. You can order the kit separately, using the part numbers that correspond to the phone.model. For example, the part number of the wall mount bracket is 700512707. The procedure describes the wall mounting procedure with illustrations as reference.

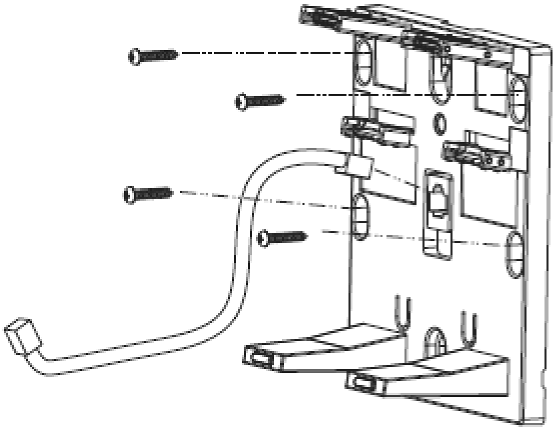
Before you begin

Get the following items:

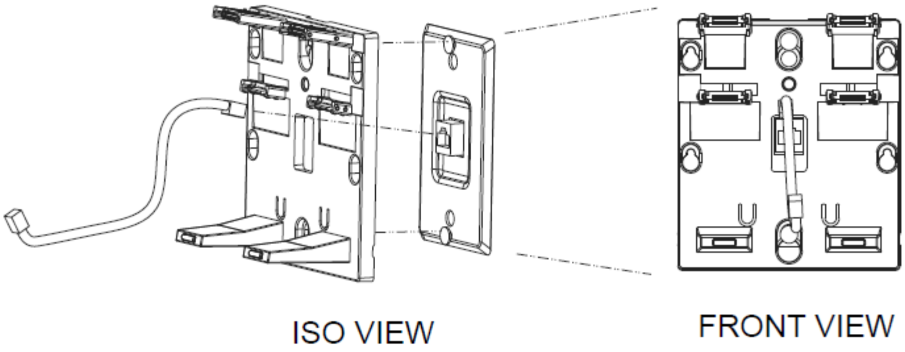
- Wall mounting kit that contains a wall mount bracket, and an Ethernet cable.
- Four #8 screws. The screws are not provided with the wall mounting kit.

Procedure

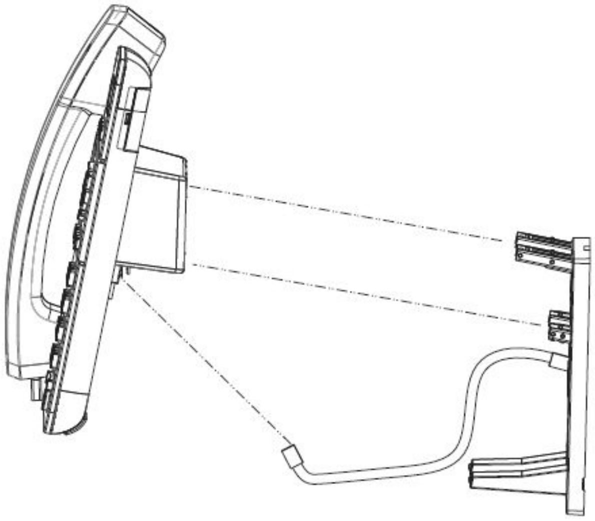
1. Do one of the following:
    - Place the bracket on the wall, drill holes, and then drill-in the #8 screws.



    - If there is a pre-installed wall plate, place the wall mount bracket over the wall plate. In this case, you do not need the screws.



2. Attach one end of the Ethernet cable to the 10/100 network port of the phone and the other end to the wall jack.
3. Attach the phone to the wall mount bracket by inserting the two upper tabs of the wall mount bracket into the slots on the back of the phone. The lower pair of tabs rest against the back of the phone and ensure that the phone does not move when the keys are pressed.



Software installation

Phone installation process

You can install Avaya J100 Series IP Phones in the following ways:

- With the Device Enrollment Server (DES) discovery process: The installation process begins after the phone is connected to a network. This is an automated process.
- Without the DES discovery process: The installation process includes a series of pre-configuration tasks.

Phone installation with DES

DES server

Device Enrollment Server (DES) redirects the out of box phone to the configuration file server after the phone is connected to a network and the installation procedure begins automatically. The DNS address of the DES server is hard coded to the phone firmware and the administrator can install the phone by connecting the out of box phone to a network. After the first boot process, the administrator can disable the DES functionality by setting DES_STAT=0 in DHCP option 242 or from the settings file by putting the parameter DES_STAT=0.

Installing the phone by using the DES eliminates the need of manual configuration of provision server.

Note:

DES only works if a provisioning server has been configured in the Avaya DES service for the phone's MAC address. This is configured by the service provider.

Installing the phone using DES server

After the phone boots up, it prompts to enable or disable DES discovery. You can select one of the following:

- Yes: The phone contacts the DES server and the DES server redirects the phone to the configuration file server. The phone receives all the configuration related parameters and upgrade file from the file server for installation.
- No: The phone skips the DES server discovery process. The administrator must provide all the configuration related parameters through the following methods:
  - Phone UI
  - Web UI
  - DCCP
  - LLDP

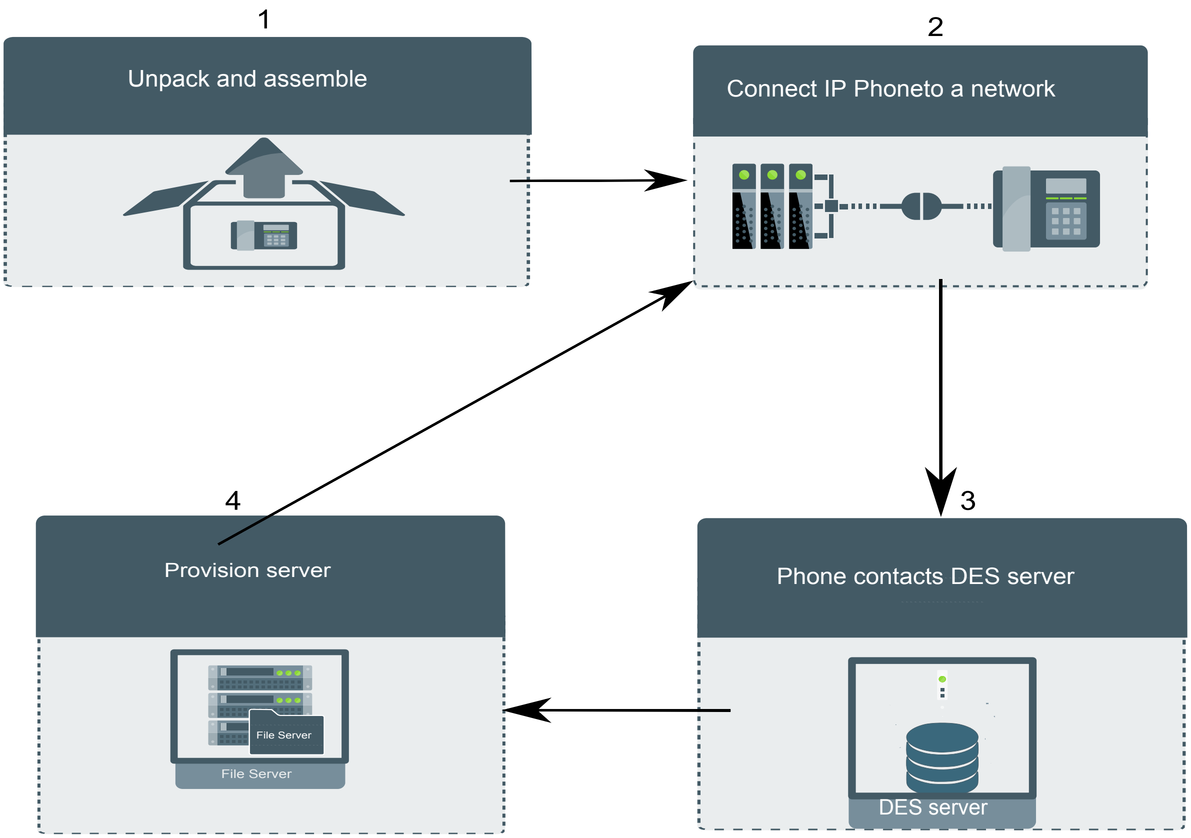
After a time out of 30 seconds of the prompt the phone initiates DES discovery and contacts the provision server for configuration parameter if a provisioning server is not obtained from DHCP. If the administrator selects **Yes** in the prompt, the phone forces DES discovery and it overrides the provision server provided by DHCP.

Related links

Diagram: Phone deployment with DES on page 24

List of configuration parameters on page 143

Diagram: Phone deployment with DES



Related links

Phone installation with DES on page 22

List of configuration parameters on page 143

Phone installation without DES

This section describes the procedure to install the phone without invoking the DES discovery process.

Initial setup checklist

Use this checklist to gather, record, and verify the information during the installation.

No.	Task	Reference	✔
1	Check the prerequisites.	See Hardware and software prerequisites on page 25 for more information.	
2	Configure system manager user profile.	See Avaya Aura System Manager user profile worksheet on page 28 for more information.	
3	Configure the servers.	See Server configuration on page 70 for more information.	
5	Configure LLDP.	See Configuration through LLDP on page 77 for more information.	
6	Configure VLAN.	See Virtual LAN (VLAN) overview on page 87 for more information.	
9	Install the phone.	See Installing the phone on page 30 for more information.	

Hardware and software prerequisites

Check the prerequisites to ensure that you have the required software and hardware before you install the Avaya J100 Series IP Phones .

Hardware prerequisites

Ensure that the LAN:

- Uses Ethernet Category 5e or Ethernet Category 6 cabling
- Has one of the following specifications:
  - 802.3af PoE
  - 802.3af PoE injector

You can also power the phone using the Avaya DC 5 volt AC power adapter which you can order with the device.

Software prerequisites

Ensure that your network already has the following components installed and configured:

- Avaya Aura® Session Manager 6.3.8 or later

- Avaya Aura® Communication Manager 6.3.6 or later
- Avaya Aura® System Manager 6.3.8 or later
- If applicable, Avaya Aura® Presence Services 6.2.4 or later
- If applicable, Avaya Aura® Session Border Controller 7.0 or later
- If applicable, IP Office IPO 11.0.0 or later
- A DHCP server for providing dynamic IP addresses to the Avaya J100 Series IP Phones.
- A file server, an HTTP, HTTPS, or the Avaya Aura® Utility Services for downloading the software distribution package and the settings file

IPv6 deployment requires Avaya Aura® Session Manager v7.1 or later, Avaya Aura® Communication Manager v7.1 or later, Avaya Aura® System Manager v7.1 or later, and Avaya Aura® Session Border Controller v7.1 or later. For more information about installing and configuring the components, see their respective documentation.

Administration methods

You can use the following methods to administer the devices. The following table lists the configuration parameters that you can administer through each of the corresponding methods.

Method	Can administer				
	IP addresses	Tagging and VLAN	Network Time Server	Quality of Service	Application-specific parameters
DHCP	✔	✔	✔	—	✔
LLDP	—	✔	—	✔	—
Settings file	—	✔	✔	✔	✔
Avaya Aura® System Manager and IP Office	—	—	—	—	✔
Administration menu on the phone	✔	✔	—	—	✔
Web UI	✔	✔	✔	✔	✔

Precedence of administration methods

Most of the parameters are configured through multiple methods. If you configure a parameter through more than one method, the device applies the settings of the method that has a higher precedence. The following list shows the precedence of the methods in the highest to lowest order:

1. Administration menu on the phone. When the parameter USE_DHCP is set to 1, the phone gets the DHCP values from the DHCP rather than Administration menu of the phone.
2. Administering the phone from the web UI.

3. Avaya Aura® System Manager and IP Office.
4. `46xxsettings.txt` file
5. DHCP.
6. LLDP. There is an exception of LLDP getting a higher precedence than the Settings file and DHCP when the layer 2 parameters, such as L2QVLAN, L2Q, L2QAUD, L2QVID, L2QSIG, DSCPAUD, DSCPSIG, DSCPVID, and PHY2VLAN are set through LLDP.

Note:

When parameters of the `46xxsettings.txt` file are removed, or are not used, they reset to their default value.

Diagram: phone deployment process

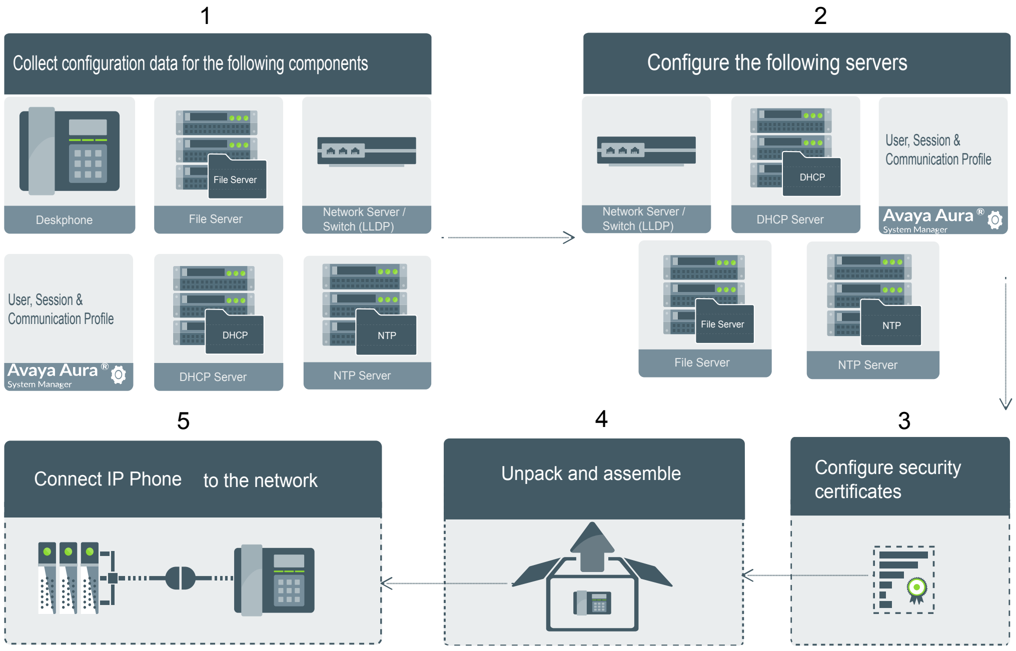
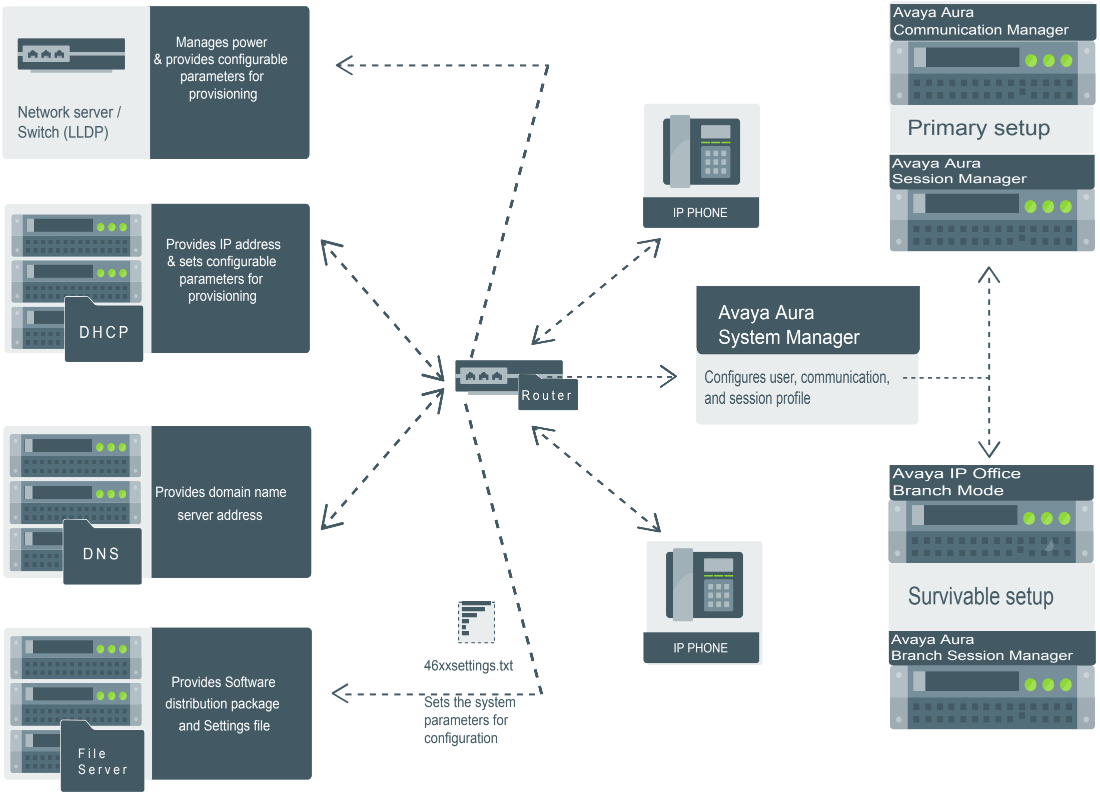

1
Collect configuration data for the following components
Deskphone
File Server
File Server
Network Server / Switch (LLDP)
User, Session & Communication Profile
Avaya Aura® System Manager
DHCP
DHCP Server
NTP
NTP Server
2
Configure the following servers
Network Server / Switch (LLDP)
DHCP
DHCP Server
User, Session & Communication Profile
Avaya Aura® System Manager
File Server
File Server
NTP
NTP Server
3
Configure security certificates
4
Unpack and assemble
5
Connect IP Phone to the network

Diagram: IP phone setup


Network server / Switch (LLDP)
Manages power & provides configurable parameters for provisioning
DHCP
Provides IP address & sets configurable parameters for provisioning
DNS
Provides domain name server address
File Server
Provides Software distribution package and Settings file
46xxsettings.txt
Sets the system parameters for configuration
Router
IP PHONE
IP PHONE
Avaya Aura System Manager
Configures user, communication, and session profile
Avaya Aura Communication Manager
Primary setup
Avaya Aura Session Manager
Avaya IP Office Branch Mode
Survivable setup
Avaya Aura Branch Session Manager

Avaya Aura® System Manager user profile worksheet

Populate the values in the corresponding fields before stating the installation process of the phone.

Data for	Field	Value	Notes
System Manager User Profile			
Identity tab			
	Login Name		
	Localized Display Name		
	Endpoint Display Name		
	Language Preference		
	Time Zone		
Presence Profile			
	System		
	IM Gateway SIP Entity		
	Publish Presence with AES collector		

Table continues…

Data for	Field	Value	Notes
Communication Profile tab			
Communication Profile section			
	Communication Profile Password		
Session Manager Profile section			
	Primary Session manager		
	Secondary Session Manager		
	Survivability Server		
CM Endpoint Profile section			
	System		
	Profile Type		
	Use Existing Endpoints		
	Extension		
	Endpoint Template		
	Voice Mail Number		
	Presence server		
	Conference server		
Messaging Profile section			Optional
	System		
	Mailbox Number		
	Template		
	Password		
SIP settings			For registering phones.
	SIP controller list		
	SIP domain		
File server address			To download the software distribution package and the `Settings` file.
	HTTP server or TLS server		Set the appropriate file server address in the `46xxsettings.txt` file, LLDP and DHCP.

Note:

For information about IP Office preinstallation data gathering, see *Avaya IP Office Platform 10.0 SIP Telephone Installation Notes*.

Installing the phone

Before you begin

You must do the following:

- Configure the file server.
- Download and extract the firmware zip file to your file server.
- Configure the `46xxsettings.txt` file.

Procedure

1. Set up the phone hardware.
2. Plug the Ethernet cable to the phone.

   The phone powers up and starts to initialize.
3. The initialization procedure consists of the following processes:
   a. The phone checks for LLDP messages.
   b. The phone sends a DHCP DISCOVER message to discover the DHCP server in the network and invokes the DHCP process.

      If the phone does not receive a provisioning server address from the configuration setup, the phone displays the Configure Provision Server screen.
   c. In the Configure Provision Server screen, press the **Config** softkey and enter the address of the provisioning server. The provisioning server address can be in the form of IP address or a Fully Qualified Domain Name (FQDN). To enter the dot symbol (.) in the field, press the alphanumeric softkey to toggle to the alphanumeric mode.
   d. The phone verifies the VLAN ID, and starts tagging the data and voice packets accordingly.
   e. The phone sends and identifies an upgrade script file, gets the `Settings` file, the language files, and any firmware updates.
      - If configured to use simple certificate enrollment protocol (SCEP), the phone downloads a valid device certificate.
      - The phone displays only the **Admin** softkey for 15 seconds, and then the **Admin** and the **Login** softkeys.

        Note:

        For subsequent restarts, if the user login is automatic and the supplied credentials are valid, the **Login** softkey is not displayed.

4. Do one of the following:
    - To access the user login screen, press the **Login** softkey.
    - To access the Admin menu, press the **Admin** softkey and enter the admin menu password.

Post installation checklist

To ensure that the phone is properly installed and running properly, verify that the following requirements are complete.

No.	Task	Reference	✔
1	Has the phone acquired an IP address?	N/A	
2	Are you able to make a call from the phone?	For more information, see device specific using guide.	
3	Are you able to modify the phone's Settings file parameters and end user settings.	List of configuration parameters on page 143	
4	Are you able to upgrade your phone?	Device upgrade process on page 125	
5	For security considerations, have you configured the phone setup with TLS signaling? Have you installed the appropriate private network authentication certificates?	Certificate management on page 104	
6	It is critical that you verify Emergency calling is working properly in your network. It may be necessary to make arrangements with the appropriate authorities to test this functionality.	For more information, see *Administering emergency numbers*	

Note:

For more information about IP Office specific installation, see the following IP Office documents:

- *Avaya IP Office™ Platform Solution Description*
- *Avaya IP Office™ Platform Feature Description*

Chapter 4: Configuring the phone using web interface

Logging in and logging out of the web UI

About this task

Use this procedure to log in or log out of the web UI. Note that the system prompts you to change your default password only after the first log in.

Before you begin

- On the phone UI, use the admin menu to do the following:
  - Change the status of **Web Server** to **On admin** menu.
  - Obtain the IP address of the phone **Administration** menu.

Procedure

1. In your browser, enter the IP address of the phone and press `Enter`.
2. On the login page, type the following:
   - **Username**: The user name is always admin.
   - **Password**: The default password is 27238.
3. Click **Login**.

   The system displays the Change Default Password.
4. In the Change Default Password dialog box, type the following:
   - **New password**
   - **Confirm password**
5. Click **Update**.

   The system displays the login page.
6. To log out of the web UI, click **Logout**.

Changing password

Procedure

1. Log in to the web GUI using your username and current password.
2. In the left pane of the screen, click **Password**.

   The **System** section is displayed.
3. In the **System** section, enter your old password in the **Old Password** field, enter your new password in the **New Password** field, re-enter your new password in the **Confirm Password** field, and click **Save**.
   - Your password must be between 8 to 31 alphanumeric characters including upper, lower and special characters.
   - Your password should contain at least 2 digits.
   - Allowed special characters are ~!@#$%^&*_-+=`|\(){}[]:;'<>,.?/.

Configuring environment settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Environment Settings**.
3. In the Environment Setting area, enable the following fields:
   - **AURA environment**: To set Avaya Aura as your environment.
   - **Discover AVAYA environment**: To discover whether the phone supports Avaya Aura SIP AST feature.
   - **IP Office**: To set IP Office as your environment.
   - **3PCC environment**: To set a third-party call controller as your environment.

Configuring date and time

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Date & Time**.
3. In the SNTP area, configure the following:
   - **SNTP Server**: Type the SNTP server IP address .

- **SNTP SYNC Interval**: Type the SNTP synchronization time interval to re-synchronize the phone's local time . The valid time interval is from 60 to 2880 minutes. The default synchronization time is 1440 minutes.
- **GMT Offset**: Type the GMT Offset value in hours and minutes between the local standard time and Greenwich Mean Time (GMT). The offset value ranges from 0:00 to ±12:59.

4. In the Daylight Saving area, configure the following:
    - **Daylight Saving Mode**: The options are :
        - **Automatic daylight saving time**:
        - **Manual daylight saving activated (time set to DSTOFFSET)**:
        - **Manual daylight saving adjustment (as specified by DSTSTART and DSTSTOP)**:
    - **DST Offset**: Specifies the offset time between standard time and daylight savings time. The options are:
        - **0**
        - **1 hour**
        - **2 hour**
    - **DST Start**: Specifies the daylight savings time start date and time with a format of either **odddmmmht** or **Dmmmht**; where:
        - **o** represents a one-character ordinal adjective. For example, 1 for first, 2 for second, 3 for third, 4 for fourth, or L for last.
        - **D** represents 1 or 2 ASCII digits or character representing the date of the month.
        - **ddd** represents a three-characters containing the English abbreviation for the day of the week. For example, Sun for Sunday, Mon for Monday.
        - **mmm** represents a three-character English abbreviation for the month. For example, Jan for January, Feb for February.
        - **h** represents one-numeric digit representing the time to make the adjustment at hAM (0h00 in military format).

          The valid values of h are "0" to "9";
        - **t** represents one-character for the time zone to which to make the adjustment. For example, "L" for local time or "U" for Universal Time.
    - **DST Stop**: Specifies the daylight savings time stop date and time with a format of either **odddmmmht** or **Dmmmht**; where:
        - **o** represents a one-character ordinal adjective. For example, 1 for first, 2 for second, 3 for third, 4 for fourth, or L for last.
        - **D** represents 1 or 2 ASCII digits or character representing the date of the month.
        - **ddd** represents a three-characters containing the English abbreviation for the day of the week. For example, Sun for Sunday, Mon for Monday.

- **mmm** represents a three-character English abbreviation for the month. For example, Jan for January, Feb for February.
- **h** represents one-numeric digit representing the time to make the adjustment at hAM (0h00 in military format).

  The valid values of h are "0" to "9";
- **t** represents one-character for the time zone to which to make the adjustment. For example, “L” for local time or “U” for Universal Time.

5. Click one of the following :
   - **Save**: To save the configuration changes.
   - **Reset to Default**: To revert to the default values.
   - **Help**: To view the online help.

Configuring Ethernet settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Ethernet**.
3. Configure the following areas:
   - IP Configuration
   - 802.1X
   - VLAN
   - QoS
   - LLDP
   - Interface
4. Click one of the following :
   - **Save**
   - **Reset to Default**
   - **Help**

Related links

Ethernet settings field descriptions on page 36

Ethernet settings field descriptions

Name	Description
IP Configuration	
Use DHCP	Provides the IP address to your phone automatically or manually. The options are: • **Yes**: To assign the IP address automatically to your phone. • **No**: To assign the IP address manually to your phone. Note that, to assign the IP address manually, you must also configure the **IP Address**, **Subnet Mask**, and **Gateway IP Address** fields manually.
Continue to use DHCP information after lease expiry	Specifies whether the DHCP information can be used after the lease expires. The options are: • **Yes**: To use the assigned IP address after the DHCP lease expires. • **No**: To stop using the assigned IP address after the DHCP lease expires.
IP Address re-use time	Specifies the time in seconds to re-use the assigned IP address after the DHCP lease expires. The default value is 60 seconds.
IP Address	Specifies the IP address of the phone. .
Subnet Mask	Specifies the network mask address. To assign the network mask address manually to your phone, type the address in the corresponding field.
Gateway IP Address	Specifies the IP address of the gateway. To assign the gateway IP address manually to your phone, type the address in the corresponding field.
802.1X	
Supplicant Operating Mode	Specifies the 802.1X supplicant operating mode. The options are: • **Disabled** • **Unicast** • **Multicast**
Pass-through Operating Mode	Specifies the 802.1X pass-through operating mode. Pass-through refers to the forwarding of EAPOL frames between the phone's Ethernet line interface and the secondary PC Ethernet interface.

Table continues…

Name	Description
	The options are: • **Without proxy logoff** • **With proxy logoff** • **disabled**
Authentication Mode	Specifies the authentication method to be used by 802.1X. The options are: • **MD5** • **TLS**
VLAN	
VLAN	Specifies whether the VLAN tagging is enabled or disabled. The options are: • **Auto**: To support VLAN functionality by using the phone network. • **On**: To support the VLAN functionality by using the internal switch of the phone. • **Off**: To disable the VLAN functionality of the phone.
VLAN ID	Specifies the VLAN ID. To assign a VLAN ID, type the VLAN ID. Configure this parameter if the phone uses a different VLAN than the default data VLAN.
VLAN Separation Mode	Specifies the VLAN separation mode. The options are : • **Disable** • **Enable**
VLAN Test - Wait Time for DHCP Offer	Specifies the wait time interval in seconds to receive a DHCPOFFER on a non-zero VLAN. The default value is 60 seconds.
PC Port VLAN ID	Specifies the VLAN ID of the PC port.
Tags to PC Eternet Interface	Specifies whether the VLAN tags are stripped from Ethernet frames that leave the computer (PC) port. The options are: • **Do not remove** • **Remove**
QoS	
Audio Priority (Layer 2)	Specifies the Layer 2 priority value for audio (RTP and RTCP) streams. Valid priority values are 0 to 7.
Signaling Priority (Layer 2)	Specifies the Layer 2 priority value for signaling protocol messages. Valid priority values are 0 to 7.

Table continues…

Name	Description
Audio DiffServ (Layer 3)	Specifies the layer 3 Differentiated Services (DiffServ) code point for audio frames generated by the phone. Valid values are 0 to 63.
Signaling DiffServ (Layer 3)	Specifies the layer 3 Differentiated Services (DiffServ) code point for signaling frames generated by the phone. Valid values are 0 to 63.
LLDP	
LLDP	Specifies the status of LLDP. The options are: • **Disabled** • **Enabled** • **Enabled- only if LLDP frame is received**
Interface	
Ethernet	Specifies the speed and duplex settings for the Ethernet line interface. The options are: • **auto-negotiate** • **10Mbps half-duplex** • **10Mbps full-duplex** • **100Mbps half-duplex** • **100Mbps full-duplex**
PC Ethernet	Specifies the speed and duplex settings for the secondary (PC) Ethernet interface. The options are: • **auto-negotiate** • **10Mbps half-duplex** • **10Mbps full-duplex** • **100Mbps half-duplex** • **100Mbps full-duplex** • **Disable**
PC Ethernet auto-MDIX	Specifies the status of the auto-MDIX Value Operation. The options are: • **Enable** • **Disable**

Related links

Configuring Ethernet settings on page 35

Configuring Wi-Fi settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Wi-Fi**.
3. Configure the following areas :
    - WiFi Setting
    - IP Configuration
    - WEP
    - WPA2 Enterprise (802.1x)
    - QoS
4. Click one of the following:
    - **Save**
    - **Reset to Default**
    - **Help**

Related links

Wi-Fi settings field descriptions on page 39

Wi-Fi settings field descriptions

Name	Description
WiFi Setting	
Country	Specifies the country code to define the Wi-Fi radio parameters permitted by the local regulatory domain.
Use of 802.11d	Configures the 802.11d specifications automatically to the local regulatory domain for the WLAN network. The options are: • **Disable** • **Enable**
SSID	Specifies the network name for the WLAN you are using. You can also type the SSID in this field.
Password	Specifies the password for the SSID. You can also type the password in this field. Note that the maximum length of a password is 63 characters.

