Installing and Administering Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP

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Note

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RSS Standards Statement

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

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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.

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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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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.

When using IEEE 802.11a wireless LAN, this product is restricted to indoor use, due to its operation in the 5.15 to 5.25GHz frequency range. The FCC requires this product to be used indoors for the frequency range of 5.15 to 5.25GHz to reduce the potential for harmful interference to co channel mobile satellite systems. High-power radar is allocated as the primary user of the 5.25 to 5.35GHz and 5.65 to 5.85GHz bands. These radar stations can cause interference with and/or damage to this device.

Class B Part 15 Statement

For product available in the USA/Canada market, only channel 1~11 can be operated. Selection of other channels is not possible.

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.

EU Countries

This device complies with the essential requirements and other relevant provisions of Directive 1999/5/EC. A copy of the Declaration may be obtained from http://support.avaya.com or Avaya Inc., 4655 Great America Parkway, Santa Clara, CA 95054–1233 USA.

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- Use only the Avaya approved Limited Power Source power supplies specified for this product.
- Ensure that you:
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Chapter 1: Introduction

Purpose

This document contains information about how to install and deploy 9600 Series IP Deskphones. It provides administration information for only the following 9600 Series IP Deskphones models:

- 9601
- 9608
- 9608G
- 9611G
- 9621G
- 9641G
- 9641GS

This document is intended for people who install and maintain the 9600 Series IP Deskphones. For example, administrators and service engineers.

Change history

Date	Summary of changes
03–04–2017	Merged documentation for installing and administering procedures. Updated for SIP 7.1 release.

Chapter 2: 9600 Series IP Deskphones overview

9600 Series IP Deskphones overview

Avaya 9600 Series IP Deskphones is a set of desk handset devices that you can use for unified communication. The series leverages the enterprise IP network and eliminates the need for a separate voice network. It works with the Avaya Aura® environment to provide a flexible architecture that works with your investments and accommodates growth as your business needs change.

The deskphones offer high audio quality and low power requirements and the flexibility to customize. With the high-performance models of this series that can operate in both the H.323 and the Session Initiated Protocol (SIP) environments, you can use the phones to:

- Make conference calls more efficient and enhance customer interactions.
- Gain access to information quickly through easy-to-read and high-resolution displays.
- Speed up completion of common telephony tasks by using prompts on touch screens.
- Improve productivity with context-sensitive graphical interfaces that enhance call control and call management.
- 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 the sleep mode, that lowers energy consumption dramatically.

9600 Series IP Deskphones models

Deskphone model	Description
9601	The 9601 deskphone is SIP-only phone that provides a four-row monochrome display and two lines with dual red and green LEDs.

Table continues…

Deskphone model	Description
	The phone has a built in 10/100 Ethernet switch with a port for your personal computer or a laptop.
9608/9608G	You can use up to eight lines for the deskphone. The deskphone supports a traditional user interface and a graphical monochrome display. The 9608 has a built in 10/100 Ethernet switch, and the 9608G has an integrated Gigabit.
9611G	The 9611G has a traditional user interface and a graphical color display. You can use up to eight lines with the 9611G deskphone. The 9611G deskphone has an integrated Gigabit and a USB interface. The deskphone has a graphical color display with a white backlight.
9621G	The 9621G IP deskphone provides gigabit capability and touch screen functionality. Customers with a need for gigabit connectivity to the desktop prefer the 9621G deskphone.
9641G/9641GS	The 9641G/9641GS deskphone provides advanced capabilities with a color touch screen, wideband speaker, USB interface, Bluetooth® enabled headset support, and gigabit connectivity to the desktop. Customers who require gigabit capability for the desktop and the option to add more advanced capabilities prefer the 9641G/9641GS deskphone.

New in this release

Security enhancements

- Supports OpenSSL FIPS 140-2 certified cryptographic algorithms.
- Supports IPv6.
- Supports OpenSSL FIPS 140-2 certified cryptographic algorithms
- Supports Department of Defense solution deployment with Joint Inter-operability Test Command (JITC) compliance.
- Supports display of SSH fingerprint in the Administration menu.
- Displays version of OpenSSH and OpenSSL in the Administration menu.
- Supports SHA2 hash algorithm and strong encryption (256 bit symmetric and RSA 2048 and 4096 bit asymmetric keys) in TLS mode.
- Supports SRTP/SRTCP and TLS v1.2.
- Supports Enhanced Avaya Services Login (EASG) feature where an administrator can escalate to root privileges using an authentication file.

IPv6 and related features

- Supports ICMPv6, PPMv6, and DHCPv6.
- SIP registration over IPv4 and IPv6.

- Supports call features and MDA devices with ANAT and non-ANAT phones.
- Supports early media, delayed media, and media reshuffling with ANAT.

AS-SIP features

- Supports Multiple Level Precedence and Preemption, DSCP, and Blind Transfer as part of the AS-SIP feature.

Chapter 3: Initial setup and connectivity

Prerequisites

Check the prerequisites to ensure that you have the required software and hardware before you install the 9600 Series IP Deskphones.

Hardware requirements

Ensure that the LAN:

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

Software requirements

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
- Avaya Aura® Presence Services 6.2.4 or later
- Avaya Aura® Session Border Controller 7.0 or later.
- IP Office 10.0.3 or later
- A DHCP server for providing dynamic IP addresses to the 9600 Series IP Deskphones.
- 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.

Installation checklist

Use the following checklist to see the tasks that you must perform to setup and connect the9600 Series IP Deskphones.

No.	Task	Reference	✔
1.	Check the prerequisites.	See *Prerequisites* topic for more information.	
2.	Collect the configuration data for the Deskphone, File Server, Network Server/ Switch (LLDP), Avaya Aura® System Manager, and DHCP Server.	See *Configuration through DHCP*, *File Server configuration*, and *Configuration through LLDP* topics for more information.	
3.	Configure the File Server, DHCP, and Network Server/Switch (LLDP).	See *Configuration through DHCP*, *File Server configuration*, and *Configuration through LLDP* topics for more information.	
4.	Create user, session, and communication profile on Avaya Aura® System Manager	See*Administering Avaya Aura® System Manager* guide for more information.	
5.	Download and install Avaya Aura® System Manager certificates on the phone.	See*Administering Avaya Aura® System Manager* guide for more information.	
6.	Unpack and assemble the deskphone components.		
7.	Connect the 9600 Series IP Deskphones to the power source and network.	See *Plugging in the deskphone* for more information.	

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	—	✔	✔	✔	✔

Table continues…

Method	Can administer				
Avaya Aura® System Manager and IP Office	—	✔	—	✔	—
Administration menu on the phone	✔	✔	—	—	✔

Precedence of administration methods