Table continues…

Name	Description
Security	Specifies the WLAN security standard for your WiFi network. The options are: • **None** • **WEP Security** • **WPA2 security with pre-shared key** • **WPA security with pre-shared key** • **WPA2 Enterprise security (802.1x auth.)**
IP Configuration	
Use DHCP	Provides the IP address to your phone automatically. You can also manually assign the IP address. The options are: • **Yes**: To assign the IP address automatically to your phone. To display the IP address automatically • **No**: To assign the IP address manually to your phone. If you assign the IP address manually, you must also configure the **IP Address**, **Subnet Mask**, and **Gateway IP Address** fields manually.
IP Address	Specifies the IP address of the phone. You can also type the IP address in this field.
Subnet Mask	Specifies the network mask address. You can also type the network mask address in this field.
Gateway IP Address	Specifies the IP address of the gateway. You can also type the gateway IP address in this field.
WEP	
WEP Authentication	Specifies the type of WEP authentication method used by your WiFi network. The options are: • **Open systems** • **Shared key**
WEP Key Length	Specifies the passcode key length for your WEP security. The options are: • **40 bit** • **64 bit** • **128 bit**
WEP Default Key	Specifies the default key for your WiFi network. You can select a default key from WEP key 1 to 4.

Table continues…

Name	Description
WEP Key 1–4	Specifies the WEP key values for the WiFi network. You can configure up to 4 WEP keys. Note that the maximum length of a WEP key is 26 alphanumeric characters that can include the following: • Blank • 0–9 • A-F Blank, 0 to 9, A to F. make a vertical list
WPA2 Enterprise (802.1x)	
EAP Authentication Method	Specifies the type of EAP authentication method. The options are: • **PEAP** • **TLS**
EAP Phase 2 Authentication Method	
Authentication Identity	
Authentication Anonymous Identity	
QoS	
Audio Priority (Layer 2)	Specifies the Layer 2 priority value for RTP and RTCP audio streams. Valid priority values are 0 to 7.
Signaling Priority (Layer 2)	Specifies the Layer 2 priority value for signaling protocol messages. Valid priority values are 0 to 7.
Audio DiffServ (Layer 3)	Specifies the layer 3 Differentiated Services (DiffServ) code point for audio frames generated by the phone. Valid values are 0 to 63.
Signaling DiffServ (Layer 3)	Specifies the layer 3 Differentiated Services (DiffServ) code point for signaling frames generated by the phone. Valid values are 0 to 63.

Related links

Configuring Wi-Fi settings on page 39

Configuring network settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Network**.

3. Configure the following areas :
    - Network
    - DNS
    - ICMP
    - TCP
    - TLS
    - Web Server
4. Click one of the following:
    - **Save**
    - **Reset to Default**
    - **Help**

Related links

Network settings field description on page 42

Network settings field description

Name	Description
Network	
Network Mode of Operation	Specifies the network mode used by the phone. The operations are: • **Ethernet only** • **Ethernet** • **Wi-Fi**
DNS	
DNS Server	Specifies the IP addresses of the DNS servers added to the network. You can type the DNS servers in this field.
DNS Domain	Specifies the domain name of the IP address. You can type the DNS domain name in this field.
ICMP	
Destination Unreachable Message Control	Specifies the type of the ICMP destination unreachable messages . The options are: • **No**

Table continues…

Name	Description
	- **Limited Port Unreachable messages** - **Protocol and Port Unreachable messages**
Redirect Message Control	Specifies whether the ICMP redirect messages are processed or not. The options are: - **Yes** - **No**
TCP	
Send TCP Keep Alive Message	Specifies whether the TCP/IP keep-alive messages are enabled or disabled at the system. The options are: - **Disable** - **Enable**
TCP Keep Alive Time	Specifies the wait time interval of the phone before sending out the TCP keep-alive message (TCP ACK message) to the far-end. The valid time interval range is from 10 to 3600 seconds.
TCP Keep Alive Interval	Specifies the TCP keep-alive packet re-transmission interval The valid time interval range is from 5 to 60 seconds.
TLC	
Use TLS Version	Specifies the TLS versions to be used in the network. The options are: - **1.0 and 1.2** - **Only 1.2**
Web Server	
Web Server	Specifies whether the web server is enabled or disabled. The options are : - **Enable** - **Disable**
HTTP Listen Port	Specifies the port number of the web server when the web interface is accessed using HTTP.

Table continues…

Name	Description
	The default port number is 80.
HTTPS Listen Port	Specifies the port number of the web server when the web interface is accessed using HTTPS. The default port number is 443.
Use custom certificate for Web Server	Specifies whether to use the custom server certificate when the web interface is accessed using HTTPS. The options are: • **No** • **Yes**

Related links

[Configuring network settings](#) on page 41

Configuring management settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Management**.
3. Configure the following areas:
    - Device Enrollment Server
    - HTTP Provisioning Server
    - HTTPS Provisioning Server
    - Configuration
    - Firmware
    - Backup/Restore User Data
4. Click one of the following:
    - **Save**
    - **Reset to Default**
    - **Help**

Related links

Management settings field descriptions on page 45

Management settings field descriptions

Name	Description
HTTP Provisioning Server	
HTTP Server Address	Specifies the IP address of the of the provisioning file server.
HTTP Server Directory Path	Specifies the path to prepare all configurations and data files the device might request when starting up, that is, the path, relative to the root of the HTTP file server, to the directory in which the device configuration and date files are stored.
HTTP Port	Specifies the HTTP port address. The default port number is 80.
HTTPS Provisioning Server	
HTTPS Server Address	Specifies the IP address of the HTTPS provisioning file server.
HTTPS Server Directory Path	Specifies the path to prepare all configurations and data files the device might request when starting up, that is, the path, relative to the root of the HTTPS file server, to the directory in which the device configuration and date files are stored.
HTTPS Port	Specifies the HTTPS port address. The default is 443.
Configuration	
Import Configuration File	Enable user to import a configuration file. To import a configuration file, click **Choose File** to browse your local PC or any PC connected to the network, select the file and click **Import**. The administrator needs to restart the phone after the configuration file is uploaded.
Export Configuration File	Enable user to export a configuration file. To export a configuration file, click **Export** .
Firmware	
Software Version	Specifies the software version of the SIP software
Backup Software Version	Specifies the backup software version.
Firmware Upgrade	Enable user to import the upgrade file from the local PC or any PC connected to the network. To upload the firmware upgrade file, click **Choose File** to browse your local PC, select the file and click **Upgrade**.

Table continues…

Name	Description
	The phone reboots after you select **Yes** in the prompt.
Language	
Language File	Enables the user to upload language files to the phone from the local PC. To import a language file from your local PC or any PC connected to the network, click **Choose File** to browse to your local PC, select the language file and click **Import**.
Language File Uploaded	Specifies the available language files for the phone to be used.

Related links

Configuring management settings on page 44

Configuring settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Settings**.
3. Configure the following areas:
    - Language
    - Feature access
    - Phone Menu Option
    - Call Log
    - Contacts
    - Emergency Call
    - Phone Lock
    - Other
    - Audio
    - Dialing
    - Enhanced Local Dialing Rules
    - Admin
4. Click one of the following:
    - **Save**
    - **Reset to Default**

- **Help**

Related links

Settings field descriptions on page 47

Settings field descriptions

Language

Name	Description
Language	
Available Language File	Specifies the name of the default system language file used in the phone. You can delete the default language file by clicking **Delete**.
Import Language File	You can browse and import a language file from your local machine by clicking **Choose File** > **Import**.
Language file to upload	Specifies the language file to be uploaded.
Phone Language	Specifies the phone language file.
Feature Access	
Call Forward	Specifies the status of the call forward feature . The options are: • **Do not allow** • **Allow**
Number of Ring cycle before Call Forward	Specifies the number of ring cycles before the call is forwarded The default delay is one ring cycle.
Do Not Disturb	Specifies the status of the DND feature . The options are: • **Do not allow** • **Allow**
DND Priority over Call Forward (Unconditional, Busy)	Specifies the IP address of the gateway. You can type or ethe gate way IP address in this field.
Auto Answer	Specifies the status of the Auto Answer feature. The options are: • **Do not allow** • **Allow**
Mute on Auto Answer	Specifies the mute status when Auto Answer feature is enabled. The options are: • **Yes**

Table continues…

Name	Description
	- **No**
Hold Reminder Timer	Specifies the number of seconds after which the phone plays the hold reminder tone .
Transfer on Conference hangup	Specifies whether a conference call continues after the host hangs up. The options are: - **Yes** - **No**
Presence	Specifies whether to enable or disable complete presence function. The options are: - **Do not allow** - **Allow**
Phone Menu Options	
Settings	Specifies whether the Options & Settings menu is provided to the user . The options are: - **Do not allow** - **Allow**
Network Info Screen	Specifies whether the Network Information screen is provided to the user. The options are: - **Do not allow** - **Allow**
Logout	Specifies whether the logout function is provided to the user. The options are: - **Do not allow** - **Allow**
SSL Version	Specifies the version of the SSL certificate.
User-ID Field	Specifies the option to disable or enable the **User-ID Field** form menu. The options are: - **Do not allow** - **Allow**
UDP Transport	Specifies whether UDP transport is allowed.

Table continues…

Name	Description
	The options are: - **Do not allow** - **Allow**
Network Configuration by User	Specifies whether network configuration can be modified by a user. The options are: - **Do not Allow to Modify** - **Allow to Modify**
Call Log	
Call Log	Specifies whether to enable or disable complete call log application. The options are: - **Do not allow** - **Allow**
Enable Redial	Specifies whether to enable or disable the capability to redial out of a list of recently dialed numbers instead of performing last number redial. The options are: - **Do not allow** - **Allow**
Contacts	
Local Contacts	Specifies whether to enable or disable complete Contact Application feature. The options are: - **Do not allow** - **Allow**
Contact Name Format	Specifies the format of the contact name to be displayed in the contact list. The options are: - **'Last Name' 'First Name'** - **'First Name' 'Last Name'**
Contact Name display logic	Specifies how to match a dialed string on an incoming call with the users contacts. The options are: - **Match the number completely**

Table continues…

Name	Description
	- **Match shorter number completely to the rightmost digits of longer number** - **Match at least 4 rightmost digits**
Emergency Call	
Emergency Numbers	Specifies the emergency contact number.
Emergency Softkey	Specifies whether the emergency softkey is displayed after the phone is registered. The options are: - **Do Not Display** - **Display without Confirmation** - **Display with Confirmation**
Softkey Emergency Number	Specifies the number to be used as a softkey for emergency numbers.
Emergency Softkey on Unregistration	Specifies whether the emergency softkey is displayed when the phone is not registered. The options are: - **Do Not Display** - **Display without Confirmation** - **Display with Confirmation**
Phone Lock	
Enable Phone Lock	Specifies whether the phone lock feature of the phone is enabled. The options are: - **Do Not Allow** - **Allow**
Phone Lock Idle Time	Specifies the idle time after which the phone is locked. The value is 0 to 10080 minutes.
Others	
Softkey Configuration	Specifies which feature will show up on which softkey on the phone screen.
Branding Volume	Specifies the level of the Avaya audio brand. The options are: - **12db below nominal** - **9db below nominal** - **6db below nominal**

Table continues…

Name	Description
	• **3db below nominal** • **Nominal** • **3db above nominal** • **6db above nominal** • **9db above nominal**
Phone Mute Alert	Specifies whether mute alert feature is blocked. The options are: • **Unblocked** • **Blocked**
Extend Ringtone	Specifies whether extended ring tone feature is enabled.
Group Number	Specifies group numbers if available. The values are 0 to 99.
Minimum delay to backup volume level to PPM	Specifies the minimum time interval between backups of the volume levels to the PPM service when the phone is registered to Avaya Aura Session Manager. The value is 2 to 900 seconds.
Audio	
Call Progress Tone Country	Specifies the country of operation.
AGC Handset	Specifies the Automatic Gain Control setting for the handset interface. The options are: • **Disable** • **Enable**
AGC Speaker	Specifies the Automatic Gain Control setting for the speaker. The options are: • **Disable** • **Enable**
Handset Sidetone Level	Specifies the level of side tone in the handset. The options are: • **Normal level** • **Three levels softer than Normal** • **Off** • **One level softer than Normal** • **Two levels softer than Normal**

Table continues…

Name	Description
	- **Four levels softer than Normal** - **Five levels softer than Normal** - **Six levels softer than Normal** - **One level louder than Normal** - **Two levels louder than Normal**
Ringtone Style	Specifies the style of classic ring tone to be used. The options are: - **North America** - **European**
Dialing	
No Digit Dial Timer	Specifies the number of seconds that the telephone waits for a digit to be dialed after going off-hook and before generating a warning tone. The valid range is 0 to 60 seconds.
Inter-digit Wait Timer	Specifies the number of seconds that the telephone waits after a digit is dialed before sending a SIP INVITE. The valid range is 1 to 10 seconds.
Dial Local Area Code	Specifies whether the user must dial the area code of calls within the same area code. The options are: - **No** - **Yes**
Local Area Code	Specifies the local area code of the phone.
Enhanced Local Dialing Rules	
Enable Local Dialing Rules	Specifies whether to process telephone numbers from the incoming Call Log or Contacts while dialling a number. The options are: - **Disable** - **Enable Without Contacts** - **Enable With Contacts**
Country Code	Specifies the country code of the user.
International Access Code	Specifies the international access code.
Long Distance Access Code	Specifies the long distance access code.
Internal Extension Number Length	Specifies the length of an internal extension number. The valid range is 3 to 13 numeric digits.

Table continues…

Name	Description
National Telephone Number Length	Specifies the length of a national telephone number. The valid range is 5 to 15 numeric digits.
Outside Line Access Code	Specifies the pre-fixed number to be used to make a local call by using a public network.
Remove PSTN access prefix from outgoing number	Specifies the removal of the PSTN access prefix from the outgoing number. The options are: • **No** • **Yes**
Admin	
Admin Access allowed from Phone	Specifies whether the admin access is allowed from the phone. The options are: • **No** • **Yes**
Admin Login fail attempt allowed	Specifies the number of failed attempts to enter the admin access code before the admin login is locked. The options are 1 to 20.
Admin Login Locked Time after fail attempt	Specifies the time interval in minutes to re-enter the admin access code after the admin login is locked. The value is 5 to 1440 minutes.

Related links

Configuring settings on page 46

Configuring certificates

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Certificates**.
3. Configure the following areas:
    - Certificates
    - Online Certificates Status Protocol (OCSP)
    - SCEP
    - PKCS12
    - Web Server

4. Click one of the following:
    - **Save**
    - **Reset to Default**
    - **Help**

Related links

Certificates field descriptions on page 54

Certificates field descriptions

Name	Description
Certificates	
Available Trusted Certificate	Specifies the file names of certificates for authentication.
Upload Trusted Certificate	Specifies the trusted certificate used by the phone. You can also browse and upload the certificates from your local machine by clicking **Choose File** > **Import**.
Trusted Certificates file to upload	Specifies the name of the certificate file to be uploaded.
Match Identity to trust certificate	Specifies the status of the TLS server identification . The options are: use the choices tag • **Yes: Identification is required.** • **No: Identification is not required.**
Server Certificate re-check hours	Specifies the time interval in hours for rechecking the expiration and revocation status of the certificates used to establish any existing TLS connections. The valid range is 0 to 32767.
Warning on number of days before Certificate expiration	Specifies the number of days before the expiration of a certificate that a warning must first appear on the phone screen.
FQDN IP Mapping	Specifies to validate an FQDN contained in the certificate when an IP address is used to establish the connection. The parameter is a comma-separated list of names or value pairs where the name is an FQDN and the value is an IP address.
Online Certificate Status Protocol (OCSP)	
Enable OCSP	Specifies the status of OSCP. The options are: • **Disable** • **Enable**
Action on Unknown Revocation Status	Specifies whether a certificate is authenticated when its revocation status cannot be determined.

Table continues…

Name	Description
	The options are: - **Certificate revocation operation is accepted** - **Certificate is considered to be revoked and TLS connection is closed**
Nonce in OCSP Request	Specifies whether a nonce is included in OCSP requests and expected in OCSP responses. The options are: - **Do not add** - **Add**
OCSP Address	Specifies a URI for an OCSP responder. The URI can be an IP address or a host name.
OCSP Address Preferred	Specifies the preferred OCSP responder URI. The options are: - **Use OCSP address configured first and then OCSP field of AIA extension of the certificate being checked** - **Use OCSP field of AIA extension of the certificate being checked first and then OCSP address configured**
OCSP Trusted Certificates	Specifies the trusted OCSP certificates to be downloaded. It also acts as a separate trusted certificate repository for the OCSP Trusted Responder Model and contains certificates that the OCSP responder can trust. This value is required if the OCSP responder uses a different CA for the server certificate than the root CA.
OCSP Hash Algorithm	Specifies the hashing algorithm for an OCSP request. value operation. discuss The options are: - **SHA-1** - **SHA-256**
Use OCSP Caching	Specifies whether OCSP caching is in use. The options are: - **Yes** - **No**
OCSP Cache Expiry	Specifies the time interval for the OCSP cache expiry in minutes. The valid range is 60 to 10080.
SCEP	
SCEP Server	Specifies the URL address of the SCEP server.
Common Name	Specifies the common name for the subject in an SCEP certificate request.

Table continues…

Name	Description
Subject	Specifies the part of SUBJECT in an SCEP certificate request that is common for requests from different device. For example, Organizational Unit, Organization, Location, State, and Country.
CA Identifier	Specifies the Certificate Authority Identifier. Certificate Authority servers may require a specific CA Identifier string to accept GetCA requests. If the device works with such a Certificate Authority, the CA identifier string can be set through this parameter.
Initiate renewal on % of Validity Interval	Specifies the percentage used to calculate the renewal time interval out of the device certificate's Validity Object. If the renewal time interval has elapsed, the phone starts to contact the SCEP server periodically to renew the certificate. The range is 0 to 99.
Phone behavior on Pending request	Specifies the functioning of the device when performing certificate enrolment. The options are: • **Poll SCEP server periodically in background** • **Wait until a certificate is received or rejected**
SCEP Password	Specifies a challenge password to use with SCEP.
PKCS12	
PKCS12 Address	Specifies the IPv4 or IPv6 URL address, or FQDN from where a PKCS#12 file is to be downloaded.
PKCS12 Password Retry Count	Specifies the number of attempts allowed for password entry.
Available Identity Certificate	Specifies the trust certificates used as trust points for TLS connections.
Upload Identity Certificate	Specifies the trust certificates to be uploaded.
Delete Installed Identity Certificate	Allows you to delete any installed identity certificate.

Related links

Configuring certificates on page 53

Configuring SIP settings

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **SIP**.
3. Configure the following areas:
    - SIP Account

  - SIP Server
  - Codecs and DTMF
  - RTP
  - SRTP
  - Timers and Count
  - Local Port
  - Miscellaneous

4. Click one of the following:
   - **Save**
   - **Reset to Default**
   - **Help**

Related links

SIP settings field descriptions on page 57

SIP settings field descriptions

Name	Description
SIP Account	
Status	Displays the SIP account status. The field is automatically populated. The values are: • Not Configured • Not Registered • Registered
SIP User ID	Specifies the SIP user ID provided by the service provider. You can also type the SIP user ID, which is a combination of the following values: • Upper and lower case characters • Numbers from 0 to 9 • Spaces • Special characters
Authentication User ID	Specifies the authentication ID. You can also type the authentication user ID in this field if authentication is enabled on the SIP server.

Table continues…

Name	Description
	The authentication user ID is a combination of the following values: • Upper and lower case characters • Numbers from 0 to 9 • Spaces • Special characters
Authentication Password	Specifies the authentication password. You can also type the password in this field if authentication is enabled on the SIP server. Note that the password can contain maximum 31 ASCII characters.
SIP Server	
SIP Domain	Specifies the SIP domain used for SIP registration. Valid values are 0 to 255 ASCII characters.
Enable PPM as source of Proxy Server	Specifies whether PPM is used as a source of SIP proxy server information. The options are: • **Yes**: The phone uses the PPM server information. • **No**: The phone does not use the PPM server information.
Use Proxy Server	Specifies whether SIP proxy servers are read-only or can be edited. The options are: • **Manual**: To configure SIP proxy server manually by using the phone or the web interface. • **Automatic**: To use the SIP proxy server settings received from the `46xxsettings.txt` file or PPM.
SIP Proxy Server (Manual)	Specifies the SIP proxy server domain. Valid values are 0 to 255 ASCII characters.
SIP Proxy Server (Automatic)	Specifies the SIP proxy server settings as received from the `46xxsettings.txt` file or PPM.
Register simultaneous to Proxy Server	Specifies whether the phone registers simultaneously to a proxy server. The options are: • **Simultaneous** • **Alternate**
Number of proxy server to register simultaneously	Specifies the number of SIP proxy controllers that the phone can register simultaneously. The options are: • **1**

Table continues…

Name	Description
	- **2** - **3**
Registration Interval	Specifies the time interval between two registrations to the SIP proxy. The default value is 900 seconds. Valid values are from 30 to 86400 seconds.
Un-registration Wait Timer (seconds)	Specifies the time for which the phone waits before terminating all SIP dialog and SIP registrations. The default value is 32 seconds. Valid values are from 4 to 3600 seconds.
Registration Wait Timer (seconds)	Specifies the number of seconds the phone waits for a response message from registration. If no response message is received within this time, the phone tries to register again. The default value is 32 seconds. Valid values are from 4 to 3600 seconds.
Codecs and DTMF	
OPUS	Specifies whether the OPUS codec capability of the phone is enabled or disabled. The options are: - **Disabled** - **Enabled WIDEBAND_20K** - **Enabled NARROWBAND_16K** - **Enabled NARROWBAND_12K**
G.722	Specifies whether the G.722 codec is enabled. The options are: - **Disable** - **Enable**
G.726	Specifies whether the G.726 codec is enabled. The options are: - **Disable** - **Enable**
G.729	Specifies whether the G.729A codec is enabled. The options are: - **Disable** - **Enable**

Table continues…

Name	Description
G.711u law	Specifies whether the G.711u law codec is enabled. The options are: • **Disable** • **Enable**
G.711a law	Specifies whether the G.711a law codec is enabled. The options are: • **Disable** • **Enable**
Send DTMF	Specifies whether the phone sends DTMF tones in-band as regular audio, or out-of-band using RFC 2833 procedures. The options are: • **In-band** • **Out-of-band**
OPUS Payload	Dynamically specifies the RTP payload type to be used for OPUS codec. The parameter is used when the media request is sent to the far-end in an INVITE or 200 OK when INVITE with no Session Description Protocol (SDP) is received. Valid values are from 96 to 127.
G.726 Payload	Specifies the RTP payload type to be used for the G.726 codec. The default value is 110. Valid values are from 96 to 127.
DTMF Payload	Specifies the RTP payload type to be used for RFC 2833 signaling. The default value is 120. Valid values are from 96 to 127.
RTP	
Play Tone till RTP	Specifies whether the locally generated ringback tone stops when SDP is received for an early media session, or whether it continues until RTP is actually received from the far-end party. The options are: • **Yes** • **No**
Symmetric RTP	Specifies whether the phone must receive RTP if the UDP source port number is not same as the UDP destination port number. The options are: • **Disable**

Table continues…

Name	**Description**
	- **Enable** The default value is Enable. tag
RTCP_XR	Specifies whether VoIP Metrics Report Block as defined in RTP Control Protocol Extended Reports (RTCP XR) (RFC 3611) is sent as part of the RTCP packets to a remote peer or an RTCP monitoring server. The options are: - **Yes** - **No** The default value is No.
SRTP	
Media Encryption	Specifies the crypto suite and session parameters for media encryption. The options are: - **aescm128-hmac80** - **aescm128-hmac32** - **aescm128-hmac80-unauth** - **aescm128-hmac32-unauth** - **aescm128-hmac80-unenc** - **aescm128-hmac32-unenc** - **aescm128-hmac80-unenc-unauth** - **aescm128-hmac32-unenc-unauth** - **none** - **aescm256-hmac80** - **aescm256-hmac32** **Note:** You should not use unauthenticated media encryption (SRTP) files.
Encrypt RTCP	Specifies whether RTCP packets are encrypted or not. The options are: - **Yes**: SRTCP is enabled. - **No**: SRTCP is disabled. The default value is No.
Enforce "SIPS" URI for SRTP	Specifies whether a SIPS URI must be used for SRTP. The options are: - **Yes**: Enforced

Table continues…

Name	Description
	- **No**: Not enforced. The default value is Yes.
SDP Negotiation Capability	Specifies the Session Description Protocol (SDP) negotiation capability. - **Yes** - **No** The default value is Yes.
Timers and Count	
SIP Timer T1	Specifies an estimate for the Round Trip Time (RTT). Valid values are from 500 to 10000 milliseconds. The default value is 500 milliseconds.
SIP Timer T2	Specifies the maximum retransmit interval for non-INVITE requests and INVITE responses. Valid values are from 2000 to 40,000 milliseconds. The default value is 4000 milliseconds.
SIP Timer T4	Specifies the maximum duration for which a message remains in the network. Valid values are from 2500 to 10,000 milliseconds. The default value is 5000 milliseconds.
INVITE Response Timeout	Specifies the maximum number of seconds that the phone waits for another response after receiving a SIP 100 Trying response. Valid values are from 30 to 180 seconds. The default value is 60.
Failed Session Removal Timer	Specifies the time to automatically remove a failed call session. Valid values are from 5 to 999 seconds. The default value is 60 seconds.
Outbound Subscription Duration Request	Specifies the Outbound subscription request duration. Valid values are from 60 to 31,53,600 seconds. The default value is 86,400 seconds.
Controller Search Interval	Specifies the time that the phone waits to complete the maintenance check for monitored controllers. Valid values are from 4 to 3600 seconds. The default value is 16 seconds.
Active subscription wait time for "avaya-cm-feature-status"	Specifies the time that the phone waits to validate an active subscription when it subscribes to the avaya-cm-feature-status package. Valid values are from 16 to 3600 seconds.

Table continues…

Name	Description
	The default value is 60 seconds.
Remote Data Source initial retry time	Specifies the number of seconds that the phone waits for the first time before trying to contact the PPM server again after a failed attempt. Each subsequent retry is delayed by double the previous delay time. Valid values are from 2 to 60 seconds. The default value is 2 seconds.
Remote Data Source maximum retry time	Specifies the maximum delay interval after which the phone stops to contact the PPM server. Valid values are from 2 to 3600 seconds. The default value is 600 seconds.
Remote Data Source initial retry attempts	Specifies the number of attempts the PPM adaptor must try to download from PPM before it stops connecting to the PPM server. Valid values are from 1 to 30. The default value is 15.
Local Port	
RTP Port (minimum)	Specifies the lower limit of a port range to be used by the following connections: • RTP • RTCP • SRTP • SRTCP Valid values are from 1024 to 65,003. The default value is 5004.
RTP Port (range)	Specifies the port range to be used by the following connections: • RTP • RTCP • SRTP • SRTCP Valid values are from 32 to 64,511. The default value is 40.
SIP Signaling Port (minimum)	Specifies the lower limit of a port range to be used for SIP signaling. Valid values are from 1024 to 65,003. The default value is 5060.
SIP Signaling Port (range)	Specifies the port range to be used for SIP signaling. Valid values are from 32 to 64,511.

Table continues…

Name	Description
	The default value is 32.
Miscellaneous	
Conference Factory URI	Specifies the URI for Avaya Aura® Conferencing or network conferencing in third-party call control environments. The value contains 0 to 255 ASCII characters.
Subscribe Event Packages	Specifies a comma-separated list of event packages to subscribe to after registration. Possible values are: • reg • dialog • mwi • ccs • message-summary, which is identical to mwi • avaya-ccs-profile, which is identical to ccs For IP Office, you must use the following: • reg • message-summary, which is identical to MWI • avaya-ccs-profile, which is identical to CCS For a third-part call control setup, you can use message-summary.
Voice Mail Access Code	Specifies the number to access the voice mail in a non-Avaya environment.
100rel	Specifies whether the 100rel option tag is included in the SIP INVITE header field. The options are: • **Disable**: The tag is not included. • **Enable**: The tag is included.
Validate Incoming messages	Specifies whether AOR received in Request-URI of an incoming call must be validated with the contact header published by phone during registration. The options are: • **Disable** • **Enable**

Related links

Configuring SIP settings on page 56

Debugging

Procedure

1. Log in to the web interface as an administrator.
2. In the navigation pane, click **Debugging**.
3. Configure the fields in the following areas:
    - Log
    - SNMP
    - RTCP Monitoring
    - Phone Report
    - SSH
    - SLA Monitor
    - Other
4. Click **Save**.
5. **(Optional)** Click **Generate Phone Report**.
6. **(Optional)** To ping, do the following:
    a. In **Ping**, enter the IP address that you want to ping.
    b. Click **Ping Test**.

Related links

Debugging field descriptions on page 65

Debugging field descriptions

Name	Description
Log	
Logging	Specifies the logging status. The options are: • **Off** • **On** The default value is **Off**.
Log Server	Specifies the IP or DNS address of the Syslog server. The value contains 0 to 255 ASCII characters.
Log Level	Specifies the severity level of the syslog messages. Events with the selected severity level and above are logged.

Table continues…

Name	Description
	The options are: • **Emergencies** • **Alerts** • **Critical** • **Errors** • **Warnings** • **Notices** • **Information** • **Debug** The default value is **Emergencies**.
Log Categories	Specifies the list of log categories. need more info
Enhanced Debugging	Specifies the status of enhanced debugging . The options are: • **Enable** • **Disable**
SNMP	
SNMP String	Specifies the SNMP community name string. The string contains maximum 32 ASCII characters.
SNMP Address	Specifies the IP addresses for SNMP queries. The address contains maximum 255 ASCII characters.
RTCP Monitoring	
RTCP Monitor Address	Specifies the IP or DNS address of the RTCP monitor. The address contains maximum 255 ASCII characters.
RTCP Monitor Port	Specifies the RTCP monitor port number. Valid values are 0 through 65535 ASCII characters. The default value is 5005.
RTCP Monitoring Report Period	Specifies the interval for sending out RTCP monitoring reports. Valid values are 5 through 30 seconds. The default value is 5 seconds.
Phone Report	
Maintenance Server Address	Specifies the file server address to send the phone report. The address contains maximum 255 ASCII characters.
SSH	

Table continues…

Name	Description
SSH Allowed	Specifies whether Secure Shell (SSH) is supported. The options are: • **Enable** • **Disable**
SSH Idle Timeout	Specifies the time after which SSH is disabled. discuss The options are: • **Enable** • **Disable** • **Configured using local craft procedure** The default value is **Disable**.
SSH Banner File	Specifies the file name or URL for a custom SSH banner file. The file contains maximum 255 ASCII characters.
EASG site certificates	Specifies a list of EASG site certificates. Support technicians use these certificates to generate EASG responses for SSH login without access to the Avaya network. The certificate contains maximum 64 ASCII characters. You can add maximum four certificates.
EASG site Authentication Factor code	Specifies the Site Authentication Factor code associated with the EASG site certificate installed. Valid values are 10 through 20 alphanumeric characters.
Days before EASG certificates expiration warning	Specifies the number of days before the expiration of EASG product certificate that a warning message appears on the phone screen. Valid values are 90 through 730. The default value is 365.
SLA Monitor	
Enable SLA Monitor Agent	Specifies the status of the SLA Monitor Agent . The options are: • **Enable** • **Disable** The default value is **Disable**.
SLA Monitor Server Address	Specifies the IP address of the SLA Monitor server in the aaa.bbb.ccc.ddd format. The IP address must contain: • Numbers 0 to 9 • 3 dots

Table continues…

Name	Description
Packet Capture (sniffing)	Specifies whether the SLA Monitor agent supports packet capture. The options are: • **Disable** • **Enable with payloads removed from RTP packets** • **Enable with payloads included in RTP packets** • **Controlled from Admin Menu** The default value is **Disable**.
Device Control	Specifies whether the SLA Monitor agent supports device control. The options are: • **Disable** • **Enable** • **Controlled from Admin Menu** The default value is **Disable**.
Device Performance Monitoring	Specifies whether the SLA Monitor agent supports access to phone performance data. The options are: • **Enable** • **Disable** The default value is **Disable**.
UDP Port for discovery and test messages	Specifies the port used to receive packets from an SLA Monitor server. Valid values are 6000 through 65535. The default value is 50011.
Other	
Serial Port	Specifies if the port for network traffic is enabled or disabled. The options are: • **Enable** • **Disable** The default value is **Disable**.
Port Mirroring	Specifies the status of port mirroring. Port mirroring is used to monitor network traffic. The options are: • **Off** • **On** The default value is **Off**.

Related links

Debugging on page 65

Chapter 5: Configuring servers and VLAN

Server configuration

To install Avaya J100 Series IP Phones, you must configure the following servers:

- HTTP or HTTPS File Server: To download and save the software distribution package and the settings file. Examples of a File Server:
  - Apache
  - Internet Information Services (IIS)
  - Avaya Utility Server
- DHCP server: To dynamically assign IP addresses and provide device configuration parameters.

Related links

File Server configuration on page 70
DHCP server configuration on page 77

File Server configuration

A file server is an HTTP or an HTTPS server that is required to download and save the software distribution package and the `Settings` file.

On restarting, the phone checks for software updates and `Settings` files on the specified file servers.

You can provide the file server addresses to phones through one of the following methods:

- DHCP
- LLDP
- Administration menu on the phone
- Settings file

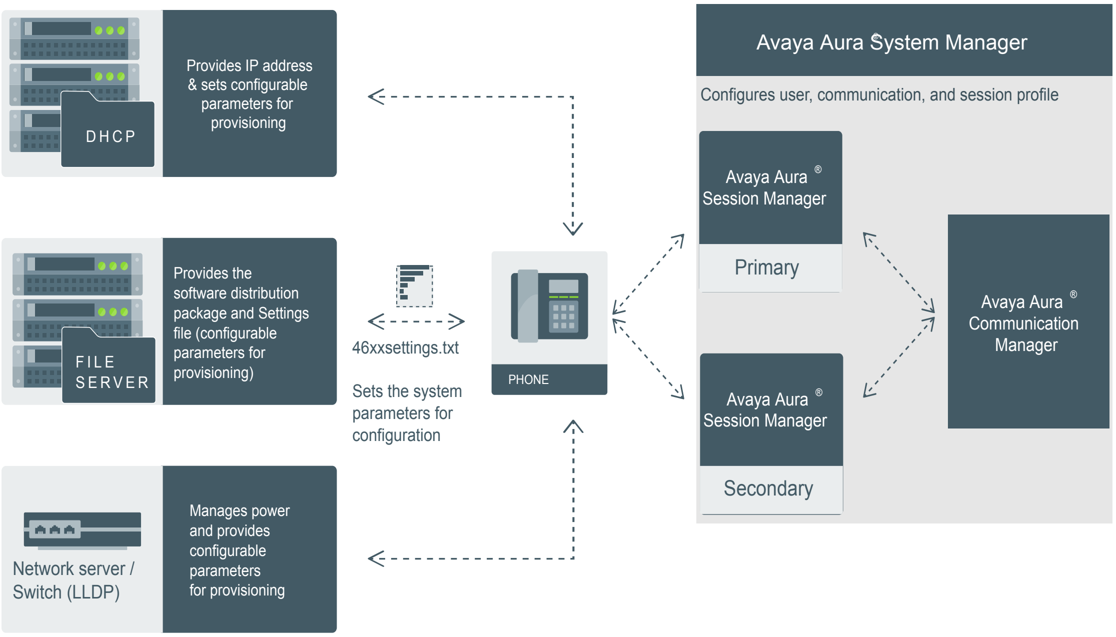

Provides IP address & sets configurable parameters for provisioning
DHCP
Provides the software distribution package and Settings file (configurable parameters for provisioning)
FILE SERVER
46xxsettings.txt
Sets the system parameters for configuration
PHONE
Network server / Switch (LLDP)
Manages power and provides configurable parameters for provisioning
Avaya Aura System Manager
Configures user, communication, and session profile
Avaya Aura ® Session Manager
Primary
Avaya Aura ® Session Manager
Secondary
Avaya Aura ® Communication Manager

Figure 1: Diagram: Phone setup in Avaya Aura® environment

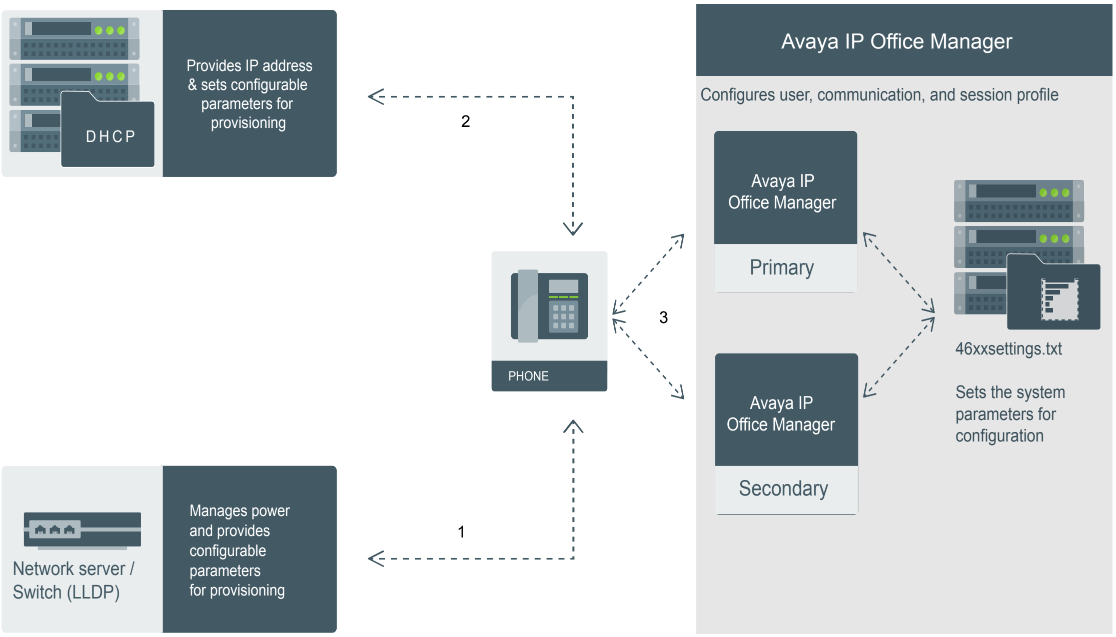

DHCP
Provides IP address & sets configurable parameters for provisioning
2
Avaya IP Office Manager
Configures user, communication, and session profile
Avaya IP Office Manager
Primary
3
PHONE
46xxsettings.txt
Sets the system parameters for configuration
Avaya IP Office Manager
Secondary
Network server / Switch (LLDP)
Manages power and provides configurable parameters for provisioning
1

Figure 2: Diagram: Phone setup in IP Office environment

Related links

- Server configuration on page 70
- Setting up a file server on page 72
- Software distribution package on page 73
- Downloading and saving the software on page 74
- Contents of the settings file on page 74
- Configuring the Settings file on page 76

Setting up a file server

About this task

Use this procedure to configure a file server. The file server is used to download and store distribution packages and settings files.

Procedure

1. Install the HTTP or HTTPS server according to the server vendor's instruction.
2. Download the software distribution package and the 46xxsettings.txt settings file.
3. Extract the distribution package and save the extracted files and the 46xxsettings.txt settings file on the file server.

Related links

File Server configuration on page 70

Software distribution package

Note:

For any new software release, ensure that you download the latest software distribution package and read any Product Support Notices (PSNs) associated with the new release. Both are available on the Avaya support website

Review the release notes and any Read Me files associated with a distribution package.

Ensure that the `Settings` file is not cached in your browser. To do this, clear the browser cache before downloading the `Settings` file from the Avaya support Web site, so that you don't get an old version.

Software distribution package containing the files needed to operate the Avaya J100 Series IP Phones are packaged together in a ZIP format. You can download the package from the Avaya support website.

Note:

From IP Office R 10.0 SP3 or later, the software distribution package for the Avaya J100 Series IP Phones is part of the IP Office admin CD.

SIP software distribution package contains:

- One or more software files
- One upgrade file (`J100Supgrade.txt`)
- Language files. For example, `Mlf_J129_BrazilianPortuguese.xml`, `Mlf_J129_Chinese.xml`.
- Files av_prca_pem_2033.txt and av_sipca_pem_2027.txt that contain a copy of the Avaya Product Root Certificate Authority certificate in PEM format that may be downloaded to phones based on the value of the TRUSTCERTS parameter.
- File named release.xml that is used by the Avaya Software Update Manager application. Avaya Software Update Manager upgrades and maintains firmware for Avaya managed devices.

Note:

Settings files are not included in the software distribution packages because they would overwrite your existing files and settings.

Two configuration files that are important to understand are as follows:

- The upgrade file, `J100Supgrade.txt` that tells the phone whether the phone needs to upgrade software. The phones attempt to read this file whenever they reset. The upgrade file is also used to point to the `Settings` file.
- The Settings file, `46xxsettings.txt`, that contains the option settings that enable, disable, or otherwise customize the settings you might need to tailor the phones for your enterprise. IP Office auto generates the Settings file (`J100settings.txt`).

Related links

File Server configuration on page 70

Downloading and saving the software

Before you begin

Ensure that your file server is set up.

Procedure

1. Go to the Avaya Support website.
2. In the **Enter Your Product Here** field, enter Avaya J100 Series IP Phones .
3. In the **Choose Release** field, click the required release number.
4. Click the **Downloads** tab.

   The system displays a list of the latest downloads.
5. Click the appropriate software version.

   The system displays the Downloads page.
6. In the **File** field, click the zipped file and save the file on the file server.
7. Extract the zipped file and save it at an appropriate location on the file server.
8. From the latest downloads list, click the settings file.

   The system displays the Downloads page.
9. In the **File** field, click the settings file and save the file at an appropriate location on the file server.

Related links

File Server configuration on page 70

Contents of the settings file

The settings file can include any of the six types of statements, one per line:

- Tags, which are lines that begin with a single "#" character, followed by a single space character, followed by a text string with no spaces.
- **Goto** commands, of the form **GOTO *tag.* Goto** commands cause the phone to continue interpreting the settings file at the next line after a ***# tag*** statement. If no such statement exists, the rest of the settings file is ignored.
- Conditionals, of the form **IF *$parameter_name* SEQ *string* GOTO *tag***. Conditionals cause the **Goto** command to be processed if the value of the parameter named ***parameter_name*** exactly matches ***string***. If no such parameter named ***parameter_name*** exists, the entire conditional is ignored. The only parameters that can be used in a conditional statement are: GROUP, MACADDR, MODEL and MODEL4.
- **SET** commands, of the form **SET *parameter_name value***. Invalid values cause the specified value to be ignored for the associated ***parameter_name*** so the default or previously administered value is retained. All values must be text strings, if the value itself is numeric, you must place the numeric value inside a pair of quotation marks. For example, "192.x.y.z"

- Comments, which are statements with characters "**##**" in the first column.
- GET commands, of the form ***GET filename.*** The phone attempts to download the file named by ***filename***, and if it is successfully obtained, it will be interpreted as an additional settings file, and no additional lines will be interpreted in the original file. If the file cannot be obtained, the phone will continue to interpret the original file.

The Avaya-provided upgrade file includes a line that tells the phones to **GET *46xxsettings.txt***. This line cause the phone 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 phone continues processing the upgrade script file. Also, if the settings file is successfully obtained but this does not change any settings, the phone continues to use HTTP.

The settings file is under your control and is where you can identify non-default option settings, application-specific parameters, etc. You can download a template for this file from the Avaya support Web site.

During a reboot, if the phone is unable to access the settings file, it does not retain the values of all the parameters. For more information on which parameter value is retained, see the following table.

Parameter	Retained
AGCHAND	Y
AGCHEAD	Y
AGCSPKR	Y
APPNAME	N
AUDIOENV	N
AUDIOSTHD	N
AUDIOSTHS	N
AUTH	Y
BAKLIGHTOFF	Y
CNGLABEL	Y
DAYLIGHT_SAVING_SET TING_MODE	Y
DHCPSTD	N
HEADSYS	N
HOMEIDLETIME	N
LOG_CATEGORY	Y
LOGSRVR	N
LOCAL_LOG_LEVEL	Y
LANG0STAT	Y
MSGNUM	N
PROCSTAT	Y
PROCPSWD	Y

Table continues…

Parameter	Retained
PHY1STAT	Y
PHY2STAT	Y
PHNCC	N
PHNDPLENGTH	N
PHNIC	N
PHNLDLENGTH	N
PHNLD	N
PHNLAC	Y
PHNOL	N
RFSNAME	N
SNMPADD	Y
SNMPSTRING	Y
SIG	Y
SCREENSAVERON	N
TEAM_BUTTON_RING_T YPE	Y
TPSLIST	N
VLANTEST	Y

Related links

File Server configuration on page 70

Configuring the Settings file

About this task

Use this procedure to modify the `Settings` file with appropriate values to provision the device configuration parameters.

✱ **Note:**

This procedure applies to Avaya Aura® environment only. In IP Office the settings file is autogenerated and cannot be modified.

Procedure

1. On the file server, go to the location where you downloaded the `46xxsettings.txt` settings file.
2. Open the `Settings` file in a text editor.
3. Set the required parameters.

Note:

Avaya J100 Series IP Phones parameters stored for a particular user are not reflected in other phones, for example, 9600 Series IP Deskphones, even if the SIP user is the same.

4. Save the `Settings` file.

Related links

File Server configuration on page 70
Phone installation with DES on page 22
Diagram: Phone deployment with DES on page 24
List of configuration parameters on page 143

DHCP server configuration

Configure the DHCP server to:

- Assign IP addresses dynamically to Avaya J100 Series IP Phones.
- Provision the phone and site-specific configuration parameters through various DHCP options.

Related links

Server configuration on page 70
Setting up a DHCP server on page 77

Setting up a DHCP server

Procedure

1. Install the DHCP server according to the DHCP server vendor’s instructions.
2. Configure the available range of IP addresses.
3. Configure the required DHCP options.

Related links

DHCP server configuration on page 77

Configuration through LLDP

Link Layer Discovery Protocol (LLDP) is an open standards, layer 2 protocol that IP phones use to advertise their identity and capabilities and to receive administration from Ethernet switches. LAN equipment can use LLDP to manage power and administer VLANs, DSCP, and 802.1p priority fields.

The transmission and reception of LLDP is specified in IEEE 802.1AB-2005. The use Type-Length-Value (TLV) elements specified in IEEE 802.1AB-2005, TIA TR-41 Committee - Media

Endpoint Discovery (LLDP-MED, ANSI/TIA-1057), and Proprietary elements. LLDP Data Units (LLDPDUs) are sent to the LLDP Multicast MAC address.

The running SIP software support IEEE 802.1AB if the value of the configuration parameter LLDP_ENABLED is “1” (On) or “2” (Auto). If the value of LLDP_ENABLED is “0” (off), the transmission and reception of Link Layer Discovery Protocol (LLDP) is not supported. When the value of LLDP_ENABLED is “2”, the transmission of LLDP frames does not begin until an LLDP frame is received. The first LLDP frame is transmitted within 2 seconds after the first LLDP frame is received. After transmission begins, an LLDPDU is transmitted every 30 seconds. A delay of up to 30 seconds in phone initialization might occur if the file server address is delivered by LLDP and not by DHCP.

These phones do not transmit 802.1AB multicast LLDP packets from an Ethernet line interface to the secondary line interface and vice versa.

By using LLDP, you can configure the following:

- Call server IP address
- File server
- PHY2VLAN
- L2QVLAN and L2Q

LLDPDU transmitted by the phones

Category	TLV Name (Type)	TLV Info String (Value)
Basic Mandatory	Chassis ID	IPADD of phone, IANA Address Family Numbers enumeration value for IPv4, or subtype 5:Network address.
Basic Mandatory	Port ID	MAC address of the device.
Basic Mandatory	Time-To-Live	120 seconds.
Basic Optional	System Name	The Host Name sent to the DHCP server in DHCP option 12.
Basic Optional	System Capabilities	Bit 2 (Bridge) will be set in the System Capabilities if the phone has an internal Ethernet switch. If Bit 2 is set in Enabled Capabilities then the secondary port is enabled.
Basic Optional	Management Address	Mgmt IPv4 IP address of device. Interface number subtype = 3 (system port). Interface number = 1. OID = SNMP MIB-II sysObjectID of the device.
IEEE 802.3 Organization Specific	MAC / PHY Configuration / Status	Reports auto negotiation status and speed of the uplink port on the device.
TIA LLDP MED	LLDP-MED Capabilities	Media Endpoint Discovery capabilities = 00-33 (Inventory, Power-via-MDI, Network Policy, MED Caps).

Table continues…

Category	TLV Name (Type)	TLV Info String (Value)
TIA LLDP MED	Network Policy	Tagging Yes/No, VLAN ID for voice, L2 Priority, DSCP Value.
TIA LLDP MED	Inventory – Hardware Revision	MODEL - Full Model Name.
TIA LLDP MED	Inventory – Firmware Revision	Firmware version.
TIA LLDP MED	Inventory – Software Revision	Software version or filename.
TIA LLDP MED	Inventory – Serial Number	Device serial number.
TIA LLDP MED	Inventory – Manufacturer Name	Avaya.
TIA LLDP MED	Inventory – Model Name	MODEL with the final Dxxx characters removed.
Avaya Proprietary	Call Server IP address	Call Server IP Address. Subtype = 3.
Avaya Proprietary	IP Phone addresses	Phone IP address, Phone Address Mask, Gateway IP Address. Subtype = 4.
Avaya Proprietary	File Server	File Server IP Address. Subtype = 6.
Avaya Proprietary	802.1Q Framing	802.1Q Framing = 1 if tagging or 2 if not.
Basic Mandatory	End-of-LLDPDU	Not applicable.

TLV impact on system parameter values

System parameter name	TLV name	Impact
PHY2VLAN	IEEE 802.1 Port VLAN ID	The value of the PHY2VLAN parameter on the phone is configured from the value of the Port VLAN identifier in the TLV.
L2QVLAN and L2Q	IEEE 802.1 VLAN Name	The value is changed to the TLV VLAN Identifier. L2Q is set to 1 (ON). A check is made as to whether a reset is necessary to obtain a new IP address due to a change in the values of the parameters L2Q or L2QVLAN. VLAN Name TLV is ignored if: • The value of USE_DHCP is 0 and the value of IPADD is not 0.0.0.0. • The current value of L2QVLAN was set by a TIA LLDP MED Network Policy TLV.

Table continues…

System parameter name	TLV name	Impact
		• The VLAN name in the TLV does not contain the substring “voice” in lower-case, upper-case or mixed-case ASCII characters anywhere in the VLAN name.
L2Q, L2QVLAN, L2QAUD, DSCPAUD	TIA LLDP MED Network Policy (Voice) TLV	L2Q - set to 2 (off) if T (the Tagged Flag) is set to 0 and to 1 (on) if T is set to 1. L2QVLAN - Set to the VLAN ID in the TLV. L2QAUD - Set to the Layer 2 Priority value in the TLV. DSCPAUD - Set to the DSCP value in the TLV. A check is made as to whether a reset is necessary to obtain a new IP address due to a change in the values of the parameters L2Q or L2QVLAN. This TLV is ignored if: • The value of USE_DHCP is 0 and the value of IPADD is not 0.0.0.0. • The Application Type is not 1 (Voice) or 2 (Voice Signaling). • The Unknown Policy Flag (U) is set to 1.
L2Q, L2QVLAN	TIA LLDP MED Network Policy (Voice Signaling)	L2Q - set to 2 (off) if T (the Tagged Flag) is set to 0 and to 1 (on) if T is set to 1. L2QVLAN - Set to the VLAN ID in the TLV. L2QAUD - Set to the Layer 2 Priority value in the TLV. DSCPAUD - Set to the DSCP value in the TLV. A check is made as to whether a reset is necessary to obtain a new IP address due to a change in the values of the parameters L2Q or L2QVLAN. This TLV is ignored if: • The value of USE_DHCP is 0 and the value of IPADD is not 0.0.0.0. • The Application Type is not 1 (Voice) or 2 (Voice Signaling). • The Unknown Policy Flag (U) is set to 1.
SIP_CONTROLLER_LIST	Proprietary Call Server TLV	SIP_CONTROLLER_LIST will be set to the IP addresses in this TLV value. **Note:** This parameter cannot be used in an environment where both SIP phones and H.323 phones exist.
TLSSRVR and HTTPSRVR	Proprietary File Server TLV	

Table continues…

System parameter name	TLV name	Impact
L2Q	Proprietary 802.1 Q Framing	If the value of TLV = 1, L2Q is set to 1 (On). If the value of TLV = 2, L2Q is set to 2 (Off). If the value of TLV = 3, L2Q is set to 0 (Auto). A check is made as to whether a reset is necessary to obtain a new IP address due to a change in the values of the parameters L2Q or L2QVLAN. This TLV is ignored if: • The value of USE_DHCP is 0 and the value of IPADD is not 0.0.0.0. • The current L2QVLAN value was set by an IEEE 802.1 VLAN name. • The current L2QVLAN value was set by a TIA LLDP MED Network Policy (Voice) TLV.

Configuration through DHCP

The obtain network and configuration information using DHCP protocol. You can configure the DHCP server to provide the following information to the device:

- Avaya Aura® Session Manager address.
- IP address
- Subnet mask
- IP address of the router
- IP address of the HTTP or HTTPS file server
- IP address of the SNTP server
- IP address of DNS

You can configure the DHCP server to:

- Dynamically assign IP addresses to the .
- Provision device and site-specific configuration parameters through various DHCP options.

DHCP Site Specific Option

The phones support DHCP configuration option called Site Specific Option(SSON). Using this parameter, custom parameters can be configured on the phone through a DHCP server. In the DHCP DISCOVER, the phone requests for the DHCP Site-specific option (SSON), typically configured in DHCP Option 242. To configure and respond to this request, configure the DHCP server with proper data supplied in the offer for the value of this option. An example of such configuration is as follows:

`option avaya-option-242 L2Q=1,L2QVLAN=1212,httpsrvr=192.168.0.100.`

Following parameters can be configured with this feature:

Parameter	Description
ADMIN_PASSWORD	Specifies the security string used to access local procedures. The default is 27238. This is meant to replace PROCPSWD as it provides a more secure password syntax.
HTTPDIR	Specifies the path to prepend to all configurations and data files the device might request when starting up, that is, the path, relative to the root of the HTTP file server, to the directory in which the device configuration and date files are stored. The path may contain no more than 127 characters and may contain no spaces. HTTPDIR is the path for all HTTP operations. The command is `SET HTTPDIR=<path>`. In configurations where the upgrade and binary files are in the default directory on the HTTP server, do not use the `HTTPDIR=<path>`.
HTTPPORT	Sets the TCP port used for HTTP file downloads from non-Avaya servers. The default is 80.
HTTPSRVR	IP addresses or DNS names of HTTP file servers used for downloading settings, language, and firmware files during startup. The firmware files are digitally signed, so TLS is not required for security.
ICMPDU	Controls the extent to which ICMP Destination Unreachable messages are sent in response to messages sent to closed ports so as not to reveal information to potential hackers. The default is 1, that is sends Destination Unreachable messages for closed ports used by traceroute.
ICMPRED	Controls whether ICMP Redirect messages are processed. The default is 0, that is, redirect messages are not processed.
L2Q	802.1Q tagging mode. The default is 0 for automatic.
L2QVLAN	VLAN ID of the voice VLAN. The default is 0.
PHY1STAT	Specifies the speed and duplex settings for the Ethernet line interface. The default value is 1 for auto-negotiate.
PHY2STAT	Specifies the speed and duplex settings for the secondary (PC) Ethernet interface. The default value is 1.
PROCPSWD	Security string used to access local procedures. The default is 27238. ADMIN_PASSWORD replaces this parameter if ADMIN_PASSWORD is set in the `46xxsettings.txt` file.
REUSETIME	Time in seconds for IP address reuse timeout, in seconds. The default is 60 seconds.
SIP_CONTROLLER_LIST	SIP proxy or registrar server IP or DNS addresses that can be 0 to 255 characters, IP address in the dotted decimal name format, separated by commas and without any intervening spaces. The default is null, that is, no controllers.

Table continues…

Parameter	Description
TLSDIR	Used as path name that is prepended to all file names used in HTTPS GET operations during initialization. The string length can be from 0 to 127.
TLSPORT	Destination TCP port used for requests to https server in the range of 0 to 65535. The default is 443, the standard HTTPS port.
TLSSRVR	IP addresses or DNS names of Avaya file servers used to download configuration files. Firmware files can also be downloaded using HTTPS. **Note:** Transport Layer Security is used to authenticate the server.
VLANTEST	Number of seconds to wait for a DHCPOFFER on a non-zero VLAN. The default is 60 seconds.

In an IP Office environment `46xxsettings.txt` and `96x1Supgrade.txt` files are autogenerated. There is a provision where you can set up a different file server with your own custom Settings file.

DHCP options

You can configure the following options in the DHCP server:

Option	Description
Option 1	Specifies the subnet mask of the network.
Option 3	Specifies the gateway IP address list. The list can contain up to 127 total ASCII characters. Separate more than one IP address with commas with no intervening spaces.
Option 6	Specifies the DNS server address list. The list can contain up to 127 total ASCII characters. Separate more than one IP address with commas with no intervening spaces. The phone supports DNS and the dotted decimal addresses. The phone attempts to resolve a non-ASCII-encoded dotted decimal IP address by checking the contents of DHCP Option 6. At least one address in option 6 must be a valid, nonzero, dotted decimal address, otherwise the DNS address fails.
Option 12	Specifies the host name. `AVohhhhhh`, where: • `AV` stands for Avaya. • `o` is one of the following values based on Object Unique Identifier (OUI) derived from the first three octets of the phone MAC address: - A if OUI is 00-04-0D - B if OUI is 00-1B-4F - E if OUI is 00-09-6E - L if OUI is 00-60-1D

Table continues…

Option	Description
	- T if the OUI is 00-07-3B - X if the OUI is anything else • `hhhhhh` are the ASCII characters for the hexadecimal representation of the last three octets of the phone MAC address.
Option 15	Specifies the domain name. The domain name is required to resolve DNS names into IP addresses. Configure this option if you use a DNS name for the HTTP server. Otherwise, you can specify a domain as part of customizing the HTTP server. This domain name is appended to the DNS addresses specified in option 6 before the phone attempts to resolve the DNS address. The phone queries the DNS address in the order they are specified in option 6. If there is no response from an address, the phone queries the next DNS address. As an alternative to administering DNS by DHCP, you can specify the DNS server and domain name in the HTTP script file. If you use the script file, you must configure the DNSSRVR and DOMAIN parameters so that you can use the values of these parameters in the script. **Note:** Administer option 6 and option 15 appropriately with DNS servers and domain names respectively.
Option 42	Specifies the SNTP IP address list. List servers in the order of preference. The minimum length is 4 and the length must be a multiple of 4.
Option 43	Specifies the encapsulated vendor-specific options that clients and servers use to exchange the vendor-specific information. Option 43 is processed only if the first code in the Option is 1 with a value of 6889. The value 6889 is an Avaya enterprise number. All values are interpreted as strings of ASCII characters that are accepted with or without a null termination character. Any invalid value is ignored and the corresponding parameter value is not set.
Option 51	Specifies the DHCP lease time. If this option is not received, the DHCPOFFER is not accepted. Assign a lease time of six weeks or greater. If this option has a value of FFFFFFFF hex, the IP address lease is assumed to be infinite, so that the renewal and rebinding procedures are not necessary even if options 58 and 59 are received. Expired leases causes the device to reboot.
Option 52	Specifies the overload option. If this option is received in a message, the device interprets the sname and file parameters.
Option 53	Specifies the DHCP message type. The value can be one of the following: • 1 for DHCPDISCOVER • 3 for DHCPREQUEST For DHCPREQUEST sent to renew the device IP address lease: • If a DHCPACK is received in response, a log event record is generated with a Log Category of DHCP.

Table continues…

Option	Description
	• If a DHCPNAK is received in response, the device immediately ceases IP address usage, generates a log event record, sets IPADD to 0.0.0.0, and enters the DHCP INIT state.
Option 55	Specifies the parameter request list. Acceptable values are: • 1 for subnet mask • 3 for router IP addresses • 6 for domain name server IP addresses • 7 for log server • 15 for domain name • 42 for NTP servers
Option 57	Specifies the maximum DHCP message size. Set the value to 1500. Set the value to 1000.
Option 58	Specifies the DHCP lease renew time. If not received or if this value is greater than that for option 51, the default value of T1, renewal timer is used.
Option 59	Specifies the DHCP lease rebind time. If not received or if this value is greater than that for Option 51, the default value of T2, rebinding timer is used.
Option 242	Specifies the site-specific option. This option is optional. If you do not configure this option, ensure that one of the following parameters is configured appropriately elsewhere: • HTTPSRVR • TLSSRVR

DHCP vendor-specific option

You can set DHCP vendor-specific parameters by using DHCP option 43. The supported codes for Option 43 and the corresponding parameters are as follows:

Code	Parameter
1	Does not set any parameter. The value must be 6889.
2	HTTPSRVR
3	HTTPDIR
4	HTTPPORT
5	TLSSRVR
6	TLSDIR
7	TLSPORT
8	TLSSRVRID
9	L2Q

Table continues…

Code	Parameter
10	L2QVLAN
11	PHY1STAT
12	PHY2STAT
14	SIG
15	SIP_CONTROLLER_LIST

Extending use of DHCP lease

support configuration of network parameters to the phone using DHCP as per RFC 2131. However, when a DHCP server becomes unreachable and the DHCP lease currently held by the phone expires, the phones continues to use the same lease until the DHCP server becomes reachable. This feature is controlled with the help of configuration parameter, DHCPSTD, as explained:

Parameter name	Default value	Description
DHCPSTD	0	Specifies it will continue to use the expired DHCP lease. Value operation: • 0: Continue use of expired DHCP lease if the lease could not be renewed. • 1: Stop using DHCP lease immediately when it expires, as per standard. The parameter is configured through `46xxsettings.txt` file.

When this feature is enabled (DHCPSTD=1), the phone will continue to use the lease data, including IP address, router and other options if the lease could not be renewed. In this state, the phone will continue attempting to reach a DHCP server every 60 seconds. When a DHCP server becomes reachable and a lease is renewed or new lease obtained, the phone performs a duplicate address detection on the offered IP address. If no conflicts are detected, this IP address is assigned to the local network interface for use.

Parameter configuration through DHCP

Parameter	Set to
DHCP lease time	Option 51, if received
DHCP lease renew time	Option 58, if received
DHCP lease rebind time	Option 59, if received

Table continues…

Parameter	Set to
DOMAIN	Option 15, if received
DNSSRVR	Option 6, if received, which might be a list of IP addresses
HTTPSRVR	The siaddr parameter, if that parameter is non-zero
IPADD	The yiaddr parameter
LOGSRVR	Option 7, if received
MTU_SIZE	Option 26
NETMASK	Option 1, if received
ROUTER	Option 3, if received, which might be a list of IP addresses
SNTPSRVR	Option 42

Virtual LAN (VLAN) overview

VLANs provide a means to segregate your network into distinct groups or domains. They also provide a means to prioritize the network traffic into each of these distinct domains. For example, a network may have a Voice VLAN and a Data VLAN. Grouping devices that have a set of common requirements can greatly simplify network design, increase scalability, improve security, and improve network management. Therefore, you must always use VLANs in your network.

The networking standard that describes VLANs is IEEE 802.1Q. This standard describes, in detail, the 802.1Q protocol and how Ethernet frames get an additional 4 byte tag inserted at the beginning of the frame. This additional VLAN tag describes the VLAN ID that a particular device belongs to, and the priority of the VLAN tagged frame. Voice and video traffic typically get a higher priority in the network as they are subject to degradation caused by network jitter and delay.

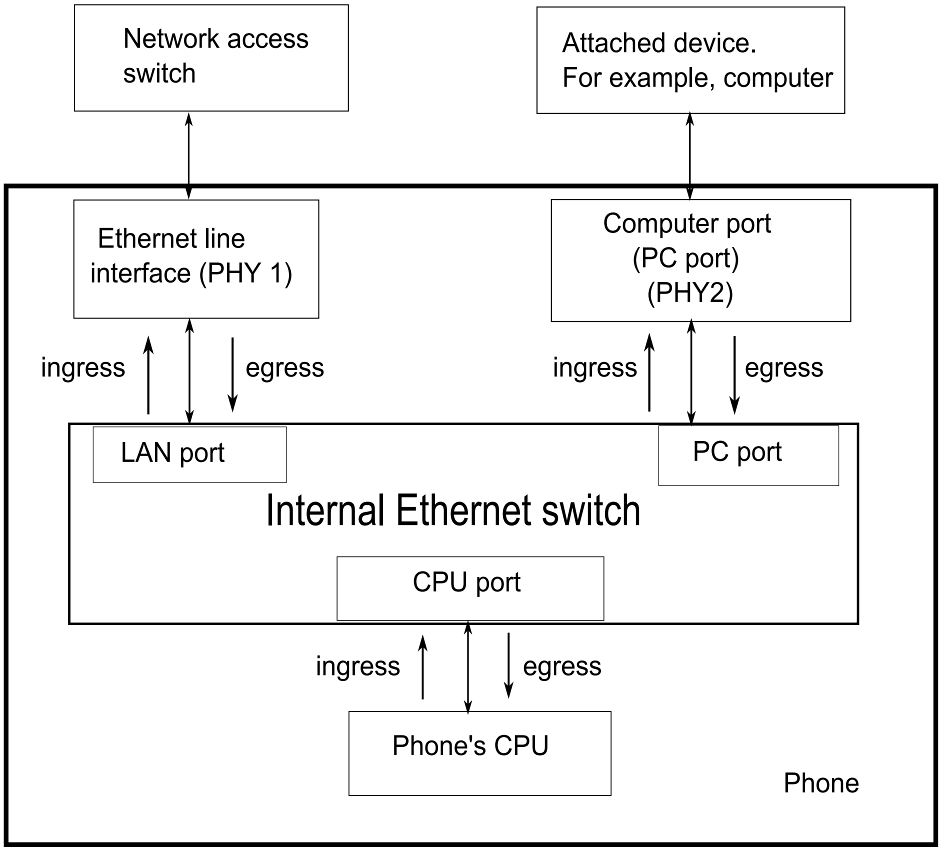
Related links

VLAN separation on page 87
External switch configuration on page 90
Exceptions to the VLAN forwarding rules on page 91
Special considerations on page 91
VLAN parameters on page 92

VLAN separation

The Avaya J100 Series IP Phones has an internal network switch that is capable of using VLANs to segregate traffic between the LAN port, the PC port and the internal port that goes to the CPU of the phone. You can have VLAN functionality on this switch and configure the switch to isolate the traffic destined for the CPU of the phone from the data destined to the PC port.

The configuration of the internal switch of the phone can be done through the `Settings` file, LLDP or DHCP. It is preferable to configure the VLAN settings on the internal switch of the phone through DHCP or LLDP as these protocols are run prior to, and during, network initialization. If that is not possible then the `Settings` file configuration parameters can be used and the VLAN can be started in automatic mode, which is the default mode.


Network access switch
Attached device. For example, computer
Ethernet line interface (PHY 1)
Computer port (PC port) (PHY2)
ingress
egress
ingress
egress
LAN port
PC port
Internal Ethernet switch
CPU port
ingress
egress
Phone's CPU
Phone

Related links

[Virtual LAN (VLAN) overview](#) on page 87
[VLAN separation modes](#) on page 90

VLAN separation modes

Avaya J100 Series IP Phones supports two VLAN separation modes:

- No VLAN separation mode: In this mode the CPU port of the port receives untagged frames and tagged VLAN frames on any VLAN irrespective of whether the phone sends untagged frames or tagged frames. This traffic can be received from the PC port or LAN port. The filtering of the frames is done by the CPU itself. In order to reduce unnecessary traffic to the CPU, the administrator should configure only the necessary VLANs on the external switch port, in particular, voice VLAN and data VLAN.
- Full VLAN separation mode: This is the default mode. In this mode the CPU port of the phone receives tagged frames with VLAN ID = L2QVLAN whether they are from the LAN port or PC port. The PC port receives untagged or tagged frames with VLAN ID = PHY2VLAN from the LAN port. The PC port cannot send any untagged frames or tagged frames with any VLAN ID, including the voice VLAN ID, to the CPU. Frames received externally on the PC port can only be sent to the LAN port if they are untagged frames or tagged frames with VLAN ID= PHY2VLAN. In this mode, there is a complete separation between CPU port and PC port. In order to configure Avaya J100 Series IP Phones to work in this mode all the following conditions must be met:
  - VLANSEPMODE = 1 (default)
  - L2Q = 0 (auto, default) or 1 (tag)
  - L2QVLAN is not equal to 0
  - PHY2VLAN is not equal to 0
  - L2QVLAN is not equal to PHY2VLAN
  - The phone actually sends tagged VLAN frames. This means that the DHCP server on voice VLAN (L2QVLAN) is reachable and the phone receives IP address on voice VLAN.

If one of these conditions is not met then the phone works in no VLAN separation mode where all kinds of traffic reaches the CPU port of the phone.

Note:

The phone can send tagged VLAN frames on the voice VLAN (L2QVLAN), but still not work in full VLAN separation mode. For example, when PHY2VLAN = 0 or VLANSEPMODE = 0.

Related links

[VLAN separation](#) on page 87

External switch configuration

Configure the following for the external switch port:

- Bind VLAN to the voice VLAN (L2QVLAN) and the data VLAN (PHY2VLAN). It is important to restrict the VLAN binding when in No VLAN separation mode. This is because there is no filtering by the internal phone switch and the CPU of the phone is subject to all the traffic going through the phone. When in Full VLAN separation mode, the internal phone switch will filter any tagged VLAN frames with VLANs other than voice VLAN (L2QVLAN) and data VLAN (PHY2VLAN) in any case. However, you must configure only the necessary VLANs on the external switch port.

- Set the default VLAN as the data VLAN (PHY2VLAN). This is the VLAN assigned by the external switch port to untagged frames received from phone LAN port.
- Configure one of the following for egress tagging:
    - Data VLAN is untagged and voice VLAN is tagged.
    - Data VLAN and voice VLAN are both tagged. You must configure this option to have Full VLAN separation.

Sending egress voice VLAN frames untagged from the external switch port to the phone LAN port means that there is no VLAN separation between the voice VLAN and data VLAN.

Related links

Virtual LAN (VLAN) overview on page 87

Exceptions to the VLAN forwarding rules

Exceptions to the VLAN forwarding rules are as follows:

- LLDP frames are always exchanged between the following in all VLAN separation modes:
    - The LAN port and CPU port
    - The CPU port and LAN port
- Spanning tree frames are always exchanged between the LAN port and PC port in all VLAN separation modes.
- 802.1x frames are always exchanged between the following in all VLAN separation modes according to DOT1XSTAT and DOT1X configuration:
    - The LAN and CPU port or PC port
    - The PC and CPU port or LAN port
    - The CPU port and LAN port

Related links

Virtual LAN (VLAN) overview on page 87

Special considerations

Special use of VLAN ID=0

The phone adds a VLAN tag to the egress voice frames with a VLAN ID=0 in certain configurations. For example, to utilize the priority functionality of the VLAN frame only and not the VLAN ID properties. In this case, use the parameter L2QAUD or L2QSIG to set the value of the VLAN priority portion of the VLAN tag.

Automatic failback of VLAN tagging

The phone connects to a network when the value of L2QVLAN does not match with the VLAN being assigned to the network access switch. When the phone starts to connect, it tries to contact the DHCP server with a VLAN ID=L2QVLAN. If the phone does not receive a DHCPOFFER with that particular VLAN ID, then it eventually fails back. The phone tries to contact the DHCP server again if the VLAN functionality of the phone is set to one of the following:

- L2Q=1: With a VLANID =0
- L2Q=0: Without any VLAN tag

The VLANTEST parameter determines how long the phone waits for a recognizable DHCPOFFER. If VLANTEST= 0, then the phone does not fail back and keeps sending DHCP requests by using tagged VLAN frames with VLAN ID = L2QVLAN.

VLAN support on the computer or PC port

In full VLAN separation mode, the phone only supports one VLAN on the computer port. In no VLAN separation mode, all VLANs pass between the LAN and PC ports. However, the CPU port receives all traffic even on VLANs that are not equal to L2QVLAN.

Related links

Virtual LAN (VLAN) overview on page 87

VLAN parameters

The following configuration parameters are used to configure VLAN functionality on the network switch internal to the phone.

Parameter name	Default value	Description
L2Q	0	Specifies the VLAN tagging is enabled or disabled. Value operation: • 0: Auto. VLAN tagging is turned on when the network can support VLAN tagging and L2QVLAN is non zero. • 1: On. VLAN tagging is turned on when the network can support VLAN tagging. The IP phone sends tagged frames with VLAN = L2QVLAN, even if L2QVLAN is set to 0. • 2: Off. VLAN functionality is disabled. L2Q is configured through: • Local admin procedure • A name equal to value pair in DHCPACK message • SET command in the `Settings` file

Table continues…

Parameter name	Default value	Description
		• DHCP option 43 • LLDP
VLANTEST	60	Specifies the number of seconds that the phone waits prior to failing back to a different VLAN ID if no response is received from the DHCP server. Valid values are 0 through 999. Value operation: • 0: The phone continues to attempt a DHCP REQUEST forever. VLANTEST is configured through: • `Settings` file • A name equal to value pair in DHCPACK message
VLANSEPMODE	1	Specifies whether the VLAN separation is enabled or disabled. Value operation: • 0: Disabled • 1: Enabled VLANSEPMODE is configured through the `Settings` file.
PHY2TAGS	0	Determines whether or not VLAN tags are stripped on Ethernet frames going out of the Computer (PC) port. Value operation: • 0: Strip tags. VLAN tags are stripped from Ethernet frames leaving the computer (PC) port of the phone. • 1: Does not strip tags. VLAN tags are not stripped from Ethernet frames leaving the Computer (PC) port of the phone. PHY2TAGS is configured through the `Settings` file.
L2QVLAN	0	Specifies the voice VLAN ID to be used by IP phones. Valid values are 0 through 4094. L2QVLAN is configured through: • Local admin procedure

Table continues…

Parameter name	Default value	Description
		- A name equal to value pair in DHCPACK message - SET command in the `Settings` file - DHCP option 43 - LLDP
PHY2VLAN	0	Specifies the value of the 802.1Q VLAN ID that is used to identify network traffic going into and coming out of the internal CPU of the phone. Valid values are 0 through 4094. PHY2VLAN is configured through: - SET command in the `Settings` file - LLDP
L2QAUD	6	Specifies the value of the VLAN priority portion of the VLAN tag when the phone generates tagged Ethernet frames from the internal CPU of the phone. These values are inserted into the VLAN tag for audio frames (RTP, RTCP, SRTP, SRTCP). All other frames except those specified by the L2QSIG parameter are set to priority 0. Valid values are 0 through 7. L2QAUD is configured through: - SET command in the `Settings` file - LLDP
L2QSIG	6	Specifies the value of the VLAN priority portion of the VLAN tag when the phone generates tagged Ethernet frames from the internal CPU of the phone. These values are inserted into the VLAN tag for signaling frames (SIP). All other frames except those specified by the L2QAUD parameter are set to priority 0. Valid values are 0 through 7. L2QSIG is configured through: - SET command in the `Settings` file - LLDP

Related links

Virtual LAN (VLAN) overview on page 87

IPv4 and IPv6 operation overview

- If IPV6STAT is set to 1, that is, IPv6 is supported, then the DHCPSTAT parameter is selected:
    - If DHCPSTAT is set to 1, that is, use DHCPv4 only, then IPv4 only is enabled.
    - If DHCPSTAT is set to 3, that is, both IPv4 and IPv6 supported, then dual-stack operation is enabled.

If IPv4-only operation is enabled, the system ignores any IPv6 addresses configured as parameter values and uses the next IPv4 address in the list. If the parameter value does not contain any IPv4 address, the system treats the value as null.

The phones in this release support the following combinations or IPv4 and IPv6 IP address configuration:

- Dual mode: Both IPv4 and IPv6 addresses are configured by using static addressing.
- Dual mode: Both IPv4 and IPv6 addresses are configured by using DHCP.
- IPv4 only mode.

The following table provides the results of the determination:

Table 1: IP Enablement Results

Manually programmed IPv4 address	IPV6STAT	Manually programmed IPv6 address	DHCPSTAT	Result	Addressing modes	
					IPv4	IPv6
No	0	NA	NA	IPv4 only	DHCP	NA
	1	No	1	IPv4 only	DHCP	NA
Yes	0	NA	NA	IPv4 only	Manual	NA
	1	No	1	IPv4 only	Manual	NA

Multiple Device Access (MDA)

Avaya J100 Series IP Phones support Multiple Device Access (MDA) with which you can simultaneously register up to 10 SIP devices for a single user.

With MDA, you can perform the following actions:

- Make and receive calls on any of the registered devices.
- Switch to another registered device during an active call.
- Bridge on to calls on multiple registered devices. Alert all other registered devices about an incoming call to your extension. When you answer a call on any registered device, the alert on all other devices stops. The other devices show indications of an active call on the same call appearance number.

- Be on multiple calls at the same time on different devices, but only one call on each device.

  For example, you can listen to a conference call on one device and answer an incoming call on a second device without putting the conference call on hold. The two calls appear on separate call appearances on all registered devices.
- Use conference and transfer features.

  When you bridge on to a call on other registered devices and start a transfer, the call drops from all devices after the transfer is complete.

For more information, refer to *Multi Device Access White Paper*, available for Session Manager on Avaya support site.

Related links

Shared control on page 97

Multi Device Access operation in dual-stack mode

When the phone is configured in the IPv4 and IPv6 dual-stack mode with Multi Device Access (MDA) support, the signaling address family is selected according to the order of precedence level. The settings are done in both `46xxsettings.txt` file and System Manager. The order of precedence is as follows:

- Phone through Administration menu settings
- Web user interface
- Avaya Aura® System Manager
- Settings File
- DHCP
- LLDP

If you log in with your extension on MDA2 during a call and the signaling address mode is different from that of MDA1, then a limited service icon momentarily displays on MDA2. MDA2 automatically switches its signalling address family to match MDA1.

Parameter	Description
SIP_CONTROLLER_LIST_2	Describes the list of SIP Proxy or Registrar servers separated by comma when the SIP device is configured for the dual-stack operation. Valid values are 0 to 255 characters in the dotted decimal or colon-hex format. The syntax is: `host[:port][;transport=xxx]`

Table continues…

Parameter	Description
	where, • Host: IP addresses in dotted-decimal format or hex format. • Port: (Optional) Port number. The default is 5060 for TCP and 5061 for TLS. • Transport: (Optional) Transport type and xxx is either TLS or TCP. The default value is TLS.
SIGNALING_ADDR_MODE	Describes the SIP registration over IPv4 or IPv6 and selects the preferred Avaya Aura® Session Manager for phones supporting the dual-stack mode. The Avaya Aura® Session Manager IP address is selected according to the parameter SIP_CONTROLLER_LIST_2. Valid values are: • 4: IPv4. This is the default value. • 6: IPv6

Shared control

With the shared control feature. the phones can be controlled from a soft phone client. The phone needs to be registered before establishing a shared control connection. To operate shared control, the value of SIP_CONTROLLER_LIST must be identical for the phone and the soft client. Depending on soft client implementation, a shared control session may not be established if multiple devices are registered to the same user at the same time, with the **`sc-enabled`** flag sent during registration.

Note:

SIP signaling must be set to TLS for the phone and the soft client. For security reasons, TCP Signaling with shared control is not supported.

Related links

Multiple Device Access (MDA) on page 95

Chapter 6: Avaya Aura configuration for phones

SIP phone administration on Communication Manager

The SIP-based calling features in the following table can be invoked directly on Avaya J100 Series IP Phones or using a feature button provisioned using Avaya Aura® Communication Manager. Communication Manager automatically processes other calling features such as call coverage, trunk selection using Automatic Alternate Routing (AAR), or Automatic Route Selection (ARS), Class Of Service/Class Of Restriction (COS/COR), and voice messaging.

Note:

- For more information, see *Avaya Aura® Communication Manager Feature Description and Implementation* and other Communication Manager administration documents at the Avaya Support website: http://support.avaya.com/
- For information about IP Office, see *Avaya IP Office™ Platform SIP Telephone Installation Notes*.

The Avaya SIP solution configures all SIP phones in Communication Manager as off-PBX station (OPS).

Feature	Survivable operation with third-party proxy	Normal operation with Communication Manager and Session Manager
3-Way Conferencing	Yes	No
Conference using conference server	—	Yes
Automatic Call Back/Cancel	—	Yes
Call Forward All Calls – on/off	Yes	Yes
Call Hold	Yes	Yes
Call Park and Unpark	—	Yes
Calling Party Number Block	—	Yes
EC500	—	Yes
Malicious Call Trace	—	Yes
Message Waiting Indication	MWI is not available. If the PSTN_VM_NUM parameter is	Yes

Table continues…

Feature	Survivable operation with third-party proxy	Normal operation with Communication Manager and Session Manager
	administered, users can gain to the voice mailbox.	
Mute alert	Yes	Yes
Presence	—	Yes
Send All Calls Enable/Disable	—	Yes
SSH support	Yes	Yes
Third Party Call Forward	—	Yes
Third Party Call Forward Busy Don’t Answer	—	Yes
Attended Transfer	Yes	Yes
Transfer upon hang-up	—	Yes

Administering emergency numbers

Set the PHNEMERGNUM configuration parameter in the settings file or in the Session Manager to assign a default emergency number. The phone automatically dials the configured number whenever a user presses the **Emerg** softkey on the Login screen, or the Phone screen, or when the user presses the **Yes** softkey on an Emergency Calling pop-up screen. The phone dials the emergency number even if the phone is locked or the user is not logged in. You must select the **Allow Unauthenticated Emergency Calls** field in System Manager so that users can dial the emergency number when the phone is not registered.

You can set up to 100 emergency numbers for the phones to dial. However, you must first configure the additional emergency numbers in System Manager. You can then use the parameter PHNMOREEMERGNUMS to specify these additional emergency numbers in the `46xxsettings.txt file` or in the Avaya Aura® System Manager.

Note:

When in failover, the Emergency Number must be provisioned on the SIP gateway or the user will not be able to dial it.

The local proxy routes emergency calls from a user at a visited phone so that the local emergency number is called. When PHNEMERGNUM is administered, using the **Emerg** softkey overrides the SPEAKERSTAT parameter setting or a user-selected preferred audio path. This means that even if the Speakerphone is disabled, it becomes the default path when the user presses the **Emerg** softkey.

When the phone is locked or when the user is not logged in, it is possible to configure phones to make emergency calls. Depending upon the configuration parameters and whether or not the SIP proxy supports emergency dialing, it is possible to enable this functionality in the overall SIP solution.

Avaya J100 Series IP Phones displays an **Emerg** softkey when the phone is not registered or when the phone is locked. When the **Emerg** softkey is pressed, the user can call a primary emergency number. There are three parameters associated with this emergency dialing:

- PHNEMERGNUM: Specifies the primary emergency number that a user calls when the **Emerg** sofkey is pressed. Also, by specifying the PHNEMERGNUM parameter a user can dial the emergency number manually.
- ENABLE_SHOW_EMERG_SK: Specifies whether the phone displays Emerg softkey when the phone is registered and whether the phone displays a confirmation dialogue box when **Emerg** softkey is pressed.
- ENABLE_SHOW_EMERG_SK_UNREG: Specifies whether the phone displays Emerg softkey when the phone is not registered and whether the phone displays a confirmation dialogue box when **Emerg** softkey is pressed.

In Avaya J100 Series IP Phones you can set up to 100 additional emergency numbers to dial. You can define the numbers using the following parameter:

- PHNMOREEMERGNUMS: Specifies the additional emergency phone numbers.

In the Avaya Aura® environment, you can configure the parameters in System Manager. You must select the **Allow Unauthenticated Emergency Calls** field in System Manager so that users can dial the emergency number when the phone is not registered. However, when a user logs into an Avaya Aura® environment, only the emergency numbers configured in SMGR will be used by the phone. If the parameters are configured in the `Settings` file, the phone can access the emergency phone numbers when the Aura proxy servers are not available.

Note:

- When in failover, the Emergency Number must be provisioned on the SIP gateway or the user will not be able to dial it.
- The local proxy routes emergency calls from a user at a visited phone so that the local emergency number is called. When PHNEMERGNUM is administered, using the **Emerg** softkey overrides the SPEAKERSTAT parameter setting or a user-selected preferred audio path. This means that even if the Speakerphone is disabled, it becomes the default path when the user presses the **Emerg** softkey.
- When you toggle between server environments, for example, changing from Avaya Aura environment to third-party call control, you must reset the phone to the default values.
- In an IP Office environment, the auto-generated `Settings` file does not configure the **Emerg** soktkey on the phone. User has to manually dial the emergency number.

SIP phone administration on Session Manager

Avaya J100 Series IP Phones might display a prompt asking for the extension and password during the administration on Avaya Aura® Session Manager. The phones use the extension and password to communicate with Session Manager, which communicates with Avaya Aura® Communication Manager.

For more information, see the following documents at the Avaya Support website: http://support.avaya.com/

- For information about the Communication Manager administration with Session Manager, see the following Session Manager and Avaya Aura® System Manager documents:
    - *Avaya Aura® Session Manager Overview and Specification*
    - *Deploying Avaya Aura® Session Manager*
    - *Upgrading Avaya Aura® Session Manager*
    - *Administering Avaya Aura® Session Manager*
    - *Maintaining Avaya Aura® Session Manager*
    - *Troubleshooting Avaya Aura® Session Manager*
    - *Avaya Aura® Session Manager Case Studies*
    - *Deploying Avaya Aura® System Manager on System Platform*
    - *Deploying Avaya Aura® System Manager*
    - *Upgrading Avaya Aura® System Manager on System Platform*
    - *Upgrading Avaya Aura® System Manager*
    - *Administering Avaya Aura® System Manager*
    - *Avaya Aura® System Manager Release Notes*
    - *Administering Avaya IP Office™ Platform with Manager*
    - *Avaya IP Office™ Platform Solution Description*
    - *Avaya IP Office™ Platform Feature Description*

About controllers

A controller is a proxy server that routes the calls. A controller, such as Avaya Aura® Session Manager or IP Office, also works as a registrar and an interface between Communication Manager and phones.

Chapter 7: Security

Security overview

Avaya J100 Series IP Phones provide several updated security features. For example:

SIP-based Avaya J100 Series IP Phones provides several updated security features. When the phone is in a locked state, a user can only receive calls or make emergency calls. User logs and data are protected with the user account.

The following security features are available:

- Account management: The phone supports the following:
    - Storage of passwords and user credentials using Federal Information Processing Standards (FIPS 140–2)
    - FIPS 140-2 cryptographic algorithms for application, processes, and users
    - Control to toggle between FIPS and non-FIPS modes
    - Identity certificate installation using Simple Certificate Enrollment Protocol (SCEP) for enrollment and encrypted PKCS#12 file format to import both private key and certificate.
- Certificate management: The phone supports the following:
    - X509v3 compliant certificates
    - Public Key Infrastructure (PKI) for users who use third-party certificates for all Avaya services including database
    - Online Certificate Status Protocol (OCSP) for obtaining the revocation status of an X.509 digital certificate according to RFC 6960
- Department of Defense solution deployment with Joint Inter-operability Test Command (JITC) compliance.
- VLAN separation mode using system parameters.
- Synchronization of the system clock at configured intervals using system parameters.
- Display of SSH fingerprint in the Administration menu.
- Display of SSH fingerprint in the Administration menu.
- Display of OpenSSH and OpenSSL version in the Administration menu.
- Display of OpenSSH and OpenSSL version in the Administration menu.

- Maintenance of integrity when the phone is under Denial of Service (DoS) attack. In this case, the phone goes into out-of-service mode.
- DRBG random number generator compliant with SSL FIPS 140–2.
- SHA2 hash algorithm and strong encryption (256 bit symmetric and RSA 2048 and 4096 bit asymmetric keys) for all cryptographic operations.
- Deprecated support for SHA1 algorithms in all cryptographic algorithms.
- SRTP/SRTCP and TLS v1.2.

  SRTP is used to encrypt and secure the audio going to and from the phone. You must configure equivalent parameters in Communication Manager or System Manager. You must configure the following three parameters on the phones and equivalent Communication Manager parameters must match one of the parameters:

  - SET ENFORCE_SIPS_URI 1
  - SET SDPCAPNEG 1
  - SET MEDIAENCRYPTION X1, X2, 9. Valid values for X are 1 to 8 for aescm128-hmac80 , and 10 or 11 for aescm256-hmac80

Note:

- The Administration menu provides access to certain administrative procedures on the phone. You must change the default password for the Administration menu to restrict users from using the administrative procedures to change the phone configuration.
- Remote access to the phone is completely disabled by default.
- You should not use unauthenticated media encryption (SRTP) files.

Access control and security

Phones provide the following security features for control and access:

Security event logging

Logs are maintained for the following events:

- Successful and failed logins, username lockouts, and registration and authorization attempts by users and administrators.
- Change in roles.
- Firewall configuration changes.
- Modification or access to critical data, applications, and files.

Private Key storage

The phone stores the private key in PKCS#12 and PEM file formats. The phone sends the device identity certificate and a private key along with the encrypted password to the WPA supplicant. EAP-MD5 password is sent to the WPA supplicant securely.

Temporary Data

The phone deletes any temporary storage data from the program, variables, cache, main memory, registers, and stack.

IP information

The phone enables the user to see the IP information on the phone screen.

The parameter PROVIDE_NETWORKINFO_SCREEN controls the display of this information.

OpenSSH/OpenSSL version

The phone displays the version of OpenSSL and OpenSSH on the VIEW screen in the Administration menu. This information is displayed when the parameter DISPLAY_SSL_VERSION is set to `1`.

SSH Fingerprint

The phone displays SSH fingerprint to manually verify that an SSH connection is established with the correct phone.

Time synchronization

The phone synchronizes the time with the configured NTP servers at intervals. The parameter SNTP_SYNC_INTERVAL checks the time interval for synchronization any time between 60 to 2880 minutes with 1440 as the default setting

- Default: 1440 minutes
- 60–2880 minutes

Certificate management

Certificates are used to establish secure communication between network entities. Server or mutual authentication can be used to establish a secure connection between a client and server. The client always validates the certificate of the server and maintains a trust store to support this validation. If the server additionally requires mutual authentication, it requests an identity certificate from the client. The identity certificate must be provided and validated by the server to establish mutual authentication. Server must validate the identity certificate to establish a secure connection..

Phones support three types of certificates:

- Trusted certificates
- Online Certificate Status Protocol (OCSP) trust certificates
- Phone identity certificates

The Trusted and OCSP trust certificates are root or intermediate Certification Authority (CA) certificates that are installed on the phone through the `46xxsettings.txt` file.

Enhancements for installing identity certificates:

- SCEP over HTTPS is supported for enrollment.

- PKCS#12 file format is supported for installation.

To check the number of days remaining for Identity certificate expiry, use the parameter CERT_WARNING_DAYS . The user is notified through a log message if the log level is maintained as WARNING with the category CERTMGMT. The logs are maintained and displayed if SYSLOG is enabled.

MIB object tables and IDs are created for certificates installed on the phone. You can view the certificate attributes through an SNMP MIB browser.

Phone identity certificates

Identity certificates are used to establish the identity of a client or server during a TLS session. Phones support the installation of an identity certificate using one of the following methods:

- Secure Certificate Enrollment Protocol (SCEP) by using the `46xxsettings.txt` file parameter MYCERTURL.

  ```
  SET MYCERTURL "http://192.168.0.1/ejbca/publicweb/apply/scep/pkiclient.exe"
  ```

- PKCS12 File by using the `46xxsettings.txt` file parameter PKCS12URL

  ```
  SET PKCS12URL http://192.168.0.1/client_$MACADDR_cert.p12
  ```

Note:

If both MYCERTURL and PKCS12URL are provided in the `46xxsettings.txt` file, then PKCS12URL takes precedence over MYCERTURL.

The attributes of an identity certificate can be viewed by using a MIB browser. The following MIB OIDs can be used for this query:

Attribute Name	MIB OID
Serial Number	endptIdentityCertSN
Subject	endptIdentityCertSubjectName
Issuer	endptIdentityCertIssuerName
Validity	endptIdentityCertValidityPeriod
Thumbprint	endptIdentityCertFingerprint
Subject Alt Name	endptIdentityCertSubjectAlternativeName
Key Usage Extension	endptIdentityCertKeyUsageExtensions
Extended Key Usage	endptIdentityCertExtendedKeyUsage
Basic Constraints	endptIdentityCertBasicContraints

Server certificate validation

A server always provides a server certificate when the phone initiates a SIP-TLS or HTTPS connection.

To validate the identity of a received server certificate, the phone verifies the following:

- The certificate chain up to the trusted certificate authority in TRUSRCERTS
- The Signature
- The Revocation status through OCSP if OCSP_ENABLED is set to 1
- Certificate validity based on the current date and not-before and not-after attributes of the certificate.
- Certificate usage restrictions.
- The Identity of the server certificate that is used to connect to the server. This is optional and depends on the value of TLSSRVRID.

The following configuration parameter can be used in this context when applicable:

Parameter name	Default value	Description
TLSSRVRID	1	Specifies how a phone evaluates a certificate trust . The options are: • 0: Identity matching is not performed. • 1: The certificate is trusted only if the identity used to connect to the server matches the certificate identity, as per Section 3.1 of RFC 2818. For SIP-TLS connections, an additional check is performed to validate the SIP domain identified in the certificate, as per RFC 5922. The parameter is configured through the `46xxsettings.txt`.

Server certificate identity validation is only performed when TLSSRVRID is set to 1. When it is enabled, the phone verifies the identity contained in the server certificate. The TLS connection fails if any aspect of identity validation fails.

All TLS connections, that is, SIP-TLS and HTTPS-TLS, verify that the identity is contained in the server certificate. The server identity that is used for verification is the address that is used to connect to the server. This might be one of the following:

- IPv4 adress. For example, 192.168.1.2
- IPv6 address. For example, 2001:db8::2:1
- FQDN. For example, hostname.domain.com

This identity must match an identity found in the certificate. The matching is case insensitive. The phone first checks for the server identity in the Subject Alternative Name (SAN). If it cannot be found in the SAN, then the phone checks the certificate common name (CN). This validation is based on RFC 2818.

The phone checks for an IP address server identity match with the following in the specified order until a match is found:

1. Field of type IP address in the SAN extension

2. Full content of one field in the CN

The phone checks for a FQDN server identity match with the following in the specified order until a match is found:

1. Field of type DNSName in the SAN extension. An exact match of the full string is required. For example, host.subdomain.domain.com does not match subdomain.domain.com.
2. Full content of one field in the CN using the same rules as DNSName in SAN.

Note:

Identities containing a wildcard are not supported and do not match. For example, *.domain.com in the certificate will not match a connection to hostname.domain.com.

In addition, all SIP-TLS connections also verify that the SIP domain configured on the phone is present in the SIP server certificate as per RFC 5922.

The phone checks for a SIP domain match with the following in the specified order until a match is found:

1. Field of type URI in the SAN extension.
2. Field of type DNSName in the SAN extension and there is no URI field in the list of SAN extensions.
3. Full content of one field in the CN and there is no URI field in the list of SAN extensions.

Note:

Only full matches are allowed. For example, a configured SIP domain of sipdomain.com will not match a SAN DNSName containing proxy1.sipdomain.com.

Trusted certificates

Trusted certificates are root certificates of the certificate authority that issued the server or client identity certificates in use. These certificates are installed on the phones through the HTTP server and are used to validate server certificates during a TLS session.

System Manager includes EJBCA, an open source PKI Certificate Authority, that can be used to issue and manage client and server certificates.

OCSP trust certificates

Online Certificate Status Protocol (OCSP) is used to check the certificate revocation status of an x509 certificate in use. The phone trusts the OCSP server and installs its CA certificates. These certificates are called OCSP Trust Certificates.

OCSP Trust Certificates are installed in the same way as those for System Manager. However, OCSP Trust Certificates use a different parameter name called OCSP_TRUSTCERTS. This parameter follows the same format as that for TRUSTCERTS.

Configuration for secure installation

For secure installation, configure the following parameters.

Parameter	Set to	Notes
TRUSTCERTS		Provides the file names of certificates to be used for authentication. It supports both root and intermediate certificates and can contain up to six certificate files.
TLSSRVRID	1	Certificates installed on the servers must have the common name that matches the device configuration.
AUTH	1	Ensures usage of HTTPS file servers for configuration and software files download. Once AUTH is set to 1 and the device downloads the trusted certificates, the device can only download files from HTTPS server with certificates that can be validated using trusted certificate repository.
SSH_ALLOWED	0	To keep SSH disabled.

SCEP parameters

Configure the following Simple Certificate Enrollment Protocol (SCEP) parameters.

The SCEP parameters are not supported in IP Office environment.

Parameter	Type	Default value	Description
MYCERTURL	String	Null	Specifies the URL to access Simple Certificate Enrollment Protocol (SCEP) server. The device attempts to contact the server only if this parameter is set to other than its default value.
MYCERTCN	String	$SERIALNO	Specifies the Common name (CN) for SUBJECT in SCEP certificate request. The values can either be $SERIALNO or $MACADDR. If the value includes the string $SERIALNO, that string will be replaced by the phones serial number. If the value includes the string $MACADDR, that string will be replaced by the phones MAC address.
MYCERTDN	String	Null	Specifies common part of SUBJECT in SCEP certificate request. This value defines the part of SUBJECT in a certificate request including Organizational Unit, Organization, Location, State, and Country that is common for requests from different devices.
MYCERTKEYLEN	Numeric	2048	Specifies the private key length in bits to be created in the device for a certificate enrollment. The range is from 1024 to 2048.
MYCERTRENEW	Numeric	90	Specifies the percentage used to calculate the renewal time interval out of the device certificate’s Validity Object.

Table continues…

Parameter	Type	Default value	Description
			If the renewal time interval has elapsed the phone starts to periodically contact the SCEP server again to renew the certificate. The range is from 1 to 99.
MYCERTWAIT	Numeric	1	Specifies the behavior of the device when performing certificate enrolment. assign one of the following values: • 0: Periodical check in the background • 1: Wait until a certificate or a denial is received or a pending notification is received
MYCERTCAID	String	CAIdentifier	Specifies the Certificate Authority Identifier. Certificate Authority servers may require a specific CA Identifier string in order to accept GetCA requests. If the device works with such a Certificate Authority, the CA identifier string can be set through this parameter.
SCEPPASSWORD	String	$SERIALNO	Specifies a challenge password to use with SCEP. The value of SCEPPASSWORD, if non-null, is included in a challengePassword attribute in SCEP certificate signing requests. If the value contains $SERIALNO, $SERIALNO is replaced by the value of SERIALNO. If the value contains $MACADDR, $MACADDR is replaced by the value of MACADDR without the colon separators.

Chapter 8: Phone administration and configuration

Accessing the Admin menu during phone startup

Before you begin

Ensure you set the following parameters in the `Settings` file:

- PROCSTAT: To administer the phone using admin menu, set the parameter to zero.
- PROCPSWD or ADMIN_PASSWORD: The default password is `27238`. You must change the default password at the time of initial installation.

Procedure

1. Press **Main Menu** softkey.
2. On the Access code screen, enter the admin menu password using the dialpad.
3. Press **Enter**.

Parameters for managing Admin menu

Parameter name	Default value	Description
PROCSTAT	0	Specifies whether Admin menu is used for device configuration. Value operation: • 0: Specifies that the phone is administered through Admin menu. • 1: Specifies that the phone is not administered through Admin menu.
PROCPSWD	27238	Specifies an authentication code for accessing Admin menu. Value operation: • 27238: Specifies that the authentication code `27238` is set for accessing Admin menu.

Table continues…

Parameter name	Default value	Description
		• ASCII numbers between 0–7: Specifies an administrator configured authentication code. You must provide at least four ASCII numbers. • Null: Specifies that no authentication code is required to access Admin menu.
ADMIN_PASSWORD	27238	Specifies an authentication code for accessing Admin menu. When the parameter ADMIN_PASSWORD is set, then the parameter PROCPSWD is not used. You must provide an authentication code by using the any of the following combinations: • Numeric (0–9) • Alphabet in upper case (A-Z) • Alphabet in lower case (a-z) • Special characters ✱ **Note:** • PROCPSWD supports only numeric values. ADMIN_PASSWORD supports both alphanumeric and special characters. Hence, for enhanced security, use ADMIN_PASSWORD instead of PROCPSWD. • You can set the PROCPSWD and the ADMIN_PASSWORD in either `46xxsettings.txt` file or Avaya Aura® System Manager. However, ADMIN_PASSWORD is supported on Avaya Aura® System Manager 7.1.0 and later.
ADMIN_LOGIN_ATTEMPT_ALLOWED	10	Specifies the allowed number of failed attempts for accessing the Admin menu for a duration as specified in the parameter. Valid values are between 1 to 20.
ADMIN_LOGIN_LOCKED_TIME	10 minutes	Specifies the duration for lockout when a user reaches the maximum attempts limit for accessing the Admin menu. Valid values are between 5 to 1440 minutes.

Accessing the Admin menu after log in

Procedure

1. Press **Main Menu** > **Administration**.

2. In the **Access code** field, enter the administration password.
3. Press **Enter**.

Accessing the Ethernet IPv4 settings

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the administration password.
3. Press **Enter**.
4. Select **IP Configuration** > **Ethernet IPv4**.

   The phone displays the parameters for IP configuration.

IP configuration field description

Configuration Parameter Name	Description
The following parameters are available in IPv4 menu:	
Use DHCP	Specifies the access to view or manually enter the IP address. Select one of the following: • **YES**: Selects the DHCP option to view the IP addresses. • **No**: Selects the DHCP option to enter the IP addresss.
Phone	Specifies the IP address of the phone. The available format is `nnn.nnn.nnn.nnn.`
Router	Specifies the router IP address. The available format is `nnn.nnn.nnn.nnn.`
Mask	Specifies the network mask. The available format is `nnn.nnn.nnn.nnn.`
The following parameters are available in VLAN menu:	
802.1Q	Choose one of the following options: • **Auto**: Automatic mode. • **On**: Turns on the configuration. • **Off**: Turns off the configuration.

Table continues…

Configuration Parameter Name	Description
VLAN ID	Specifies the ID for VLAN. The available format is `dddd`.
VLAN Test	Specifies the time in seconds, the phone waits for the DHCP server response. The available format is `ddd`.
The following parameters are available in Servers menu:	
HTTP server	Specifies the IP address of the HTTP file server. The available format is `nnn.nnn.nnn.nnn`.
HTTPS server	Specifies the IP address of the HTTPS file server. The available format is `nnn.nnn.nnn.nnn`.
DNS server	Specifies the IP address of the DNS server. The available format is `nnn.nnn.nnn.nnn`.
SNTP server	Specifies the time server settings.

Using the debug mode

About this task

Use this procedure to activate or deactivate the debugging options.

Before you begin

You must set a HTTP server in the BRURI parameter in the `Settings` file that is capable of receiving a phone report from the phone. BRURI parameters can receive only phone report. It has no effect on any other debugging setting.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **Debug**.

   The phone displays the following debug options:

   - **Phone Report**
   - **Serial port mode**
   - **Port mirroring**
   - **Port Mirroring**
   - **SSH access**
   - **SSH fingerprint**

    - **Clear SSH lockout**
    - **Service mode control**
    - **Service mode record**
5. Use the appropriate keys to enable or disable the options.
6. Press **Save**.

Setting the Ethernet interface control

Procedure

1. Press **Main Menu** > **Admin**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Use the **Down Arrow** to select **Network interface**.
5. Use the **Right Arrow** key to change **Network mode** to **Ethernet** and do one of the following settings:
    - **Network config**: To change the network configuration to either Auto or Manual.
    - **Ethernet**: To change the Ethernet setting, go to step 6.
    - **PC Ethernet**: To change the PC Ethernet setting, go to step 7.
6. Use the **Right Arrow** key or the **Change** softkey to change the Ethernet setting to one of the following:
    - **Auto**
    - **10Mbps half**
    - **10Mbps full**
    - **100Mbps half**
    - **100Mbps full**
7. Use the **Right Arrow** key or the **Change** softkey to change the PC Ethernet setting to one of the following:
    - **Auto**
    - **10Mbps half**
    - **10Mbps full**
    - **100Mbps half**
    - **100Mbps full**
    - **Disabled**

8. Press **Save**.

Group identifier

A group identifier is a number assigned to a particular community of IP phone users in an organization. The group identifier number can be a number from 0 to 999 and the default number is 0.

With a group identifier, you can provide administration settings to each phone used by different communities of end users. For example, you might want to group users by time zones or work activities.

You can configure group identifier from the phone UI as a local administration process.

Related links

Setting the group identifier on page 115

Setting the group identifier

About this task

Use this procedure to set or change the group identifier only if the LAN Administrator instructs you to do so.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **Group**.
5. Enter any Group value between 0 to 999.

   When you change the Group value, the phone restarts after you exit the admin menu.
6. Press **Save**.

Related links

Group identifier on page 115

Setting event logging

Procedure

1. Press **Main Menu** > **Administration**.

2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **Log**.
5. Use the **Right** and **Left Arrow** keys to select one of the following settings associated with the corresponding SYSLOG_LEVEL:
    - **Emergencies**: SYSLOG_LEVEL=0
    - **Alerts**: SYSLOG_LEVEL=1
    - **Critical**: SYSLOG_LEVEL=2
    - **Errors**: SYSLOG_LEVEL=3
    - **Warnings**: SYSLOG_LEVEL=4
    - **Notices**: SYSLOG_LEVEL=5
    - **Information**: SYSLOG_LEVEL=6
    - **Debug**: SYSLOG_LEVEL=7
6. Press **Save**.

Administering enhanced local dialing

Phones automatically prepend a number from the incoming call log or from web pages with a digit to dial an outside number. This feature is called enhanced local dialing (ELD). For example, if you get a call from an international number and want to call back, the phone determines the number to be called and prepends the number to get an outside line. The phone then dials the number.

The following configuration parameters are applicable to this feature:

Parameter name	Default value	Description
ELD_SYSNUM	1	Specifies whether enhanced local dialing algorithm will be applied for system numbers. Value operation: • 0: Disable enhanced local dialing for system numbers. • 1: Enable enhanced local dialing for system numbers.
ENHDIALSTAT	1	Specifies if the algorithm defined by the parameter is used during certain dialing behaviors.

Table continues…

Parameter name	Default value	Description
		Value operation: • 0: Disables algorithm. • 1: Enables algorithm, but not for contacts. • 2: Enables algorithm, including contacts.
PHNCC	1	Specifies the international country code of the Communication Manager call server. For example, 1 for the United States, 44 for the United Kingdom, and so on. Valid values are from 1 to 999.
PHNDPLENGTH	5	Specifies the internal dial plan number length. For example, if the extension number is 12345, then the dial plan length is 5. This value must match the extension length set on your call server. Valid values are from 3 to 13.
PHNIC	011	Specifies the international access code. Valid values are from 0 to 4 characters such as numbers 0–9, and special symbols such as star key (*), and pound key (#).
PHNLD	1	Specifies long distance access code. Valid values are from 0 through 9 and empty string.
PHNLDLENGTH	10	Specifies the maximum length, in digits, of the national telephone number for the country in which the Communication Manager call server is located. For example, 800-555-1111 has a length of 10. Valid values are from 5 to 15.
PHNOL	9	Specifies the outside line access code. Valid values are from 0 to 2 characters such as numbers 0–9, and special symbols such as star key (*), and pound key (#).

Note:

- The parameter values must be relevant to the location of the Avaya Media Server where the IP phones are registered. For example, if a phone is in Japan and its media server is in the United States, set the PHNCC value to 1 for the United States.
- The digits the phones insert and dial are subject to standard Avaya Media Server features and administration. This includes Class of Service (COS), Class of Restriction (COR), Automatic Route Selection (ARS), and so on.

- Phones will not insert the expected digits when calling back from call history or contacts list if the configured SIP user extension is equal to or longer than the number stored in the call history.

Enhanced Local Dialing scenarios

The PHNOL parameter is applied without modification in the following scenario:

- ELD is applied to incoming history by setting the ENHDIALSTAT parameter to 1 or 2. A user calls a number from the incoming or missed call history. The number of digits in the number:
    1. Is greater than the national number length (PHNLDLENGTH).
    2. Is greater than the internal number length (PHNDPLENGTH) but lesser than the national number length (PHNLDLENGTH). (PHNDPLENGTH < length of the number < PHNLDLENGTH)

The PHNOL parameter is added to the called number in the following scenario:

- ELD is applied to Contacts by setting the ENHDIALSTAT parameter to 2. A user calls a number from Contacts. The number of digits in the number:
    1. Is greater than the national number length (PHNLDLENGTH), and PHNOL is not equal to the first digit of the number.
    2. Is greater than the internal number length (PHNDPLENGTH), and the length of this number is lesser than the national number length (PHNLDLENGTH). (PHNDPLENGTH < length of the number < PHNLDLENGTH)

PHNOL and PHNLD are applied to the number in the following scenario:

- A user calls a number from the incoming or missed call history (ENHDIALSTAT >= 1) or Contacts (ENHDIALSTAT = 2), and the length of this number is equal to the national number length (PHNLDLENGTH).

    Note:

    When the first digit of the called number matches PHNLD, only PHNOL is applied.

Restarting the phone

Procedure

1. Press **Main Menu** > **Admin**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **Restart phone**.
5. Press **Restart** when the phone prompts for confirmation.

    A restart does not affect user-specified data and settings, such as contact data or the phone login and password.

Configuring SIP settings

About this task

Use this procedure to set up SIP-related settings, such as identifying the SIP proxy server.

Note:

In IP Office the autogenerated `J100 settings.txt` includes the settings for the SIP servers and protocols. The settings are based on the SIP values set in the IP Office system configuration.

Procedure

1. Press **Main Menu** > **Admin**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **SIP**.
5. Choose one of the following:
    - **SIP global settings**
    - **SIP proxy server**
6. Press **Select** or **OK** to change any of the following SIP global settings:
    - **Domain**: Changes the domain parameter of SIP.
    - **Avaya Environment**: Specifies whether the available SIP Avaya environment is in effect.

      The two modes to detect the available environment are as follows:
        - **Auto**: Detects the Avaya environment automatically.
        - **No**: Does not detect the Avaya environment and switches to a non-AST mode.
    - **Reg. policy**: Specifies the registration policy for SIP.

      The two modes are as follows:
        - **Alternate**: Supports registration to one of the active controllers.
        - **Simultaneous**: Supports registration to both the active controllers.
    - **Failback policy**: Specifies the fall back policy.

      The two modes are as follows:
        - **Auto**: Active controller automatically recovers after failback.
        - **Admin**: Active controller uses failback policy defined by the administrator.
    - **Proxy policy**: Specifies whether the settings of SIP proxy servers are read-only or can be edited by the user.

The two modes are as follows:

- **Auto**: The user can only view the settings.
- **Manual**: The user can edit, delete, or create new server properties.

7. Select **SIP proxy server** to change SIP proxy server settings.

   ⚠ **Caution:**

   Do not configure proxy settings manually while a user is logged in to the phone.

   The phone displays the IP address of the server that you selected.
8. Press **Details** and use the **Up** and **Down Arrow** keys to view, add, or change the following settings:
   - **Proxy**: Specifies the IP address or DNS for Avaya Aura® Session Manager deployments. The corresponding parameter is SIP_CONTROLLER_LIST.
   - **Protocol**: Specifies the type of protocol. The options are TCP, UDP, or TLS. The corresponding parameter is SIPSIGNAL.
   - **SIP Port**: Specifies the SIP port. If no value is entered, SIP port uses 5060 as the default port for UDP/TCP or 5061 for TLS. If Transport Type is UDP/ TCP, the corresponding parameter is SIP_PORT_ SECURE.
9. Press **Save**.

Setting Site Specific Option Number (SSON)

About this task

The Site Specific Option Number (SSON) is used by the phones to request information from a DHCP server. This number must match a similar number option set on the DHCP server. The number option set on the DHCP server defines the various settings required by the phone.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the administation menu password.
3. Press **Enter**.
4. Select **SSON**.
5. In the **SSON** field, enter the new SSON.

   The number must be between 128 to 254.
6. Press **Save**.

⚠ **Caution:**

Do not perform this procedure if you are using static addressing. Perform this procedure if you are using DHCP addressing and the DHCP option number is changed from the default number.

Using the VIEW administrative option

About this task

Use this procedure to view the parameters associated with the admin procedures.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **View**.
5. Press **Back** to return to the main menu.

VIEW field description

Setting	Description	Associated Configuration Parameter
Model	The model of the phone that is set by factory procedures.	MODEL
Backup SW version	The version of the software backup.	
Gateway	The address of the gateway.	
Group	The group identifier to download during start-up a specific configuration set for a dedicated user group.	GROUP
MAC	The MAC address of the phone.	MACADDR
Serial number	The serial number of the phone.	
SIP Proxy Server	The SIP proxy server to which the phone registered successfully.	SIPPROXYSRVR_IN_ USE
Presence Server	The IP address of the presence server.	

Table continues…

Setting	Description	Associated Configuration Parameter
The setting is only available in an Avaya Aura® environment.		
HTTPS Server	The list of IP or DNS addresses of TLS servers for HTTPS file download, settings file or language files, during startup procedure.	TLSSRVR
HTTP Server	The IP address of the HTTP server that the phone accessed before successfully.	HTTPSRVR_IN_USE
DNS Server	The IP address of the DNS server that the phone accessed before successfully.	DNSSRVR_IN_USE
SW version	The version of the software.	
Protocol	Signaling protocol in effect, such as SIP.	

Setting the 802.1x operational mode

Before you begin

Set the following parameters:

- DOT1X: To support 802.1X Pass-thru operation, set the parameter to zero or one.
- DOT1XSTAT: To support supplicant operation, set the parameter to one or two.

Procedure

1. Press **Main Menu** > **Administration**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **802.1X**.

   The phone displays the following settings:

   - **Supplicant**
   - **Pass-thru mode**
5. Select the setting that you want to change.

6. Press the **Change** softkey or the **Left** and **Right Arrow** keys to cycle through the following settings:
    - For the Pass-thru mode:
        - **On**: If DOT1X = 0
        - **On & proxy logoff**: If DOT1X = 1
        - **Off**: If DOT1X = 2
    - For the Supplicant:
        - **Disabled**: If DOT1XSTAT = 0
        - **Unicast**: If DOT1XSTAT = 1
        - **Multicast**: If DOT1XSTAT = 2
7. Press **Save**.

    When you change the 802.1X data, the phone restarts after you exit the administration menu.

Chapter 9: Maintenance

Resetting system values

About this task

Use this 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.

Procedure

1. Press **Admin menu** > **Administration**.
2. In the **Access code** field, enter the admin menu password.
3. Press **Enter**.
4. Select **Reset to defaults**.
5. Press **Reset** when the phone prompts for confirmation.

   The phone resets from the beginning of registration, which might take a few minutes. The phone resets all settings to the defaults except user data stored remotely , for example: user data stored in PPM or on an external server specified by USER_STORE_URI parameter.

   After reset, the phone displays the Log In screen.

   Note:

   To reset the phone default value when both phone and web admin passwords are lost, press the key sequence of ‘Mute button’ ‘<phone mac address>’ ‘ #’. For MAC address, ‘2’ is mapped to a, b, c and ‘3’ is mapped to d, e, f.

   For example, if the phone mac address is A0:09:ED:05:80:51, key sequence would be ‘Mute 200933058051 #’.

   This is applicable to the phones in 3PCC environment only.

   Note:

   Avaya J100 Series IP Phones parameters stored for a particular user are not reflected in other phones, for example, 9600 Series IP Deskphones, even if the SIP user is the same.

Device upgrade process

1. During boot-up, the phone receives the file server address from DHCP, LLDP, or the device interface.
2. The phone contacts the provisioning server to download the firmware upgrade file, `J100Supgrade.txt`.
3. In `J100Supgrade.txt`, the APPNAME parameter contains the firmware version.
4. The phone compares the currently installed software version with the version specified in the APPNAME parameter.
5. If the firmware version specified in the APPNAME parameter differs from the currently running software version, the phone downloads the software files for upgrade.
6. The phone automatically restarts to apply the upgraded firmware.

   Tip:

   The upgrade events are logged under NOTICES level in the `Syslog` file.

User profile backup on Personal Profile Manager (PPM)

Phone supports data backup by saving all non-volatile user parameters on PPM . When the user logs in to any registered device, PPM restores all user data on the device.

Note:

PPM is only available in an Avaya Aura® environment.

User profile parameters for backup

The following table lists the parameters that are backed up on Personal Profile Manager (PPM).

Parameter	Default value	Description
CLICKS	1	Specifies if the phone button can generate click sounds.
OUTSIDE_CALL_RING_TYPE	1	Specifies the default outside call ring type.
CALL_PICKUP_INDICATION	3	Specifies the following call pickup indication types: • Audio • Visual • None

Table continues…

Parameter	Default value	Description
AMPLIFIED_HANDSET	0	Specifies whether the handset amplification is enabled.
AMPLIFIED_HANDSET_NOMINAL_LEVEL_CALL_END	0	Specifies whether to set the volume level in amplified mode to nominal when all calls end.
TIMEFORMAT	0	Specifies whether the time format is the am-pm format or the 24–hour format.
DATE_FORMAT_OPTIONS	1	Specifies the date display format.
CALL_LOG_ACTIVE	1	Specifies whether to activate call logging.
DEFAULT_CONTACTS_STORE	1	Specifies the account where all user contacts are added by default.
ENABLE_PHONE_LOCK	0	Specifies whether the **Lock** softkey and the Lock feature button are displayed on the phone.
SHOW_CALL_APPEARANCE_NUMBERS	0	Specifies whether for a user the device displays call appearance numbers in the call containers.

SLA Mon™ agent

SLA Mon™ technology is a patented Avaya technology embedded in Avaya products to facilitate advanced diagnostics. The phones support SLA Mon™ agent which works with Avaya Diagnostic Server (ADS). SLA Mon™ server controls the the SLA Mon™ agents to execute advanced diagnostic functions, such as:

- Endpoint Diagnostics
    - The ability to remotely control IP phones, to assist end users with IP Phone configuration and troubleshooting.
    - The ability to remotely generate single and bulk test calls between IP phones.
    - The ability to remotely execute limited packet captures on IP phones to troubleshoot and diagnose IP phone network traffic.
- Network Monitoring
    - The ability to monitor multiple network segments for performance in terms of packet loss, jitter, and delay.
    - The ability to monitor hop-by-hop QoS markings for voice and video traffic.

Note:

The root trusted certificate used for the SLA Mon™ server certificate must be added to the trusted certificate list administered using TRUSTCERTS.

For example: SET TRUSTCERTS *slamonRootCA.crt, rootCertRNAAD.cer*

Chapter 10: System failover

Supported SIP environments

Avaya J100 Series IP Phones work on the following environments:

- Avaya Aura® Session Manager with Avaya Aura® Communication Manager
- IP Office
- Failover and survivable interoperability with the following SIP gateways:
    - Session Manager for survivable remote gateway
    - Avaya Secure Router 2330 and 4134
    - Audiocodes MP-series analog and BRI gateways
    - Avaya Aura® Media Server 7.7.0.334.
    - IP Office

For information about configuring the phone features, see the following documents:

- *Avaya Aura® Communication Manager Feature Description and Implementation*
- *Administering Avaya Aura® Communication Manager*
- *Avaya IP Office™ Platform SIP Telephone Installation Notes*

Failover and survivability overview

The phone detects a network or server failure in approximately 90 seconds. After a failure is detected, the phone selects an active controller in approximately five seconds. During network or server failures, multiple controllers or servers are supported for the following operations:

- Making a call including emergency calls
- Receiving a call
- Call transfer
- Call forward
- Mid call features: Call hold and mute
- Audio Conference: Local three-way audio conference

Phone resiliency and transition states

The transition happens in the following order:

1. Limbo: Connection to the primary server is lost but the failover is not detected.
2. Moving Subscriptions Interval (MSI): Connection to the primary server is lost, and the phone is currently registered to the survivability server. Successful subscription to the survivability server is incomplete.
3. Acquiring services: Connection to the primary server is lost, and the phone displays the following message in the idle state `Acquiring Service`
4. Failover to the secondary/survivability server: Connection to the secondary/survivability server is active. All the supported features are also active. The phone performs the following intermediate steps:
    - Selection of active controller: The phone attempts to select the monitored active controller.
    - Successful subscription: Connection to the monitored controller is made with successful subscription.
    - Call/media preservation: During an active call, the phone detects that the connection is lost with the primary controller and the call/media is preserved. Media preservation is only available in an Avaya Aura® environment.
    - Advanced SIP Telephony (AST) feature determination: The phone verifies whether the controller supports the AST feature. AST feature is only available in an Avaya Aura® environment.
    - Personal Profile Manager (PPM) synchronization: When AST mode is determined and enabled, then the phone starts the PPM synchronization process. PPM is only available in an Avaya Aura® environment.
5. Failback to the primary server: Connection to the primary server is established when the phone detects that the primary server is functional again. The changes that were cached earlier are now synced with the PPM server. Failback does not happen during an active call.

Avaya J100 Series IP Phones survivability in the Avaya Aura® environment

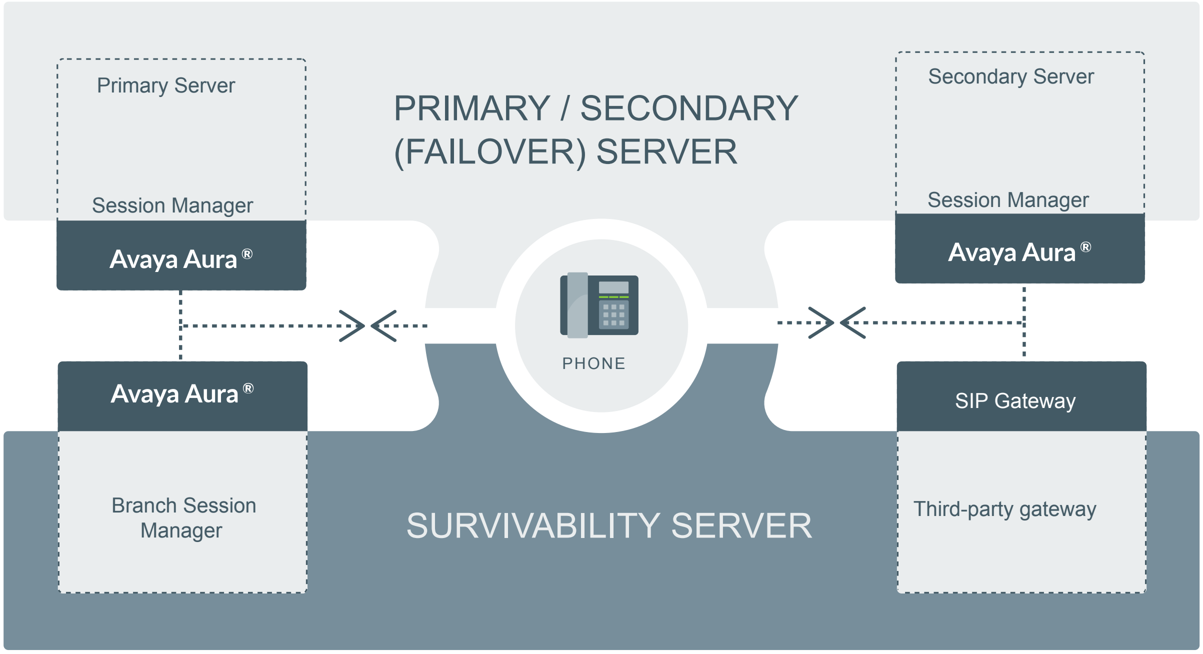


Figure 3: Survivability in Avaya Aura®

Note:

For more information on survivability about IP Office environment, see *Administering Avaya IP Office Platform with Web Manager*.

Survivability controller determination

The order of precedence in determining the active controller is:

1. Phone user interface
2. PPM server

3. Settings file
4. DHCP server (Option 242)

The phone performs the DNS queries to resolve hostnames and the signaling protocol. The order is set as TLS, TCP, and then UDP when there is no DNS NAPTR or SIP URI. The phone sends the SIP REGISTER request for each CONTROLLER_SEARCH_INTERVAL. The phone retries the monitoring attempt using the RECOVERYREGISTERWAIT parameter.

If the value of the SIPREGPROXYPOLICY parameter is alternate and a user is logged in, the phone maintains a single active SIP registration. SIP registration is done with the highest priority available controller. Also, the parameter FAILBACK_POLICY controls the SIP registration priority. If the FAILBACK_POLICY parameter is configured with automatic instead of admin, then the phone's active controller has the highest priority.

If the value of the SIPREGPROXYPOLICY parameter is simultaneous and a user is logged in, the phone maintains all active SIP registrations. The phone simultaneously registers using the value provided in the SIMULTANEOUS_REGISTRATIONS and SIPDOMAIN parameters.

The phone uses a SIP URI instead of SIPS URI unless SRTP is enabled. When registration is successful, the phone sets the SIPPROXYSRVR_IN_USE parameter to the IP address of this active controller.

The phone starts a search for a new active controller whenever it encounters one of the following triggers :

- Trigger 1: The TCP socket closes or TCP Keep-alive timeout occurs.
- Trigger 2: The phone receives an administrative failback trigger from a Configured Controller.
- Trigger 3: Fast Response Timer.
- Trigger 4: The phones receives n incoming INVITE from a non-active controller.
- Trigger 5: Re-registration with the active controller is timed out.

Advanced SIP Telephony feature determination

The parameter DISCOVER_AVAYA_ENVIRONMENT determines whether the selected controller supports the Advanced SIP Telephony (AST) feature. When the parameter value is set to 1, the phone sends a SUBSCRIBE request to the active controller for the Feature Status Event Package (avaya-cm-feature-status).

The phone determines the AST mode based on the response 202. Then it starts an internal timer of 16 seconds and waits to receive a NOTIFY message as `active`.

If the phone does not receive a NOTIFY message and receives a termination message instead, then the non-AST mode is enabled. Synchronization with the Personal Profile Manager (PPM) server starts when the AST mode is enabled.

Synchronization with the Personal Profile Manager server

The phone performs the synchronization with the Personal Profile Manager (PPM) server only when the `getAllEndpointConfiguration` request is successful. If the `getAllEndpointConfiguration` request is unsuccessful, the `getContactList` request is also ignored. This request contains the following fields:

- VolumeSettings
- LinePreferenceInfo
- ListOfOneTouchDialData
- ListOfButtonAssignments
- SoftMenuKeyList
- DialPlanData
- ListOfSpeedDialData
- ListOfMaintenanceData
- ListOfTimers
- VMONInfo
- ListOfRingerOnOffData
- ListOfNumberFormatRules: Applicable only when registered to Avaya Aura® Session Manager.
- ListOfIdentities: Applicable only when registered to Avaya Aura® Session Manager.

  MWExt: Applicable only when registered to Avaya Aura® Session Manager.
- VMNumber: Applicable only when registered to Avaya Aura® Session Manager.

Provisioning survivability for SIP phones

About this task

Use this procedure to provision survivability.

In IP Office environment, survivability is provisioned in the autogenerated Settings file.

Procedure

1. Set the applicable failover configuration parameters in the `46xxsettings.txt` file.
2. Provision the gateway per the Application Notes, available on the https://support.avaya.com/ website.
3. Load the latest SIP Release software and associated files on the file server.
4. Reboot all registered phones from SIP Enablement Services or Avaya Aura® Session Manager.
5. Power up other phones.

Configuring survivability

Use the `46xxsettings.txt` file to set survivability configuration parameters.

In IP Office, the autogenerated `J100 settings.txt` file contains the survivability configuration parameter details.

By administering survivability configuration parameters using the `46xxsettings.txt` file (or using the default values if applicable), the SIP phones can quickly switch to an active controlling server and experience minimal disruption.

Failover or failback parameters

Parameter name	Default value	Description
CONTROLLER_SEARCH_INTERVAL	16	Specifies the time the phone waits to complete the maintenance check for Monitored Controllers. Valid values are from 4 to 3600.
DISCOVER_AVAYA_ENVIRONMENT		Specifies dynamic feature set discovery. Value operation: 1: The phone discovers and verifies if the controller supports the AST feature set or not. The phone sends a SUBSCRIBE request to the active controller for the Feature Status Event Package (avaya-cm-feature-status). If the request succeeds, the phone proceeds with PPM synchronization. If the request is rejected, or is proxied back to the phone, or does not receive a response, the phone assumes that AST features are not available. 0: The phone operates in a mode where AST features are not available.
ENABLE_REMOVE_PSTN_ACCESS_PREFIX		Allows phone to perform digit manipulation during failure scenarios. This parameter allows removal of PSTN access prefix from the outgoing number. The parameter is not supported in Value operation: 0: PSTN access prefix is retained in the outgoing number 1: PSTN access prefix is stripped from the outgoing number.

Table continues…

Parameter name	Default value	Description
PSTN_VM_NUM		Specifies the phone number to be dialed when the phone is in failover and the Message button is pressed. **Note:** This parameter is applicable in IP Office, 3PCC environment or incase of Avaya Aura environment failover.
REGISTERWAIT		Specifies the number of seconds between re-registrations with the current server.
SIP_CONTROLLER_LIST	Null	Specifies a list of SIP controller designators, separated by commas without any spaces. Controller designator has the following format: host[:port][;transport=xxx], where, host is an proxy address in dotted-decimal or DNS name format. In third-party call control setup, only DNS format is supported. [:port] is an optional port number. [;transport=xxx] is an optional transport type where xxx can be TLS, TCP, or UDP.
SIMULTANEOUS_REGISTRATIONS	3	The number of Session Managers with which the phone will simultaneously register. Valid values are 1, 2 or 3.
SIPREGPROXYPOLICY	Simultaneous	Specifies whether the phone will attempt to maintain one or multiple simultaneous registrations. Value operation: • Alternate: The phone attempts and maintains only a single registration. • Simultaneous: The phone attempts and maintains simultaneous registrations will be attempted and maintained with all available controllers. In IP Office environment and third-party call control setup, set the parameter to Alternate.

Configuring AudioCodes server for survivability

If you set AudioCodes server in the Avaya environment for survivability, you must configure the following options:

- Connection reuse

- Connection reuse in survivability mode
- Record-Route

Enabling connection reuse

Procedure

1. Go to the audio codes URL and click **Configuration** > **VoIP** > **SIP Definitions** > **General Parameters**.
2. Set **Enable TCP Connection Reuse** to **Enable**.
3. Click **Submit**.

Enabling connection reuse in a failover environment

Procedure

1. Go to the audio codes URL and click **Configuration** > **VoIP** > **SAS** > **Stand Alone Survivability**.
2. Set **SAS Connection Reuse** to **Enable**.
3. Click **Submit**.

Enabling Record Route in invite messages

Procedure

1. Go to the audio codes URL and click **Configuration** > **VoIP** > **SAS** > **Stand Alone Survivability**.
2. Set **Enable Record-Route** to **Enable**.
3. Click **Submit**.

User experience during failover

Feature	Normal Operation with Communication Manager	Failover Operation with a Generic SIP Gateway	IP Office branch mode
Modifying contacts	Yes	Yes	No

Table continues…

Feature	Normal Operation with Communication Manager	Failover Operation with a Generic SIP Gateway	IP Office branch mode
Make call	Yes	Yes	Yes
Receive call	Yes	Yes	Yes
Call Hold	Yes	Yes	Yes
Consultative Hold	Yes	Yes	Yes
Ad hoc conferencing	Yes, up to 6 parties	Yes, up to 3 parties	Yes, up to 3 parties
Forward all my calls/SAC	Yes	Yes	Yes In IP Office the feature is handled using shortcodes.
Forward my calls when busy/no answer	Yes	Yes	Yes In IP Office the feature is handled using shortcodes.
Attended call transfer	Yes	Yes	Yes
Inbound call management	Yes (Communication Manager COR)	Yes (depends on local proxy capabilities and provisioning)	Yes (depends on local proxy capabilities and provisioning)
Outbound call management	Yes (Communication Manager COR)	Yes (proxy)	Yes (proxy)
Calling party block	Yes	No	No
Call park	Yes	No	Yes In IP Office the feature is handled using shortcodes.
Call unpark	Yes	No	Yes In IP Office the feature is handled using shortcodes.
Auto callback	Yes	No	No
Malicious call trace	Yes	No	No
EC500 on/off	Yes	No	No
Transfer to voice mail	Yes	No	No
Extend-call	Yes	No	No
Hold recall	Yes	No	No
Transfer recall	Yes	No	No
Message waiting indicator	Yes	No	No

Note:

If the phone displays the message `Limited phone service`, press **OK** to acknowledge and clear the message.

Chapter 11: Troubleshooting

SLA Mon™ agent

SLA Mon™ technology is a patented Avaya technology embedded in Avaya products to facilitate advanced diagnostics. The phones support SLA Mon™ agent which works with Avaya Diagnostic Server (ADS). SLA Mon™ server controls the the SLA Mon™ agents to execute advanced diagnostic functions, such as:

- Endpoint Diagnostics
  - The ability to remotely control IP phones, to assist end users with IP Phone configuration and troubleshooting.
  - The ability to remotely generate single and bulk test calls between IP phones.
  - The ability to remotely execute limited packet captures on IP phones to troubleshoot and diagnose IP phone network traffic.
- Network Monitoring
  - The ability to monitor multiple network segments for performance in terms of packet loss, jitter, and delay.
  - The ability to monitor hop-by-hop QoS markings for voice and video traffic.

Note:

The root trusted certificate used for the SLA Mon™ server certificate must be added to the trusted certificate list administered using TRUSTCERTS.

For example: SET TRUSTCERTS *slamonRootCA.crt, rootCertRNAAD.cer*

Phone displays Acquiring Service screen

Cause

The configured SIP proxy servers are not accessible from the phone.

Solution

1. On the Acquiring Service screen, press **Cancel** to logout from the phone and go to the **Admin** menu.
2. Press **SIP** > **SIP proxy server**.

3. Check the number of SIP proxy servers that are configured. If the connections are properly configured, then ensure the following:
    - SIP proxy servers are specified by IP address and not by FQDN.
    - There are only two proxy servers configured.

    A filled in circle implies a successful configuration. A circle with a line through it implies a failed connection.

Cause

The configured SIP proxy servers are accessible. However, TLS is being used and there is an issue with the certificate configuration.

Solution

1. On the Acquiring Service screen, press **Cancel** to logout from the phone and go to the **Admin** menu.
2. Press **SIP** > **SIP global settings**.
3. Use the **Up** and **Down** arrow keys to go to the Reg. policy screen.
4. Use the **Left** arrow key to configure the Reg. policy as **Alternate** and press **Save**.
5. Use the **Up** and **Down** arrow keys to go to the Avaya Environ screen.
6. Use the **Left** arrow key to configure the Avaya Environ as **No** and press **Save**.

Cause

There is a problem with the SIP proxy configuration.

Solution

1. On the Acquiring Service screen, press **Cancel** to logout from the phone and go to the **Admin** menu.
2. Press **SIP** > **SIP proxy server**.
3. If one or more configured SIP proxy server connections shows as failed, press **Ping**.

    The circle is filled in if the connection is properly configured. Circle with a line through it is a failed connection.
4. Ping each SIP proxy server.

Chapter 12: Resources

Documentation

See the following related documents at http://support.avaya.com.

Title	Use this document to:	Audience
Overview		
Avaya Aura® Session Manager Overview and Specification	See characteristics and capabilities, including feature descriptions, interoperability, performance specifications, security and licensing requirements of the Avaya Aura® Session Manager.	For people who want to gain a high-level understanding of the Avaya Aura® Session Manager features, functions, capacities, and limitations.
Avaya IP Office™ Platform Feature Description	See information about the feature descriptions.	For people who perform system administration tasks.
Avaya IP Office™ Platform Solution Description	See information about how the products and services that interoperate with this solution.	For people who want to gain a high-level understanding of the IP Office features, functions, capacities, and limitations.
Implementing		
Deploying Avaya Aura® Session Manager	See the installation procedures and initial administration information for Avaya Aura® Session Manager.	For people who install, configure, and verify Avaya Aura® Session Manager on Avaya Aura® System Platform.
Upgrading Avaya Aura® Session Manager	See upgrading checklists and procedures.	For people who perform upgrades of Avaya Aura® Session Manager.
Deploying Avaya Aura® System Manager on System Platform	See the installation procedures and initial administration information for Avaya Aura® System Manager.	For people who install, configure, and verify Avaya Aura®

Table continues…

Title	Use this document to:	Audience
		System Manager on Avaya Aura® System Platform at a customer site.
Avaya IP Office™ Platform SIP Telephone Installation Notes	See the installation procedures and initial administration information for IP Office SIP telephone devices.	For people who install, configure and verify SIP telephone devices on IP Office.
Administering		
Administering Avaya Aura® Session Manager	See information about how to perform Avaya Aura® Session Manager administration tasks including how to use management tools, how to manage data and security, an how to perform periodic maintenance tasks.	For people who perform Avaya Aura® Session Manager system administration tasks.
Administering Avaya Aura® System Manager	See information about how to perform Avaya Aura® System Manager administration tasks including how to use management tools, how to manage data and security, an how to perform periodic maintenance tasks.	For people who perform Avaya Aura® System Manager administration tasks.
Administering Avaya IP Office™ Platform with Manager	See information about short code configurations for the feature list	For people who need to access IP Office features using short codes.
Administering Avaya IP Office™ Platform with Web Manager	See information about IP Office Web Manager administration tasks including how to use the management tool, how to manage data and security, and how to perform maintenance tasks.	For people who perfrom IP Office Web Manager administration tasks.
Maintaining		
Maintaining Avaya Aura® Session Manager	See information about the maintenance tasks for Avaya Aura® Session Manager.	For people who maintain Avaya Aura® Session Manager.
Troubleshooting Avaya Aura® Session Manager	See information for troubleshooting Avaya Aura® Session Manager, resolving alarms, replacing hardware, and alarm codes and event ID descriptions.	For people who troubleshoot Avaya Aura® Session Manager.
Using Avaya IP Office™ Platform System Status Application	See information about the maintenance tasks for System Status Application.	For people who maintain System Status Application.
Using Avaya IP Office™ Platform System Monitor	See information about the maintenance tasks for SysMonitor.	For people who maintain SysMonitor.

Finding documents on the Avaya Support website

Procedure

1. Navigate to http://support.avaya.com/.
2. At the top of the screen, type your username and password and click **Login**.
3. Click **Support by Product** > **Documents**.
4. In **Enter your Product Here**, type the product name and then select the product from the list.
5. In **Choose Release**, select an appropriate release number.
6. In the **Content Type** filter, click a document type, or click **Select All** to see a list of all available documents.

   For example, for user guides, click **User Guides** in the **Content Type** filter. The list displays the documents only from the selected category.
7. Click **Enter**.

Viewing Avaya Mentor videos

Avaya Mentor videos provide technical content on how to install, configure, and troubleshoot Avaya products.

About this task

Videos are available on the Avaya Support website, listed under the video document type, and on the Avaya-run channel on YouTube.

Procedure

- To find videos on the Avaya Support website, go to http://support.avaya.com and perform one of the following actions:
  - In **Search**, type `Avaya Mentor Videos` to see a list of the available videos.
  - In **Search**, type the product name. On the Search Results page, select **Video** in the **Content Type** column on the left.
- To find the Avaya Mentor videos on YouTube, go to www.youtube.com/AvayaMentor and perform one of the following actions:
  - Enter a key word or key words in the **Search Channel** to search for a specific product or topic.
  - Scroll down Playlists, and click the name of a topic to see the available list of videos posted on the website.

Note:

Videos are not available for all products.

Support

Go to the Avaya Support website at http://support.avaya.com for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

Appendix A: List of configuration parameters

Parameter name	Default value	Description
A		
100REL_SUPPORT	1	Specifies whether the 100rel option tag is included in the SIP INVITE header field. Value Operation: • 0: The tag is not included. • 1: The tag is included.
ADMIN_HSEQUAL	1	Specifies handset audio equalization standards compliance. This parameter impacts the phone only if the handset equalization is not set by the user or by the HSEQUAL local procedure for that phone. Value Operation: • 1: Use handset equalization that is compliant with TIA 810/920. • 2: Use handset equalization that is compliant with FCC Part 68 HAC requirements.
ADMIN_LOGIN_ATTEMPT_ALLOWED	10	Specifies the allowed number of failed attempts to enter the access code before the local or craft procedures gets locked. Valid values are from 1 to 20.
ADMIN_LOGIN_LOCKED_TIME	10	Specifies the duration for lockout when a user reaches the maximum attempts limit for accessing the Administration menu. Valid values are from 5 min. to 1440 min.
ADMIN_PASSWORD	27238	Specifies an access code for accessing the Admin menu. Valid values are from 6 to 31 alphanumeric characters including upper case, lower case characters and special characters. However,

Table continues…

Parameter name	Default value	Description
		double quote character (“) cannot be used for a value of this parameter. ✱ **Note:** - If this parameter length is set below 6 or above 31 alphanumeric characters, then the parameter is treated as not defined. - If this parameter is set in the `46xxsettings.txt` file, then it replaces PROCPSWD parameter. - If you set ADMIN_PASSWORD in the Avaya Aura® System Manager you require at least Avaya Aura® System Manager 7.1.0. - Setting this parameter through PPM is more secure because this file can usually be accessed and read by anyone on the network. Setting the value in this file is intended primarily for configurations with versions of phone or if server software that do not support setting this value from the server.
AGCHAND	1	Specifies the status of Automatic Gain Control (AGC) for the handset. Value Operation: • 0: Disables AGC for the handset. • 1: Enables AGC for the handset.
AGCSPKR	1	Specifies the status of Automatic Gain Control (AGC) for the speaker. Value Operation: • 0: Disables AGC for the speaker. • 1: Enables AGC for the speaker.
AMADMIN		Specifies the URI used for WML-applications under (AVAYA) Menu. You must specify HTTP server and directory path to administration file (`AvayaMenuAdmin.txt`). Do not specify the administration file name.
ASTCONFIRMATION	60	Specifies the number of seconds that the phone waits to validate an active subscription when it subscribes to the avaya-cm-feature-status package. Valid values are 16 through 3600.

Table continues…

Parameter name	Default value	Description
		This parameter is not supported in IP Office environment as there is no subscription to Avaya-cm-feature-status.
AUDIOSTHS	0	Specifies the level of sidetone in the handset. Value Operation: • 0: Normal level for most users • 1: Three levels softer than normal • 2: Inaudible • 3: One level softer than normal • 4: Two levels softer than normal • 5: Four levels softer than normal • 6: Five levels softer than normal • 7: Six levels softer than normal • 8: One level louder than normal • 9: Two levels louder than normal
AUTH		Specifies whether the script files are downloaded from an authenticated server over an HTTPS link. Value Operation: • 0: Optional • 1: Mandatory
AUTHCTRLSTAT	0	Specifies if the enhanced debugging capabilities can be activated from the SSH server by the Avaya technicians only. Value Operation: • 0: Enhanced debugging capabilities are disabled. • 1: Enhanced debugging capabilities are enabled. The parameter must be set to 1 only for the debugging period by Avaya technicians. Set the parameter back to 0 when the debugging period completes.
BACKGROUND_IMAGE		Specifies custom background images that can be loaded from the provisioning server. Phone supports up to 5 background images with the following limitation: • Only jpeg format files are supported. • The maximum file size is 256 KB.

Table continues…

Parameter name	Default value	Description
		• The file names are case sensitive. Example: SET BACKGROUND_IMAGE [xxx.jpg]
BACKGROUND_IMAGE_DISPLAY		Specifies the background image to be displayed. Note that, If BACKGROUND_IMAGE_SELECTABLE is set to 1 then the end user may override this setting.
BACKGROUND_IMAGE_SELECTABLE	1	Allows the end user to select background images. Value operations: • 0: The user can not use a background images from the phone UI. • 1: The user can select a background images from the phone UI.
BAKLIGHTOFF	120	Specifies the number of minutes of idle time after which the display backlight will be turned off. Phones with gray-scale displays do not completely turn backlight off, they set it to the lowest non-off level. Valid values are 0 through 999. A value of 0 means that the display backlight will not be turned off automatically when the phone is idle.
BRANDING_VOLUME	5	Specifies the volume level at which the Avaya audio brand is played. Value Operation • 8: 9db above nominal • 7: 6db above nominal • 6: 3db above nominal • 5: nominal • 4: 3db below nominal • 3: 6db below nominal • 2: 9db below nominal • 1:12db below nominal
BRURI	Null	Provides the capability to send a phone report to a server with the URI of the server defined by this parameter. To send the report, go to **Main Menu** > **Admin** > **Debug** > **Phone report**.
C		

Table continues…

Parameter name	Default value	Description
CALL_TRANSFER_MODE	0	Determines the call transfer mode in 3rd party environments. Valid value is 0 or 1.
CALLFWDADDR The parameter is only available in an Avaya Aura® environment.	Null	Sets the address to which calls are forwarded for the call forwarding feature. Users can change or replace this administered value if CALLFWDSTAT is not 0.
CALLFWDDELAY The parameter is only available in an Avaya Aura® environment.		Sets the number of ring cycles before the call is forwarded to the forward or coverage address. The default delay is one ring cycle.
CALLFWDSTAT The parameter is only available in an Avaya Aura® environment.	0	Sets the call forwarding mode of the phone by summing the following values: • 1: Permits unconditional call forwarding. • 2: Permits call forward on busy. • 4: Permits call forward/no answer. • 0: Disables call forwarding. Example: a value of 6 allows call forwarding on busy and on no answer.
CERT_WARNING_DAYS	60	Specifies the number of days before the expiration of a certificate that a warning will first appear on the phone screen. Certificates include trusted certificates, OCSP certificates and identity certificate. Log and syslog message will also be generated. The warning will reappear every seven days.Valid values are from 0 to 99. Value operation: • 0: No certificate expiration warning will be generated.
CERT_WARNING_DAYS_EASG	365	Specifies how many days before the expiration of EASG product certificate that a warning should first appear on the phone screen. Syslog message will be also generated. Valid values are from 90 to 730.
CNGLABEL	1	Determines if personalize button labels can be displayed to the user. Value Operation: • 0: Capability not displayed to the user. • 1: Capability displayed to the user.
CONFERENCE_FACTORY_URI	Null	Specifies the URI for Avaya Aura Conferencing. Valid values contain zero or one URI, where a URI consists of a dial string followed by @, and then

Table continues…

Parameter name	Default value	Description
		the domain name, which must match the routing pattern configured in System Manager for Adhoc Conferencing. Depending on the dial plan, the dial string can need a prefix code, such as a 9 to get an outside line. The domain portion of the URI can be in the form of an IP address or an FQDN. The value can contain 0 to 255 characters. The default value is null.
CONFERENCE_TYPE	1	Determines the selection of the Conference Method. Value Operation: • 0: Local conferencing is supported based on sipping services. • 1: Server based conferencing is supported. • 2: Click-to conference server based conferencing is supported. If the parameter is set to a value that is outside the range then default value is selected. **Note:** The parameter is set to 0 in IP Office environment.
CONFIG_SERVER	Null	Specifies the address of the Avaya configuration server. Valid values contain zero or one IP address in dotted decimal or DNS name format, optionally followed by a colon and a TCP port number. The value can contain 0 to 255 characters. The default value is null. This parameter is not supported in IP Office environment as PPM is not supported.
CONFIG_SERVER_SECURE_MODE	1	Specifies whether HTTP or HTTPS is used to access the configuration server. Value Operation: • 0: HTTP • 1: HTTPS • 2: Use HTTPS if SIP transport mode is TLS, otherwise use HTTP.

Table continues…

Parameter name	Default value	Description
		This parameter is not supported in IP Office environment as PPM is not supported.
CONNECTION_REUSE	1	Specifies whether the phone will use two UDP, TCP, or TLS connection (for both outbound and inbound) or one UDP, TCP, or TLS connection. Value operation: • 0: Disabled. The phone opens outbound connection to the SIP Proxy and listening socket for inbound connection from SIP proxy in parallel. • 1: Enabled. The phone does not open a listening socket and will maintain and re-use the sockets it creates with the outbound proxies. ✱ **Note:** On Avaya J129 IP Phone, only 1 is supported.
CONTACT_NAME_FORMAT	0	Specifies how contact names are displayed. Value operation • 0: The name format is Last name, First name. • 1: The name format is First name, Last name.
CONTROLLER_SEARCH_INTERVAL	16	Specifies the number of seconds the phone will wait to complete the maintenance check for monitored controllers. Valid values are 4 through 3600.
COUNTRY		Used for network call progress tones. • For Argentina use keyword Argentina. • For Australia use keyword Australia. • For Brazil use keyword Brazil. • For Canada use keyword USA. • For France use keyword France. • For Germany use keyword Germany. • For Italy use keyword Italy. • For Ireland use keyword Ireland. • For Mexico use keyword Mexico. • For Spain use keyword Spain. • For United Kingdom use keyword UK. • For United States use keyword USA.

Table continues…

Parameter name	Default value	Description
		Country names with spaces must be enclosed in double quotes.
COVERAGEADDR	Null	Sets the address to which calls will be forwarded for the call coverage feature. Users can change or replace this administered value if CALLFWDSTAT is not 0.
CURRENT_LOGO	None	Specifies if custom logo or wallpaper is selected for display. The CURRENT_LOGO is used in the following cases: • The phone is not registered to Avaya Aura® Session Manager. • The phone is registered to Avaya Aura® Session Manager and - there is no information stored for the current logo file for this specific user, and - there is no support of Profile Settings in the Endpoint Template. This is supported by Avaya Aura® System Manager 6.3.8 and later. If none is used for logo or wallpaper display, then the phone only displays time or date.
D		
DATEFORMAT		Specifies the format for dates displayed in the phone. • Use %d for day of month • Use %m for month in decimal format. • Use %y for year without century (For example, 07). • Use %Y for year with century (For example, 2007). Any character not preceded by % is reproduced exactly.
DAYLIGHT_SAVING_SETTING_MODE		Specifies daylight savings time setting for phone. Value Operation: • 0: Daylight saving time not activated • 1: Daylight saving time is activated. Time set to DSTOFFSET.

Table continues…

Parameter name	Default value	Description
		- 2: Activates automatic daylight savings adjustment as specified by DSTSTART and DSTSTOP.
DES_ACTIVE	0	Specifies if the DES discovery process is to be activated or deactivated after the DES is enabled. Value operation: - 0: DES discovery is disabled. - 1: DES discovery to be initiated if there is no configuration server defined. - 2: DES discovery must be initaiated.
DES_BLOCK	0	Specifies if DES discovery is allowed to be attempted during the boot process. Specifies if DES discovery is to be attempted during the boot process if there is no configuration file server provisioned on the phone. Value operation: - 0: DES discovery is allowed. - 1: DES discovery is not attempted. DES can still be initiated from the Administration menu. - 2: DES discovery is prohibited .
DES_STAT	2	Specifies if DES discovery is to be attempted during the boot process if there is no configuration file server provisioned on the phone. Value operation: - 0: DES discovery is disabled and can only be restored with Reset to Defaults - 1: DES discovery is disabled - 2: DES discovery is enabled
DELETE_MY_CERT	0	Specifies whether the installed identity certificate, using SCEP or PKCS12 file download, will be deleted. - 0: Installed identity certificate remains valid. - 1: Installed identity certificate is removed.
DHCPSTAT	1	Specifies whether DHCPv4, DHCPv6, or both will be used in case IPv6 support is enable by using IPV6STAT.

Table continues…

Parameter name	Default value	Description
		Value operation: • 1: Run DHCPv4 only. IPv4only-mode, if no own IPv6 address is programmed statically • 2: Run DHCPv6 only. Pv6only-mode, if no own IPv4 address is programmed statically • 3: Run both DHCPv4 & DHCPv6. Dual-stack mode Value 2 or 3 run both DHCPv4 and DHCPv6.
DHCPSTD	0	Specifies whether DHCP complies with the IETF RFC 2131 standard. Value Operation: • 0: Continue using the address in an extended rebinding state. • 1: Immediately stop using the address.
DIALPLAN	Null	Specifies the dial plan used in the phone. Dialplan accelerates dialing by eliminating the need to wait for the INTER_DIGIT_TIMEOUT timer to expire. The value can contain 0 to 1023 characters. The default value is null.
DISCOVER_AVAYA_ENVIRONMENT		Specifies dynamic feature set discovery Value Operation: • 1: The phone discovers and verifies if the controller supports the AST feature set or not. The phone sends a SUBSCRIBE request to the active controller for the Feature Status Event Package (avaya-cm-feature-status). If the request succeeds, the phone proceeds with PPM Synchronization. If the request is rejected, or is proxied back to the phone, or does not receive a response, the phone assumes that AST features are not available. • 0: The phone operates in a mode where AST features are not available. **Note:** Set the parameter to 0 for IP Office environment.

Table continues…

Parameter name	Default value	Description
DISPLAY_NAME_NUMBER	0	Specifies whether the name or number will be displayed for incoming calls, and if both are displayed, the order in which they are displayed. Value Operation: • 0: Display calling party name only. • 1: Display calling party name followed by calling party number. • 2: Display calling party number only. • 3: Display calling party number followed by calling party name.
DISPLAY_SSL_VERSION	0	Specifies whether OpenSSL and OpenSSH versions are displayed in the **Administration** menu. Value Operation: • 0: OpenSSL and OpenSSH versions are not displayed. • 1: OpenSSL and OpenSSH versions are displayed.
DNSSRVR		Domain Name Server for Access Profile 2
DOMAIN	Null	Specifies a character string that will be appended to parameter values that are specified as DNS names, before the name is resolved. The value can contain 0 to 255 characters. The default value is null.
DOT1X		Specifies the 802.1X pass-through operating mode. Pass-through is the forwarding of EAPOL frames between the phone's ethernet line interface and its secondary (PC) ethernet interface Value Operation: • 0: EAPOL multicast pass-through enabled without proxy logoff. • 1: EAPOL multicast pass-through enabled with proxy logoff. • 2: EAPOL multicast pass-through disabled.
DOT1XEAPS	MD5	Specifies the authentication method to be used by 802.1X. Valid values are MD5, and TLS.

Table continues…

Parameter name	Default value	Description
DOT1XSTAT	0	Specifies the 802.1X supplicant operating mode. Value Operation: • 0: Supplicant disabled. • 1: Supplicant enabled, but responds only to received unicast EAPOL messages. • 2: Supplicant enabled; responds to received unicast and multicast EAPOL messages.
DSCPAUD	46	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for audio frames generated by the phone. Valid values are from 0 to 63. This parameter can also be set through the LLDP, which overwrites any value set in this file.
DSCPAUD_FO	41	Specifies the DSCP value for flash Override precedence or priority level voice call. Valid values are from 0 to 63.
DSCPAUD_FL	43	Specifies the DSCP value for flash precedence or priority level voice call. Valid values are from 0 to 63.
DSCPAUD_IM	45	Specifies the DSCP value for immediate precedence or priority level voice call. Valid values are from 0 to 63.
DSCPAUD_PR	47	Specifies the DSCP value for priority precedence or priority level voice call. Valid values are from 0 to 63.
DSCPMGMT	16	Specifies the DSCP value for OA&M management packet. Valid values are from 0 to 63.
DSCPSIG	34	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for signaling frames generated by the phone. Valid values are 0 through 63. This parameter can also be set through LLDP, which overwrites any value set in this file.
DSCPVID	34	Specifies the layer 3 Differentiated Services (DiffServ) Code Point for video frames generated by the phone.

Table continues…

Parameter name	Default value	Description
		Valid values are 0 through 63. The default value is 34.
DSTOFFSET	1	Specifies the time offset in hours of daylight savings time from local standard time. Valid values are 0, 1, or 2. The default value is 1.
DSTSTART	2SunMar2L	Specifies when to apply the offset for daylight savings time. The default value is 2SunMar2L (the second Sunday in March at 2AM local time).
DSTSTOP	1SunNov2L	Specifies when to stop applying the offset for daylight savings time. The default value is 1SunNov2L (the first Sunday in November at 2AM local time).
DTMF_PAYLOAD_TYPE	120	Specifies the RTP payload type to be used for RFC 2833 signaling. Valid values are 96 through 127.
E		
EASG_SITE_AUTH_FACTOR	Null	Specifies Site Authentication Factor code associated with the EASG site certificate being installed. Valid values are 10 to 20 character alphanumeric string.
EASG_SITE_CERTS	Null	Specifies list of EASG site certificates which are used by technicians when they don't have access to the Avaya network to generate EASG responses for SSH login. The URLs must be separated by commas without any intervening spaces. Valid values are 0 to 255 ASCII characters.
ELD_SYSNUM	1	Controls whether Enhanced Local Dialing algorithm will be applied for System Numbers-Busy Indicators and Auto Dials. Value operation: • 0: Disable ELD for System Numbers • 1: Enable ELD for System Numbers
EEESTAT	1	Specifies Energy-Efficient Ethernet (802.3az) is enabled on PHY1 and PHY2. Value operation: • 0: EEE is disabled on both PHY1 and PHY2. • 1; EEE is enabled on both PHY1 and PHY2.

Table continues…

Parameter name	Default value	Description
ENABLE_AVAYA_ENVIRONMENT	1	Specifies whether the phone is configured to be used in an Avaya (SES) or a third-party proxy environment. Value Operation: • 0: Configured for 3rd party proxy with SIPPING 19 features. • 1: Configured for Avaya SES with AST features and PPM. **Note:** Set the parameter to 0 for IP Office environment.
ENABLE_BLIND_TRANSFER	1	Specifies that whether the blind transfer is enabled or not. Value Operation: • 0: Disabled. • 1: Enabled.
ENABLE_CALL_LOG		Species if call logging and associated menus are available on the phone. Value Operation: • 0: No • 1: Yes
ENABLE_CONTACTS	1	Specifies if the contacts application and associated menus are available on the phone. Value Operation: • 0: No. The phone disables the **Contacts** option on the interface. • 1: Yes **Note:** The parameter is set to 1 in IP Office 10.1 or later. In previous releases it is set to 0.
ENABLE_EARLY_MEDIA		Specifies if the phone sets up a voice channel to the called party before the call is answered. Value Operation: • 0: No • 1: Yes

Table continues…

Parameter name	Default value	Description
		Setting this parameter to 1 can speed up call setup.
ENABLE_EXCHANGE_REMINDER	0	Specifies whether or not exchange reminders will be displayed. Value Operation: • 0: Not displayed • 1: Displayed
ENABLE_G711A	1	Specifies if the G.711 a-law codec is enabled. Value Operation: • 0: Disabled • 1: Enabled
ENABLE_G711U	1	Specifies if the G.711 mu-law codec is enabled. Value Operation: • 0: Disabled • 1: Enabled
ENABLE_G722	1	Specifies if the G.722 codec is enabled. Value Operation: • 0: Disabled • 1: Enabled
ENABLE_G726	1	Specifies if the G.726 codec is enabled. Value Operation: • 0: Disabled • 1: Enabled
ENABLE_G729	1	Specifies if the G.729A codec is enabled. Value Operation: • 0: Disabled • 1: Enabled without Annex B support (default). • 2: Enabled with Annex B support.
ENABLE_IPOFFICE	0	Specifies whether the J100 phone can operate in 2 different modes with IP Office. The first mode allows native support of the J100 phone with IP Office with a limited feature set. The second mode allows support of the J100 phone with additional feature support driven by the IP Office proxy.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: The phone does not support IP Office (except in Avaya Aura failover mode). • 1: The phone supports IP Office in a native environment. • 2: The phone supports IP Office with additional features driven by the IP Office proxy
ENABLE_MLPP	0	Specifies that whether the Multiple Level Precedence and Preemption (MLPP) is enabled or not. Value Operation: • 0: Disabled. • 1: Enabled.
ENABLE_MODIFY_CONTACTS		Specifies if the list of contacts and the function of the contacts application can be modified on the phone. Value Operation: • 0: No • 1: Yes
ENABLE_MULTIPLE_CONTACT_WARNING		Specifies if a warning message must be displayed if there are multiple phones registered on a user's behalf. Value Operation: • 0: No • 1: Yes **Note:** Multiple registered phones can lead to service disruption.
ENABLE_OOD_MSG_TLS_ONLY	1	Specifies if an Out-Of-Dialog (OOD) REFER must be received over TLS transport to be accepted. Value Operation: • 0: No, TLS is not required. • 1: Yes, TLS is required. **Note:** A value of 0 is only intended for testing purposes.

Table continues…

Parameter name	Default value	Description
ENABLE_OPUS	1	Specifies if the OPUS codec capability of the phone is enabled or disabled. Value Operation: • 0: Disabled. • 1: Enabled OPUS wideband with bitrate of 20KBps. • 2: Enabled OPUS narrowband with bitrate of 16KBps. • 3: Eanbled OPUS narrowband with bitrate of 12KBps. **Note:** Avaya J129 IP Phone does not support third-party local call conference with OPUS.
ENABLE_PHONE_LOCK	0	Specifies whether the **Lock** softkey and lock feature button are enabled on the phone. If you enable the parameter, then a user can lock the phone by pressing the button or selecting the feature. Value Operation: • 0: Disabled. **Lock** softkey and feature button are not displayed. • 1: Enabled. **Lock** softkey and feature button are displayed.
ENABLE_PUBLIC_CA_CERTS	1	Specifies whether the out-of-the-box phone can validate server certificates against a list of well-known public Certificate Authority certificates Value operation: • 0: Embedded public CA certificates are only trusted when TRUSTCERTS is empty. • 1: Embedded public CA certificates are always trusted.
ENABLE_PPM_SOURCED_SIPPROXYSRVR The parameter is only available in an Avaya Aura® environment.	1	Enables PPM as a source of SIP proxy server information. Value Operation: • 0: Proxy server information received from PPM is not used. • 1: Proxy server information received from PPM is not used.

Table continues…

Parameter name	Default value	Description
ENABLE_PRESENCE	1	Specifies if presence will be supported. Value Operation: • 0: Disabled • 1: Enabled ✳ **Note:** This parameter is set to 0 in IP Office environment.
ENABLE_PRECEDENCE_SOFTKEY	1	Specifies that whether the precedence soft key is enabled or not on the idle line appearances on Phone Screen. Value Operation: • 0: Disabled. • 1: Enabled.
ENABLE_RECORDING	0	Specifies if audio debug recording is enabled for users. Value Operation: • 0: Audio debug recording is disabled. • 1: Audio debug recording is enabled.
ENABLE_REDIAL		Specifies if **Redial** softkey is available. Value Operation: • 0: No • 1: Yes
ENABLE_REDIAL_LIST		Specifies if the phone redials last number or displays list of recently dialed numbers. Value Operation: • 0: Last number redial • 1: User can select between the last redialled number and the redial list.
ENABLE_REMOVE_PSTN_ACCESS_PREFIX		Allows phone to perform digit manipulation during failure scenarios. This parameter allows removal of PSTN access prefix from the outgoing number. Value Operation; • 0: PSTN access prefix is retained in the outgoing number. • 1: PSTN access prefix is removed from the outgoing number.

Table continues…

Parameter name	Default value	Description
ENABLE_SHOW_EMERG_SK	2	Specifies whether an **Emergency** softkey, with or without a confirmation screen, is displayed when the phone is registered. All emergency numbers are always supported. Value Operation: • 0: **Emergency** softkey is not displayed. • 1: **Emergency** softkey is displayed without a confirmation screen. • 2: **Emergency** softkey is displayed with a confirmation screen. **Note:** The parameter is set to 0 for IP Office environment.
ENABLE_SHOW_EMERG_SK_UNREG	2	Specifies whether an **Emergency** softkey, with or without a confirmation screen, is displayed when the phone is not registered. All emergency numbers will always be supported. Value Operation: • 0: **Emergency** softkey is not displayed. • 1: **Emergency** softkey is displayed without a confirmation screen. • 2: **Emergency** softkey is displayed with a confirmation screen. **Note:** The parameter is set to 0 for IP Office environment.
ENCRYPT_SRTCP	0	Specifies whether RTCP packets are encrypted or not. SRTCP is only used if SRTP is enabled using MEDIAENCRYTIONRTCP. ENCRYPT_SRTCP parameter controls RTCP encryption for RTCP packets exchanged between peers. RTCP packets sent to Voice Monitoring Tools are always sent unencrypted. Value Operation: • 0: SRTCP is disabled. • 1: SRTCP is enabled.
ENFORCE_SIPS_URI	1	Specifies if a SIPS URI must be used for SRTP.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Not enforced • 1: Enforced
ENHDIALSTAT	1	Specifies if the algorithm defined by the parameter is used during certain dialing behaviors. Value Operation: • 0: Disables algorithm. • 1: Enables algorithm, but not for contacts. • 2: Enables algorithm including contacts. **Note:** The parameter is set to 0 for IP Office environment.
ENTRYNAME	0	Specifies if the calling party name, or the VDN or the skill name must be used in **History** entries. Value Operation: • 0: Calling Party Name is used. • 1: VDN or the skill name is used.
EVENT_NOTIFY_AVAYA_MAX_USERS	20	Specifies the maximum number of users to be included in an event notification message from CM/AST-II or Avaya Aura® Conferencing 6.0 or later. Valid values are 0 through 1000. This parameter is used only for development and debugging purposes.
ENABLE_WEBSERVER	1	Enables or disables the web server to configure the phones in a web browser. Value operation: • 0: Disable • 1: Enable
EXCHANGE_AUTH_USERNAME_FORMAT	0	Specifies the necessary format of the username for http authentication. Value operation: • 0: Office 2003/Office2016 username format. Username= <ExchangeUserDomain\ExchangeUserAccount> or Username= <ExchangeUserAccount> if <ExchangeUserDomain> is empty.

Table continues…

Parameter name	Default value	Description
		• 1: Office 365 format. Username= <ExchangeUserAccount@ExchangeUserDomain> or Username= <ExchangeUserAccount> if <ExchangeUserDomain> is empty.
EXCHANGE_EMAIL_DOMAIN	Null	Specifies the Exchange email domain. The value can contain 0 to 255 characters.
EXCHANGE_NOTIFY_SUBSCRIPTION_PERIOD	180	Specifies the number of seconds between re-syncs with the Exchange server. Valid values are 0 through 3600.
EXCHANGE_REMINDER_TIME	5	Specifies the number of minutes before an appointment at which a reminder will be displayed. Valid values are 0 through 60.
EXCHANGE_REMINDER_TONE	1	Specifies whether or not a tone will be generated the first time an Exchange reminder is displayed. Value Operation: • 0: Tone not generated. • 1: Tone generated.
EXCHANGE_SERVER_LIST	Null	Specifies a list of one or more Exchange server IP addresses. Addresses can be in dotted-decimal or DNS name format, separated by commas without any intervening spaces. The list can contain up to 255 characters.
EXCHANGE_SERVER_MODE	3	Specifies the protocol to be used to contact Exchange servers. Value Operation: • 1: Use WebDAV • 2: Use Exchange Web Services (EWS) • 3: Try EWS first, if that fails, try WebDAV.
EXCHANGE_SERVER_SECURE_MODE	1	Specifies if HTTPS should be used to contact Exchange servers. Value Operation • 0: Use HTTP • 1: Use HTTPS
EXCHANGE_SNOOZE_TIME	5	Specifies the number of minutes in which a reminder must be displayed again after it is temporarily dismissed.

Table continues…

Parameter name	Default value	Description
		Valid values are 0 through 60.
EXCHANGE_USER_DOMAIN	Null	Specifies the domain for the URL used to obtain Exchange contacts and calendar data. The parameter is used as a part of the user authentication. The value can contain 0 to 255 characters.
EXTEND_RINGTONE	Null	Provides a way to customize ring tone files. This is a comma separated list of file names in xml format.
F		
FAILED_SESSION_REMOVAL_TIMER	30	Specifies the number of seconds the phone displays a session line appearance and generates re-order tone after an invalid extension is dialed and user does not press the **End Call** softkey. Valid values are 5 through 999.
FAST_RESPONSE_TIMEOUT	4	Specifies the number of seconds the phone will waits before terminating an INVITE transaction if no response is received. Valid values are 0 through 32. Value of 0 means that this timer is disabled.
FIPS_ENABLED	0	Specifies whether only FIPS-approved cryptographic algorithms will be supported. Value Operation: • 0: No restriction on using non FIPS-approved cryptographic algorithms. • 1: Use only FIPS-approved cryptographic algorithms using embedded FIPS 140-2-validated cryptographic module.
FORBIDDEN_SESSION_REMOVAL_TIMER	10	Specifies the duration of an off-hook session before a call automatically ends. This is valid when there are no call appearances available on the called or remote party. Valid values are from 5 to 20 seconds.
FQDN_IP_MAP	Null	Specifies a comma separated list of name or value pairs where the name is an FQDN and the value is an IP address. The IP address may be IPv6 or IPv4 but the value can only contain one IP address. String length is up to 255 characters without any intervening spaces inside the string. The purpose of this parameter is to support cases where the server certificate Subject Common

Table continues…

Parameter name	Default value	Description
		Name of Subject Alternative Names includes FQDN, instead of IP address, and the SIP_CONTROLLER_LIST is defined using IP address. This parameter is supported with phone service running over TLS, however, the main use case is for Avaya Aura SM/PPM services. This parameter must not to be used as an alternative to a DNS lookup or reverse DNS lookup.
G		
G726_PAYLOAD_TYPE	110	Specifies the RTP payload type to be used for the G.726 codec. Valid values are 96 through 127.
GMTOFFSET	0:00	Specifies the time offset from GMT in hours and minutes. The format begins with an optional + or - (+ is assumed if omitted), followed by 0 through 12 (hours), followed by a colon (:), followed by 00 through 59 (minutes).
GROUP	0	Specifies specifically-designated groups of phones by using IF statements based on the GROUP parameter. The value of GROUP can be set manually in a phone by using the GROUP local admin procedure. The default value of GROUP in each phone is 0, and the maximum value is 999.
GUESTLOGINSTAT	0	Specifies whether the Guest Login feature is available to users. Value Operation: • 0: The feature is not available. • 1: The feature is availble..
GUESTDURATION	2	Specifies the duration (in hours) before a Guest Login or a visiting user login is automatically logged off if the phone is idle. Valid values are integers from 1 to 12.
GUESTWARNING	5	Specifies the number of minutes, before time specified by GUESTDURATION, that a warning of the automatic logoff is initially presented to the Guest or Visiting User. Valid values are integers from 1 to 15.
H		

Table continues…

Parameter name	**Default value**	**Description**
HANDSET_PROFILE_DEFAULT	1	Specifies the number of the default handset audio profile. Valid values are 1 through 20.
HANDSET_PROFILE_NAMES	Null	Specifies an ordered list of names to be displayed for handset audio profile selection. The list can contain 0 to 255 UTF-8 characters. Names are separated by commas without any intervening spaces. Two commas in succession indicate a null name, which means that the default name should be displayed for the corresponding profile. Names might contain spaces, but if any do, the entire list must be quoted. There is no way to prevent a profile from being displayed.
HTTPEXCEPTIONDOMAINS	Null	Specifies a list of one or more domains, separated by commas without any intervening spaces, for which HTTPPROXY is not used. The value can contain 0 to 255 characters. The default value is null.
HTTPPORT	80	Sets the TCP port used for HTTP file downloads from non-Avaya servers. Values range from 0 to 65535.
HTTPPROXY	Null	Specifies the address of the HTTP proxy server used by SIP phones to access an SCEP server that is not on the enterprise network. Valid value can contain zero or one IP address in dotted decimal or DNS name format, optionally followed by a colon and a TCP port number. The value can contain 0 to 255 characters.
HTTPSRVR	Null	Specifies zero or more HTTP server IP addresses to download configuration script files. The addresses must be separated by commas without any intervening spaces. The format of specifying IP addresses are: • Dotted decimal • Colon-hex • DNS name The parameter can be set by using LLDP. Valid values contains 0 to 255 ASCII characters.
I		

Table continues…

Parameter name	Default value	Description
ICMPDU		Specifies if ICMP Destination Unreachable messages are generated. Value Operation: • 0: No messages are generated. • 1: Limited port unreachable messages are generated. • 2: Protocol and port unreachable messages are generated.
ICMPRED		Specifies if received ICMP Redirect messages are processed. Value Operation: • 0: No • 1: Yes
INGRESS_DTMF_VOL_LEVEL	-12dBm	Specifies the power level of tone, expressed in dBm0. Values can range from -20dBm to -7dBm.
INSTANT_MSG_ENABLED	1	Specifies whether Instant Messaging is enabled or disabled. Value Operation: • 0: Disabled • 1: Enabled
INTER_DIGIT_TIMEOUT	5	Specifies the number of seconds that the phone waits after a digit is dialed before sending a SIP INVITE. Valid values are 1 through 10.
IPV6DADXMITS	1	Specifies whether Duplicate Address Detection is performed on tentative addresses, as specified in RFC 4862. Value operation: • 0: DAD is disabled • 1 to 5: Maximum number of transmitted Neighbor Solicitation messages.
IPV6STAT	0	Specifies whether IPv6 will be supported or not. Value operation: • 0: IPv6 will not be supported. • 1: IPv6 will be supported.

Table continues…

Parameter name	Default value	Description
K		
L		
L2Q	0	Specifies whether the VLAN tagging is enabled or disabled. Value Operation: • 0: Auto. VLAN tagging is turned on when the network can support VLAN tagging and L2QVLAN is non zero. • 1: On. VLAN tagging is turned on when the network can support VLAN tagging. The IP phone sends tagged frames with VLAN = L2QVLAN, even if L2QVLAN is set to 0. • 2: Off. VLAN functionality is disabled. **Note:** This parameter can also be set through: • Local admin procedure • A name equal to value pair in DHCPACK message • SET command in a settings file • DHCP option 43 • LLDP
L2QAUD	6	Specifies the value of the VLAN priority portion of the VLAN tag when the phone generates tagged Ethernet frames from the internal CPU of the phone. These values are inserted into the VLAN tag for audio frames (RTP, RTCP, SRTP, SRTCP). All other frames except those specified by the L2QSIG parameter are set to priority 0. Valid values are 0 through 7. **Note:** This parameter can also be set through: • SET command in a settings file • LLDP
L2QSIG	6	Specifies the value of the VLAN priority portion of the VLAN tag when the phone generates tagged Ethernet frames from the internal CPU of the phone. These values are inserted into the VLAN tag for signaling frames (SIP). All other frames

Table continues…

Parameter name	Default value	Description
		except those specified by the L2QAUD parameter are set to priority 0. Valid values are 0 through 7. ✳ **Note:** This parameter can also be set through: • SET command in a settings file • LLDP
L2QVLAN	0	Specifies the voice VLAN ID to be used by IP phones. Valid values are 0 through 4094. ✳ **Note:** This parameter can also be set through: • Local admin procedure • A name equal to value pair in DHCPACK message • SET command in a settings file • DHCP option 43 • LLDP
LANGLARGEFONT	Null	Specifies the name of the language file for the display of large text. The file name can contain 0-32 ASCII characters. When you set the parameter to the default value null, the **Text Size** option is not available.
LANGUAGES		Specifies the language files that must be installed or downloaded to the phone. Filenames can be full URL, relative pathname, or filename. Valid values can contain 0 to 1096 ASCII characters, including commas. Filenames must end in `.xml`
LLDP_ENABLED	2	Specifies whether LLDP is enabled. Value operation: • 0: Disabled • 1: Enabled • 2: Enabled, but only begins transmitting if an LLDP frame is received.

Table continues…

Parameter name	Default value	Description
LOCAL_CALL_PREFIX	DIAL_AS_IS	Sets the prefix for local calls. Permissible values are the Area Code denoted by AC, a string of digits, or the default, DIAL_AS_IS.
LOCAL_DIAL_AREA_CODE		Specifies if user must dial area code for calls within same area code regions. Value Operations: • 0: User does not need to dial area code. • 1: User need to dial area code. When enabled, the area code parameter (PHNLAC) should also be configured. ✱ **Note:** This parameter is supported when the phone is failed over.
LOCAL_LOG_LEVEL	3	Specifies the severity levels of events logged in the `endptRecentLog`, `endptResetLog`, and `endptStartupLog` objects in the SNMP MIB. Events with the selected severity level and above are logged. Lower numeric severity values correspond to higher severity levels Value Operation: • 0: Emergency events are logged. • 1: Alert and Emergency events are logged. • 2: Critical, Alert and Emergency events are logged. • 3: Error, Critical, Alert and Emergency events are logged (default). • 4: Warning, Error, Critical, Alert and Emergency events are logged. • 5: Notice, Warning, Error, Critical, Alert and Emergency events are logged. • 6: Informational, Notice, Warning, Error, Critical, Alert and Emergency events are logged. • 7: Debug, Informational, Notice, Warning, Error, Critical, Alert and Emergency events are logged

Table continues…

Parameter name	Default value	Description
		⚠ **Warning:** Setting the value to 7 can impact the performance of the phone because of the number of events generated.
LOCALLY_ENFORCE_PRIVACY_HEADER	0	Specifies whether the phone displays Restricted instead of CallerId information when a Privacy header is received in a SIP INVITE message for an incoming call. Value Operation: • 0: Disabled. CallerID information is displayed. • 1: Enabled. Restricted is displayed.
LOG_CATEGORY	Null	Specifies a list of categories of events to be logged through syslog and locally. This parameter must be specified to log events below the Error level. The list can contain up to 255 characters. Category names are separated by commas without any intervening spaces.
LOG_DIALED_DIGITS	1	Specifies if the call log will contain digits dialed by a user or information about a remote party when the user dials a FAC code. The FAC code is identified by * or # entered as a first character. Value Operation: • 0: Allow dialed FAC code to be replaced with a remote party number in the call history • 1: Dialed digits are logged in call history exactly as they were entered by the user (default).
LOGSRVR	Null	Specifies one address for a syslog server in dotted-decimal formatl (IPv4), colon-hex format (IPv6, if supported), or DNS name format. The value can contain 0 to 255 characters.
M		
MATCHTYPE	0	Specifies how an incoming or outgoing phone number is compared with the contacts on the phone to display the contact name. 0: Displays the contact name if all the digits match. 1: Displays the contact name if all the digits of the shorter number match with the right-most digits of

Table continues…

Parameter name	Default value	Description
		the longer number. For example, a 5-digit extension number can be matched with the 8-digit phone number saved in the contacts. 2: Displays the contact name if atleast the last four digits match. If the contacts are saved in multiple sources, for example, PPM, Exchange, or locally, the contact name saved first is displayed.
MAX_TRUSTCERTS	6	Specifies the maximum number of trusted certificates files defined by this parameter that can be downloaded to the phone. Valid values are from 1 to 10.
MEDIA_ADDR_MODE	4	Specifies the IP address of the endpoint when both IPv4 and IPv6 addresses are provided. This parameter is used for SIP signalling. Value operation: • 4: IPv4 • 6: IPv6 • 46: Prefer IPv4 over IPv6 • 64: Prefer IPv6 over IPv4
MEDIAENCRYPTION	9	Specifies which media encryption (SRTP) options is supported. 3 options are supported in a comma-separated list. Options must match to those specified in CM IP-codec-set form. • 1: aescm128-hmac80 • 2: aescm128-hmac32 • 3: aescm128-hmac80-unauth • 4: aescm128-hmac32-unauth • 5: aescm128-hmac80-unenc • 6: aescm128-hmac32-unenc • 7: aescm128-hmac80-unenc-unauth • 8: aescm128-hmac32-unenc-unauth • 9: none (default) • 10: aescm256-hmac80 • 11: aescm256-hmac32 The list of media encryption options is ordered from high (left) to the low (right) options. The phone publishs this list in the SDP-OFFER or chooses

Table continues…

Parameter name	Default value	Description
		from SDP-OFFER list according to the list order defined in MEDIAENCRYPTION. Avaya Aura® Communication Manager has the capability to change the list order in the SDP-OFFER (for audio only) when the SDP-OFFER is pass through. **Note:** You should not use unauthenticated media encryption (SRTP) files.
MEDIA_NEG_PREFERENCE	0	Specifies the address family preference used by a dual mode answer in non-Avaya environment. This parameter is not applicable for single mode phones. Value operation: • 0: Remote or offerer's preference • 1: Local
MEDIA_PRESERVATION	1	Supports media preservation when ENABLE_IPOFFICE is set to 2. Value operation: • 0: Phone tries to preserve a call for a duration specified by PRESERVED_CALL_DURATION settings parameter. • 1: Phone does not preserve a call. As soon as the phone detects link failure to IP Office, the phone drops a call and makes re-registration attempt.
MLPP_MAX_PREC_LEVEL	1	Specifies the maximum allowed precedence level for the user. Value Operation: • 1: Routine • 2: Priority • 3: Immediate • 4: Flash • 5: Flash Override
MLPP_NET_DOMAIN	Null	Specifies the MLPP network domain. Value Operation: • Null: No domain configured

Table continues…

Parameter name	Default value	Description
		• DSN: DSN network. • UC: UC network.
MSGNUM		Specifies the phone number to be dialed automatically when the user presses the Message button. The phone number connects to the user's voice mail system. **Note:** This parameter is applicable in Avaya Aura environment. In case of IP Office and third party environment, use the parameter PSTN_VM_NUM.
MUTE_ON_REMOTE_OFF_HOOK	0	Controls the speakerphone muting for a remote-initiated (a shared control or OOD-REFER) speakerphone off-hook. Value Operation: • 0: The speakerphone is unmuted. • 1: The speakerphone is muted. The value is applied to the phone only when the phone is deployed with a Avaya Aura® Communication Manager 6.2.2 and earlier releases. If the phone is deployed with Avaya Aura® Communication Manager 6.3 or later, the setting is ignored. Instead the feature is delivered through PPM. The Turn on mute for remote off-hook attempt parameter is enabled in the station form through the Avaya Aura® Session Manageror Avaya Aura® Communication Manager (SAT) administrative interfaces. **Note:** This parameter is set to 0 in IP Office environment.
MWISRVR	Null	Specifies a list of addresses of Message Waiting Indicator servers. Addresses can be in dotted-decimal or DNS name format, separated by commas without any intervening spaces. The value can contain 0 to 255 characters.
MYCERTCAID	CAIdentifier	Specifies an identifier for the CA certificate with which the SCEP certificate request is to be signed, if the server hosts multiple Certificate Authorities.

Table continues…

Parameter name	Default value	Description
		The value can contain zero to 255 ASCII characters. The parameter is only available in an Avaya Aura® environment.
MYCERTCN	$SERIALNO	Specifies the Common Name (CN) used in the SUBJECT of an SCEP certificate request. The value must be a string that contains either $SERIALNO" (which will be replaced by the phone's serial number) or $MACADDR (which will be replaced by the phone's MAC address), but it can contain other characters as well, including spaces. The value can contain eight ($MACADDR) to 255 characters.
MYCERTDN	Null	Specifies the part the SUBJECT of an SCEP certificate request that is common for all phones. The value must begin with a / and can include Organizational Unit, Organization, Location, State and Country. The value can contain Zero to 255 ASCII characters. **Note:** / must used as a separator between components. Commas do not work with some servers
MYCERTKEYLEN	2048	Specifies the bit length of the public and private keys generated for the SCEP certificate request. The value is a 4 ASCII numeric digits. The phone supports only value 2048.
MYCERTRENEW	90	Specifies the percentage of the identity certificate's validity interval after which renewal procedure is initiated. Valid values are 1 through 99.
MYCERTURL	Null	Specifies the URL of the SCEP server for obtaining an identity certificate. The URL can be HTTP or HTTPS. The valid values can range from Zero to 255 ASCII characters.

Table continues…

Parameter name	Default value	Description
MYCERTWAIT	1	Specifies the phone's behavior if the SCEP server indicates that the certificate request is pending for manual approval. Value Operation: • 0: Poll the SCEP server periodically in the background. • 1: Wait until a certificate is received or the request is rejected.
N		
NO_DIGITS_TIMEOUT	20	Specifies the number of seconds the phone waits for a digit to be dialed after going off-hook and before generating a warning tone. Valid values are 1 through 60.
O		
OCSP_ACCEPT_UNK	1	Specifies whether in cases where certificate revocation status for a specific certificate cannot be determined to bypass certificate revocation operation for this certificate. Value operation: • 0: Certificate is considered to be revoked if the certificate revocation status is unknown. TLS connection will be closed. • 1: Certificate revocation operation will accept certificates for which the certificate revocation status is unknown.
OCSP_CACHE_EXPIRY	2880	Specifies the time interval for the OCSP cache expiry in minutes. OCSP response cache expiry uses nextUpdate value in OCSP response message. If nextUpdate is not present, then OCSP_CACHE_EXPIRY parameter value is used. Valid range is from 60 to 10080
OCSP_ENABLED	0	Specifies that OCSP is used to check the revocation status of the certificates. Value operation: • 0: Disabled. Certificate revocation checking is not performed. • 1: Enabled. Certificate revocation checking is performed.
OCSP_HASH_ALGORITHM	0	Specifies the hashing algorithm for OCSP request.

Table continues…

Parameter name	Default value	Description
		Value operation: • 0: SHA1 hash algorithm • 1: SHA256 hash algorithm
OCSP_NONCE	1	Specifies whether a nonce is added in OCSP requests and expected in OCSP responses. Value operation: • 0: Not added to OCSP request. • 1: Added to OCSP request.
OCSP_TRUSTCERTS		Specifies a comma separated list of OCSP trusted certificates that are used as OCSP signing authority for checking the revocation status of the certificate. This applies to when the OCSP responder is using a different CA. Spaces are not permitted in this parameter.
OCSP_URI	Null	Specifies the URI of an OCSP responder. The URI can be an IP address or hostname. Valid values contain 0 to 255 ASCII characters, zero or one URI.
OCSP_USE_CACHE	1	Specifies that the OCSP caching is in use. Value operation: • 0: OCSP is not used. Always check with OCSP responder. • 1: OSCP cache caching is used.
OCSP_URI_PREF	1	Specifies the preferred URI for use in an OCSP request when more than one source is available. Value operation: • 1: Use the OCSP_URI and then the OCSP field of the Authority Information Access (AIA) extension of the certificate. • 2: Use the OCSP field of the Authority Information Access (AIA) extension of the certificate and then the OCSP_URI.
OUTBOUND_SUBSCRIPTION_REQUEST_DURATION	86400	Specifies the duration in seconds requested by the phone in SUBSCRIBE messages, which can be decreased depending on the response from the server. Valid values are 60 through 31536000 (one year). The default value is 86400 (one day).
OPUS_PAYLOAD_TYPE	116	Dynamically specifies the RTP payload type to be used for OPUS codec. The parameter is used

Table continues…

Parameter name	Default value	Description
		when the media request is sent to the far-end in an INVITE or 200 OK when INVITE with no Session Description Protocol (SDP) is received. The range is between 96 to 127.
P		
PHNCC	1	Specifies the country code for United States. The value is 1. Valid values 1 through 999.
PHNDPLENGTH	5	Specifies the internal extension number length. If your extension is 12345, and your dial plan length is 5. The maximum extension length is 13. This value must match the extension length set on your call server. Valid values are 3 through 13.
PHNEMERGNUM	Null	Specifies an emergency phone number to be dialed if the associated button is selected. Valid values can contain up to 30 dialable characters (0 to 9, *, #).
PHNMOREEMERGNUMS	Null	Specifies list of emergency numbers separated by comma. Valid values may contain up to 30 dialable characters (0 to 9, *, #).
PHNIC	011	Specifies the international access code For the United States, the value is 011. Valid values are from 0 to 4 dialable characters (0-9,*,#).
PHNLAC		Phone's Local Area Code indicates the phone's local area code, which along with the parameter LOCAL_DIAL_AREA_CODE, allows users to dial local numbers with more flexibility. PHNLAC is a string representing the local area code the phone. **Note:** This parameter is supported when the phone is failed over.
PHNLD	1	Specifies the long distance access code Valid values are 0 through 9 and empty string. If long distance access code is not needed then set the parameter to null.

Table continues…

Parameter name	Default value	Description
PHNLDLENGTH	10	Specifies the national phone number length. For example, 800-555-1111 has a length of 10. Valid values are 5 through 15.
PHNMUTEALERT_BLOCK	1	Specifies if the **Mute Alert** feature is blocked or unblocked. Value Operation: • 0: Unblocked • 1: Blocked
PHNNUMOFSA	3	Specifies the number of session appearances the phone must support while operating in a non-Avaya environment. Valid values are 1 through 10.
PHNOL	9	Specifies the outside line access code. This is the number you press to make an outside call. Valid values are 0 to 2 dialable characters (0-9, *, #).
PHONE_LOCK_IDLETIME	0	Specifies the interval of idle time, in minutes, after which the phone will automatically lock. The phone will lock irrespective of the value of ENABLE_PHONE_LOCK.
PHY1STAT	1	Specifies the speed and duplex settings for the Ethernet line interface. Value Operation: • 1: auto-negotiate • 2: 10Mbps half-duplex • 3: 10Mbps full-duplex • 4: 100Mbps half-duplex • 5: 100Mbps full-duplex • 6: 1Gbps full-duplex, if supported by hardware, otherwise auto-negotiated
PHY2_AUTOMDIX_ENABLED	1	Specifies whether auto-MDIX is enabled on PHY2. Value Operation: • 0: auto-MDIX is disabled. • 1: auto-MDIX is enabled.
PHY2PRIO	0	Specifies the layer 2 priority value to be used for frames received on the secondary Ethernet interface when VLAN separation is enabled. The

Table continues…

Parameter name	Default value	Description
		parameter is not supported when VLANSEPMODE is 1. Valid values are 0 through 7.
PHY2STAT	1	Specifies the speed and duplex settings for the secondary (PC) Ethernet interface. Value Operation: • 0: disabled • 1: auto-negotiate • 2: 10Mbps half-duplex • 3: 10Mbps full-duplex • 4: 100Mbps half-duplex • 5: 100Mbps full-duplex • 6: 1Gbps full-duplex, if supported by hardware, otherwise auto-negotiated
PHY2TAGS	0	Determines whether or not VLAN tags are stripped on Ethernet frames going out of the Computer (PC) port. Value Operation: • 0: Strip tags. VLAN tags are stripped from Ethernet frames leaving the computer (PC) port of the phone. • 1: Does not strip tags. VLAN tags are not stripped from Ethernet frames leaving the Computer (PC) port of the phone. **Note:** This parameter is configured through the settings file.
PHY2VLAN	0	Specifies the value of the 802.1Q VLAN ID that is used to identify network traffic going into and coming out of the internal CPU of the phone. Valid values are 0 through 4094. **Note:** The parameter is configured through the following: • SET command in a settings file • LLDP

Table continues…

Parameter name	Default value	Description
PKCS12URL	Null	Specifies the URL to be used to download a PKCS #12 file containing an identity certificate and its private key. Valid values contain 0 to 255 ASCII characters, zero or one URL. The value can be a string that contains either $SERIALNO or $MACADDR, but it may contain other characters as well. If $MACADDR is added to the URL, then the PKCS12 filename on the file server includes MAC address without colons. PKCS12 file download is preferred over SCEP if PKCS12URL is defined.
PKCS12_PASSWD_RETRY	3	Specifies the number of retries for entering PKCS12 file password. If user failed to enter the correct PKCS12 file password after PKCS12_PASSWD_RETRY retries, then the phone will continue the startup sequence without installation of PKCS12 file. Valid values are from 0 to 100. Value operation: • 0: No retry
PLAY_TONE_UNTIL_RTP	1	Specifies whether locally-generated ringback tone stops as soon as SDP is received for an early media session, or whether it will continue until RTP is actually received from the far-end party. Value Operation: • 0: Stop ringback tone as soon as SDP is received. • 1: Continue ringback tone until RTP is received (default).
PLUS_ONE	0	Specifies if pressing **1** on the dialpad during dialing a number will alternate between 1 and +. Value Operation: • 0: Dials 1. • 1: Alternates between 1 and +. When the user goes off-hook, the user can use the star key (*) to append the plus (+) symbol.
POE_CONS_SUPPORT		Enables power over Ethernet conservation mode. Value Operation: • 0: Power conservation mode is not supported. • 1: Power conservation mode is supported.

Table continues…

Parameter name	Default value	Description
PRESENCE_ACL_CONFIRM	0	Specifies the handling of a Presence ACL update with pending watchers. Value Operation: • 0: Auto confirm. Automatically send a PUBLISH to allow presence monitoring (default). • 1: Ignore. Take no action This parameter is not supported in IP Office environment as presence is not supported.
PRESENCE_SERVER	Null	Specifies the address of the Presence server. This parameter is supported only for backward compatibility. The value of this parameter is used from PPM and not from the settings file. This parameter is not supported in IP Office environment as presence is not supported.
PRESERVED_CALL_DURATION	120	Specifies the time interval in minutes if ENABLE_IPOFFICE is set to 2 and if MEDIA_PRESERVATION is set to 1 . The time interval can be from 10 minutes to 120 minutes.
PROCPSWD	27238	Specifies an access code to access the admin menu procedures. Valid values contain 0 through 7 ASCII numeric digits. The default value is 27238 unless indicated otherwise below. A null value implies that an access code is not required for access. **Note:** • Setting this parameter through PPM is more secure because this file can usually be accessed and read by anyone on the network. Setting the value in this file is intended primarily for configurations with versions of phone or if server software that do not support setting this value from the server. • For enhanced security, use ADMIN_PASSWORD instead of PROCPSWD.
PROCSTAT	0	Specifies an access code to access the admin menu procedures.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Local procedures can be used (default). • 1: Local procedures cannot be used.
PROVIDE_CF_RINGTONE	0	Specifies if the call forward ringtone option is provided to the user. Value Operation: • 0: The call forward ringtone option is not provided (default). • 1: The call forward ringtone option is provided.
PROVIDE_EXCHANGE_CALENDAR	1	Specifies if menu items for exchange calendar are displayed. Value Operation: • 0: Not displayed • 1: Displayed (default)
PROVIDE_EXCHANGE_CONTACTS	1	Specifies if menu items for exchange contacts are displayed. Value Operation: • 0: Not displayed • 1: Displayed (default)
PROVIDE_KEY_REPEAT_DELAY	0	Specifies how long a navigation button must be held down before it begins to auto-repeat, and if an option is provided by which the user can change this value. Value Operation: • 0: Default (500ms) with user option (default). • 1: Short (250ms) with user option. • 2: Long (1000ms) with user option. • 3: Very Long (2000ms) with user option. • 4: No Repeat with user option. • 5: Default (500ms) without user option. • 6: Short (250ms) without user option. • 7: Long (1000ms) without user option. • 8: Very Long (2000ms) without user option. • 9: No Repeat without user option.
PROVIDE_LOGOUT		Specifies if user can log out from the phone.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: No • 1: Yes **Note:** This parameter is set to 0 in IP Office environment.
PROVIDE_NETWORKINFO_SCREEN		Specifies if the **Network Information** menu is displayed on the phone. Value Operation: • 0: No • 1: Yes
PROVIDE_OPTIONS_SCREEN		Specifies if **Options & Settings** menu is displayed on phone. Value Operation: • 0: No • 1: Yes
PROVIDE_TRANSFER_TYPE	0	Provides the call transfer type in 3rd party environments. Value 0 or 1.
PUSHCAP	0000	Controls the modes of individual push types. The value is a 3, 4 or 5 digit number, of which each digit controls a push type and can have a value of 0, 1 or 2. Value Operation: • 0: Push requests are ejected for that push type. • 1: Only push requests with a mode of barge are accepted for that push type. • 2: Push requests with a mode of barge or normal are accepted for that push type. The Push types controlled by each digit (11111) are as follows: • ||||+- The rightmost digit controls top line Push requests. • |||+-- The next digit to the left controls display (WML browser) push requests.

Table continues…

Parameter name	Default value	Description
		• ||+--- The next digit to the left controls receive audio push requests. • |+---- The next digit to the left controls transmit audio push requests. • +----- The next digit to the left controls phonexml push requests.
PUSHPORT	80	Specifies the TCP port number to be used by the HTTP server in the phone for push. Valid values are 80 through 65535.
Q		
QLEVEL_MIN	1	Specifies the minimum quality level for which a low local network quality indication will not be displayed. Value Operation: • 1: Never display icon (default) • 2: Packet loss is > 5% or round trip network delay is > 720ms or jitter compensation delay is > 160ms. • 3: Packet loss is > 4% or round trip network delay is > 640ms or jitter compensation delay is > 140ms. • 4: Packet loss is > 3% or round trip network delay is > 560ms or jitter compensation delay is > 120ms. • 5: Packet loss is > 2% or round trip network delay is > 480ms or jitter compensation delay is > 100ms. • 6: Packet loss is > 1% or round trip network delay is > 400ms or jitter compensation delay is > 80ms.
R		
RDS_INITIAL_RETRY_ATTEMPTS	15	Specifies the number of retries after which the phone abandons its attempt to contact the PPM server. Valid values are 1 through 30.
RDS_INITIAL_RETRY_TIME	2	Specifies the number of seconds that the phone waits for the first time before trying to contact the PPM server again after a failed attempt. Each subsequent retry is delayed by double the previous delay.

Table continues…

Parameter name	Default value	Description
		Valid values are 2 through 60.
RDS_MAX_RETRY_TIME	600	Specifies the maximum delay interval in seconds after which the phone abandons its attempt to contact the PPM server. Valid values are 2 through 3600.
RECORDINGTONE	0	Specifies whether call recording tone is generated on active calls. Value Operation: • 0: Call recording tone is not generated (default). • 1: Call recording tone is not generated.
RECORDINGTONE_INTERVAL	15	Specifies the number of seconds between call recording tones. Valid values are 1 through 60.
RECORDINGTONE_VOLUME	0	Specifies the volume of the call recording tone in 5dB steps. Value Operation: • 0: The tone volume is equal to the transmit audio level (default). • 1: The tone volume is 45dB below the transmit audio level. • 2: The tone volume is 40dB below the transmit audio level. • 3: The tone volume is 35dB below the transmit audio level. • 4: The tone volume is 30dB below the transmit audio level. • 5: The tone volume is 25dB below the transmit audio level. • 6: The tone volume is 20dB below the transmit audio level. • 7: The tone volume is 15dB below the transmit audio level. • 8: The tone volume is 10dB below the transmit audio level. • 9: The tone volume is 5dB below the transmit audio level. • 10: The tone volume is equal to the transmit audio level.

Table continues…

Parameter name	Default value	Description
RECOVERYREGISTERWAIT	60	Specifies a number of seconds. If no response is received to a REGISTER request within the number of seconds specified by WAIT_FOR_REGISTRATION_TIMER, the phone will try again after a randomly selected delay of 50% to 90% of the value of RECOVERYREGISTERWAIT. Valid values are 10 through 36000.
REDIRECT_TONE	1	Specifies the tone to play when a call goes to coverage. Valid values are from 1 to 4.
REGISTERWAIT	900	Specifies the number of seconds between re-registrations with the current server. Valid values are from 30 to 86400.
REUSETIME	60	Specifies the number of seconds that the DHCP is attempted: • With a VLAN ID of zero. True when L2Q is set to 1. • Or with untagged frames. True if L2Q is set to 0 or 2. • And before reusing the IP address and the associated address information, that the phone had the last time it successfully registered with a call server. While reusing an address, DHCP enters the extended rebinding state described above for DHCPSTD. Valid values are 0 and 20 through 999. The default value is 60. A value of zero means that DHCP will try forever and there will be no reuse.
RINGTONES	Null	Specifies a list of display names and file names or URLs for a custom ring tone files to be downloaded and offered to users. The list can contain 0 to 1023 UTF-8 characters. The default value is null. Values are separated by commas without any intervening spaces. Each value consists of a display name followed by an equals sign followed by a file name or URL. Display names can contain spaces, but if any do, the entire list must be quoted. Ring tone files must be single-channel

Table continues…

Parameter name	Default value	Description
		WAV files coded in ITU-T G.711 u-law or A-law PCM with 8-bit samples at 8kHz.
RINGTONES_UPDATE	0	Specifies if the phone queries the file server to determine if there is an updated version of each custom ring tone file each time the phone starts up or resets. Value Operation: • 0: Phone only tries to download ring tones with new display names. • 1: Phone checks for updated version of each ring tone file at startup.
RINGTONESTYLE	0	Specifies the style of ring tones that are offered to the user for personalized ringing when **Classic** is selected, as opposed to **Rich**. Value Operation: • 0: North American ring tones are offered (default). • 1: European ring tones are offered.
RTCP_XR	0	Specifies if VoIP Metrics Report Block as defined in RTP Control Protocol Extended Reports (RTCP XR) (RFC 3611) is sent as part of the RTCP packets to remote peer or to RTCP monitoring server. Value Operation: • 0: No • 1: Yes
RTCPCONT		Specifies if the sending of RTCP is enabled. Value Operation: • 0: No • 1: Yes
RTCPMON	Null	Specifies the IP or DNS address for the RTCP monitor. You can set this parameter only if the environment is not an Avaya environment. The values can range from 0 through 255 characters.
RTCPMONPERIOD	5	Specifies the interval, in seconds, for sending out RTCP monitoring reports. Valid values are from 5 to 30 seconds.
RTCPMONPORT	5005	Specifies the RTCP monitor port number.

Table continues…

Parameter name	Default value	Description
		You can set this parameter only if the environment is not an Avaya environment. The values can range from 0 through 65535. Default is 5005.
RTP_PORT_LOW		Specifies the lower limit of the UDP port range to be used by RTP or RTCP and SRTP or SRTCP connections. The values can range from 1024 through 65503.
RTP_PORT_RANGE		Specifies the range or number of UDP ports available for RTP or RTCP and SRTP or SRTCP connections This value is added to RTP_PORT_LOW to determine the upper limit of the UDP port range. The values can range from 32 through 64511.
S		
SCEPPASSWORD	$SERIALNO	Specifies the password to be included in the change password attribute of an SCEP certificate request. Values can contain 0 to 32 ASCII characters (50 ASCII characters. If the value contains $SERIALNO, it is replaced by the phone's serial number. If the value contains $MACADDR, it is replaced by the phone's MAC address in hex. **Note:** • A password prompt is invoked when SCEP is set for identity certificate enrollment and the parameter value is empty. • This parameter must not be set in a file that is accessible on an enterprise network, and only in a restricted staging configuration.
SCREENSAVERON	240 (4 hours)	Specifies the number of minutes of idle time after which the screen saver is displayed. If an image file is downloaded based on the LOGOS and CURRENT_LOGO parameter, it is used as the screen saver. Otherwise, the built-in Avaya one-X(TM) screen saver is used. Valid values are 0 through 999. The default value is 240 (4 hours). A value of 0 means that the screen saver will not be displayed automatically when the phone is idle.

Table continues…

Parameter name	Default value	Description
SCREENSAVER_IMAGE	N/A	Specifies the screen saver images those can be loaded from the provisioning server. Maximum five custom images can be uploaded onto the phone. Only the .jpeg file format are supported and the maximum file size is 256KB. Note that the image file name is case sensitive.
SCREENSAVER_IMAGE_DISPLAY	N/A	Allows the administrator to display the desired screen saver image. Note that If BACKGROUND_IMAGE_SELECTABLE is set to 1 then the end user may override this setting.
SCREENSAVER_IMAGE_SELECTABLE	1	Allows the end user to select and change the screen saver images. Value operation: • 0: End user can not select and change the screen saver images from the settings menu. • 1: End user can select and change the screen saver images from the settings menu.
SDPCAPNEG	1	Specifies if SDP capability negotiation is enabled. Value Operation: • 0: SDP capability negotiation is disabled. • 1: SDP capability negotiation is enabled.
SEND_DTMF_TYPE	2	Specifies if DTMF tones are sent in-band as regular audio, or out-of-band using RFC 2833 procedures. Value Operation: • 1: In-band • 2: Out-of-band
SERVER_CERT_RECHECK_HOURS	24	Specifies the number of hours after which certificate expiration and OCSP will be used, if OCSP is enabled, to recheck the revocation and expiration status of the certificates that were used to establish a TLS connection. Valid values are from 0 to 32767. Value operation: • 0: Periodic checking is disabled.
SIG	0	Specifies the type of software to be used by the phone by controlling which upgrade file is requested after a power-up or a reset. Value Operation

Table continues…

Parameter name	Default value	Description
		0: Download the upgrade file for the same signaling protocol that is supported by the current software (default) 2: Download J100Supgrade.txt
SIGNALING_ADDR_MODE	4	Specifies the SIP controller IP address from SIP_CONTROLLER_LIST_2. This parameter is used by SIP signaling on a dual mode phone. Value operation: • 4: IPv4 • 6: IPv6
SIG_PORT_LOW		Specifies the minimum port value for SIP signaling. (1024 -65503).
SIG_PORT_RANGE		Specifies the range or number of SIP signaling ports. This value is added to SIG_PORT_LOW to determine the upper limit of the SIP signaling port range (32-64511).
SIMULTANEOUS_REGISTRATIONS	3	Specifies the number of Session Managers with which the phone simultaneously register. Valid values are 1, 2 or 3. The default value is 3. **Note:** This parameter is set to 1 in IP Office environment.
SIP_CONTROLLER_LIST	Null	Specifies a list of SIP controller designators, separated by commas without any spaces. Controller designator has the following format: host[:port][;transport=xxx], where host is an proxy address in dotted-decimal or DNS name format. In third-party call control setup, only DNS format is supported. [:port] is an optional port number. [;transport=xxx] is an optional transport type In third-pary call control setup, only one SIP controller is supported.
SIP_CONTROLLER_LIST_2	Null	Specifies the registration address. This parameter replaces SIP_CONTROLLER_LIST for dual mode phones. The parameter contains a comma separated list of SIP proxy or registrar servers. The list has the following format: host[:port][;transport=xxx].

Table continues…

Parameter name	Default value	Description
SIPCONFERENCECONTINUE	0	Specifies if a conference call continues after the host hangs up. Value Operation: • 0: Drop all parties. • 1: Continue conference ✳ **Note:** This parameter is set to 1 in IP Office environment.
SIPDOMAIN	Null	Specifies the domain name to be used during SIP registration. The value can contain 0 to 255 characters. The default value is null.
SIPPORT	5060	Specifies the port the phone opens to receive SIP signaling messages. Valid values are 1024 through 65535. The default value is 5060.
SIPREGPROXYPOLICY	Simultaneous	Specifies if the phone attempts to maintain one or multiple simultaneous registrations. Value Operation: • Alternate: Only a single registration is attempted and maintained. • Simultaneous: Simultaneous registrations is attempted and maintained with all available controllers.
SIPSIGNAL	2	Specifies the type of transport used for SIP signaling. Value Operation: • 0: UDP • 1: TCP • 2: TLS
SKILLSCREENTIME	5	Specifies the duration, in seconds, that the **Skills** screen is displayed. Valid values are 0 through 60. The default value is 5. A value of 0 means that the **Skills** screen in not removed automatically when the agent logs in.
SLMCAP	0	Specifies if the SLA Monitor agent is enabled for packet capture.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Disabled (default) • 1: Enabled and payloads are removed from RTP packets • 2: Enabled and payloads are included in RTP packets • 3: Controlled from admin menu - Allows you to enable or disable of RTP packets capture using local admin procedures.
SLMCTRL	0	Specifies whether the SLA Monitor agent is enabled for phone control. Value Operation: • 0: Disabled • 1: Enabled • 2: Controlled from admin menu.
SLMPERF	0	Specifies whether the SLA Monitor agent is enabled for phone performance monitoring. Value Operation: • 0: Disabled • 1: Enabled
SLMPORT	50011	Specifies the UDP port that will be opened by the SLA Monitor agent to receive discovery and test request messages. Valid values are 6000 through 65535. The default value is 50011. **Note:** If default port is not used, both the SLA Mon agent and the server must be configured with the same port. This parameter impacts the phone's SLA Mon agent configuration. A corresponding configuration must also be made on the SLA Mon server `agentcom-slamon.conf` file.
SLMSRVR		Specifies the IP address and the port number of the SLA Mon server in the aaa.bbb.ccc.ddd:n format. Set the IP address of the SLA Mon server in the aaa.bbb.ccc.ddd format to restrict the registration of agents only to that server.

Table continues…

Parameter name	Default value	Description
		Specifying a port number is optional. If you do not specify a port number, the system takes 50011 as the default port. If the value of the port number is 0, than any port number is acceptable. The IP address must be in the dotted decimal format, optionally followed by a colon and an integer port number from 0 to 65535. To use a non-default port, set the value in the aaa.bbb.ccc.ddd:n format, where aaa.bbb.ccc.ddd is the IP addressof the SLA Mon server. **Note:** If default port is not used, both the SLA Mon agent and server must be configured with the same port. SLMSRVR impacts the phone's SLA Mon agent configuration. A corresponding configuration must also be made on the SLA Mon server `agentcom-slamon.conf` file
SLMSTAT	0	Specifies if the SLA Monitor agent is enabled or not. Value Operation: • 0: Disabled • 1: Enabled
SNMPADD	Null	Specifies a list of source IP addresses from which SNMP query messages will be accepted and processed. Addresses can be in dotted-decimal format (IPv4), colon-hex format (IPv6, if supported), or DNS name format, separated by commas without any intervening spaces. The list can contain up to 255 characters. The default value is null.
SNMPSTRING	Null	Specifies a security string that must be included in SNMP query messages for the query to be processed. Valid values contain 0 through 32 ASCII alphanumeric characters. The default value is null. Null disables SNMP.
SNTPSRVR	Null	Specifies a list of addresses of SNTP servers.

Table continues…

Parameter name	Default value	Description
		Addresses can be in dotted-decimal or DNS name format, separated by commas without any intervening spaces. The list can contain up to 255 characters.
SOFTKEY_CONFIGURATION	0,1,2	Specifies which feature will show up on which softkey on the Avaya J100 Series IP Phones screens. The features are defined as follows: • 0 = Redial • 1 = Contacts • 2 = Emergency • 3 = Recents • 4 = Voicemail
SNTP_SYNC_INTERVAL	1440 minutes	Specifies the time interval, in minutes, during which the phone will attempt to synchronize its time with configured NTP servers. Valid values are from 60 to 2880 minutes.
SP_DIRSRVR	Null	Sets the IP address or Fully-Qualified Domain Name (FQDN) of the LDAP Directory Server. Valid values are zero or more IP addresses in dotted-decimal or DNS format, separated by commas without intervening spaces, to a maximum of 255 ASCII characters. The default is null.
SPEAKERSTAT	2	Specifies the operation of the speakerphone. Value Operation: • 0: Speakerphone disabled • 1: One-way speaker (also called monitor) enabled. • 2: Full (two-way) speakerphone enabled.
SSH_ALLOWED	2	Specifies if SSH is supported. Value Operation: • 0: Disabled • 1: Enabled • 2: Configured using local admin procedure. When this mode is configured, then by default the SSH server is disabled.
SSH_BANNER_FILE	Null	Specifies the file name or URL for a custom SSH banner file.

Table continues…

Parameter name	Default value	Description
		If the value is null, english banner is used for SSH. The value can contain 0 to 255 characters.
SSH_IDLE_TIMEOUT	10	Specifies the idle time in minutes after which an SSH connection is terminated Valid values are 0 through 32767. A value of 0 means that the connection will not be terminated.
SUBSCRIBE_LIST_NON_AVAYA		Specifies comma separated list of event packages to subscribe to after registration. Possible values are: reg, dialog, mwi, ccs, message-summary which is identical to mwi, avaya-ccs-profile which is identical to ccs. The values are case insensitive. For IPO the recommended value shall be reg, message-summary, avaya-ccs-profile.
SUBSCRIBE_SECURITY		Specifies the use of SIP or SIPS for subscriptions. Value Operation: • 0: The phone uses SIP for both the request URI and the contactheader regardless of whether SRTP is enabled. • 1: The phone uses SIPS for both the request URI and the contact header if SRTP is enabled. TLS is on and MEDIAENCRYPTION has at least one valid crypto suite. • 2: SES or PPM does not show a FS-phoneData FeatureName with a Feature Version of 2 in the response to the getHomeCapabilities request. For IP office environment, the applicable values are 0 and 1.
SUBSCRIBELIST	Null	Specifies a list of URIs to which the phone will send a subscribe message after the phone successfully registers with a call server, or when a subscribe push request is received with a type attribute all. The message is an HTTP GET for the URI with the phone's MAC address, extension number, IP address and model number appended as query values) The list can contain up to 255 characters. Values are separated by commas without any intervening spaces.

Table continues…

Parameter name	Default value	Description
		If the value is set to null, subscribe messages are not sent.
SYMMETRIC_RTP	1	Specifies if the phone must discard received RTP or SRTP datagrams if their UDP source port number is not the same as the UDP destination port number included in the RTP or SRTP datagrams of that endpoint. Value Operation: • 0: Ignore the UDP source port number in received RTP/SRTP datagrams. • 1: Discard received RTP/SRTP datagrams if their UDP Source Port number does not match the UDP Destination Port number that the phone includes in RTP/SRTP datagrams intended for that phone.
SYSTEM_LANGUAGE		Contains the name of the default system language file used in the phone. The filename should be one of the files listed in the LANGUAGES parameter. If no filename is specified, or if the filename does not match one of the LANGUAGES values, the phone uses the built-in English text strings. Valid values range from 0 through 32 ASCII characters. Filename must end in .xml
T		
TCP_KEEP_ALIVE_STATUS	1	Specifies if the phone sends TCP keep alive messages. Value Operation: • 0: Keep-alive messages are not sent. • 1: Keep-alive messages are sent (default).
TCP_KEEP_ALIVE_INTERVAL	10	Specifies the number of seconds that the telephone waits before re-transmitting a TCP keep-alive (TCP ACK) message. Valid values are from 5 through 60.
TCP_KEEP_ALIVE_TIME	60	Specifies the number of seconds that the telephone waits before sending out a TCP keep-alive (TCP ACK) message. Valid values are from 10 through 3600
TEAM_BUTTON_REDIRECT_INDICATION	0	Specifies if the redirection indication must be shown on a team button on the monitored station, if

Table continues…

Parameter name	Default value	Description
		it is not a redirect destination of the monitored station. Value Operation: • 0: Disabled. The redirect indication is shown only on a monitoring station which is redirection destination. • 1: Enabled. The redirection indication is displayed on all monitoring stations.
TEAM_BUTTON_RING_TYPE	1	Specifies the alerting pattern to use for team buttons. Valid values are 1 through 8. The default value is 1.
TIMEFORMAT		Specifies the format for time displayed in the phone. Value Operation: • 0: AM or PM format. • 1: 24hour. format
TLS_VERSION	0	Specifies the TLS version used for all TLS connections (except SLA monitor agent) Value Operation 0: TLS versions 1.0 and 1.2 are supported. 1: TLS version 1.2 only is supported.
TLSSRVR		Specifies zero or more HTTPS server IP addresses, which is used to download configuration script files. The IP addresses can be specified in dotted-decimal, or DNS name format separated by commas without any intervening spaces. Valid values contain 0 to 255 ASCII characters, including commas. This parameter can also be changed through LLDP.
TLSSRVRID	1	Specifies how a phone evaluates a certificate trust. Value Operation: • 0: Identity matching is not performed. • 1: The certificate is trusted only if the identity used to connect to the server matches the certificate identity, as per Section 3.1 of RFC 2818. For SIP-TLS connections, an additional check is performed to validate the SIP domain identified in the certificate, as per RFC 5922. The parameter is configured through the `46xxsettings.txt` file.

Table continues…

Parameter name	Default value	Description
TPSLIST	Null	Specifies a list of URI authority components (optionally, including scheme and path components) to be trusted. A URI received in a push request is only used to obtain push content, if it matches one of these values. The list can contain up to 255 characters. Values are separated by commas without any intervening spaces. If the value of TPSLIST is null, push is disabled.
TRUSTCERTS		Specifies a list of names of files that contain copies of CA certificates (in PEM format) that are downloaded, saved in non-volatile memory, and used by the telephone to authenticate received identity certificates
U		
USBPOWER	2	Controls USB power when power is provided to the USB interface. Value operation: • 0: Turn off USB power regardless of power source. • 1: Turn on USB power only if Aux powered. • 2: Turn on USB power regardless of power source. • 3: Turn on USB power if Aux powered or PoE Class 3 power.
USER_STORE_URI		Specifies the URI path of IP Office for storing user data. ✱ **Note:** If the value of this parameter is set to null, then the addition, deletion, and modification of **Contacts** is disabled.
USE_QUAD_ZEROES_FOR_HOLD		Specifies how Hold will be signaled in SDP. Value Operation: • 1: “a=directional attributes" will be used • 0: "c=0.0.0.0" will be used
UUIDISPLAYTIME	10	Specifies the duration, in seconds, that the **UUI Information** screen is be displayed.

Table continues…

Parameter name	Default value	Description
		Valid values are 5 through 60.
V		
VLANSEPMODE	1	Specifies whether full VLAN separation will be enabled by the built-in Ethernet switch while the telephone is tagging frames with a non-zero VLAN ID. PHY2PRIO is not supported when VLANSEPMODE is 1. Value operation: • 0: Disabled • 1: Enabled **Note:** This parameter is configured through the settings file.
VLANTEST	60	Specifies the number of seconds that the phone waits prior to failing back to a different VLAN ID if no response is received from the DHCP server. Valid values are 0 through 999. A value of zero means that DHCP tries with a non-zero VLAN ID forever. **Note:** This parameter is configured through: • Settings file • A name equal to value pair in DHCPACK message
VOLUME_UPDATE_DELAY	2	Specifies the minimum interval, in seconds, between backups of the volume levels to PPM service when the phone is registered to Avaya Aura® Session Manager. If there is no change to volume levels, there will be no backup to PPM service. Valid values are 2 through 900. The default value is 2.
W		
WAIT_FOR_INVITE_RESPONSE_TIMEOUT	60	Specifies the maximum number of seconds that the phone waits for another response after receiving a SIP 100 Trying response. Valid values are 30 through 180.

Table continues…

Parameter name	Default value	Description
WAIT_FOR_REGISTRATION_TIMER	32	Specifies the number of seconds that the phone waits for a response to a REGISTER request. If no response message is received within this time, registration will be retried based on the value of RECOVERYREGISTERWAIT. Valid values are 4 through 3600.
WAIT_FOR_UNREGISTRATION_TIMER	32	Specifies the number of seconds the phone waits before assuming that an un-registration request is complete. Un-registration includes termination of registration and all active dialogs. Valid values are 4 through 3600.
WARNING_FILE	Null	Specifies the file name or URL for a custom single-channel WAV file coded in ITU-T G.711 u-law or A-law PCM with 8-bit samples at 8kHz to be used as a call recording warning instead of the built-in English warning. The value can contain 0 to 255 characters.
WEB_ADMIN_PASSWORD	27238	Specifies the password to access the phone through a web browser as an administrator. The value set from the web server interface has a higher priority than that of the Settings file. If the Web admin password is changed using the web server, then the web admin password set through settings file is not used until either the web admin password is set to default through the phone admin menu or the phone is reset to default. Valid values are from 8 to 31 alphanumeric characters including upper, lower and special characters.
WEB_HTTP_PORT	80	Specifies the port on which the Web Server running on the phone will be accessed using HTTP. Valid values are 0, 80, 1024 to 65535.
WEB_HTTPS_PORT	443	Specifies the port on which the Web Server running on the phone will be accessed using HTTPS. Valid values are 443, 1024 to 65535.
WBCSTAT	1	Specifies whether a wideband codec indication is displayed when a wideband codec is used.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Disabled • 1: Enabled
WEBLMSRVR	Null	Sets the IP address or Fully-Qualified Domain Name (FQDN) of the licensing server. Valid values are zero or more IP addresses in dotted-decimal or DNS format, separated by commas without intervening spaces, to a maximum of 255 ASCII characters.

Index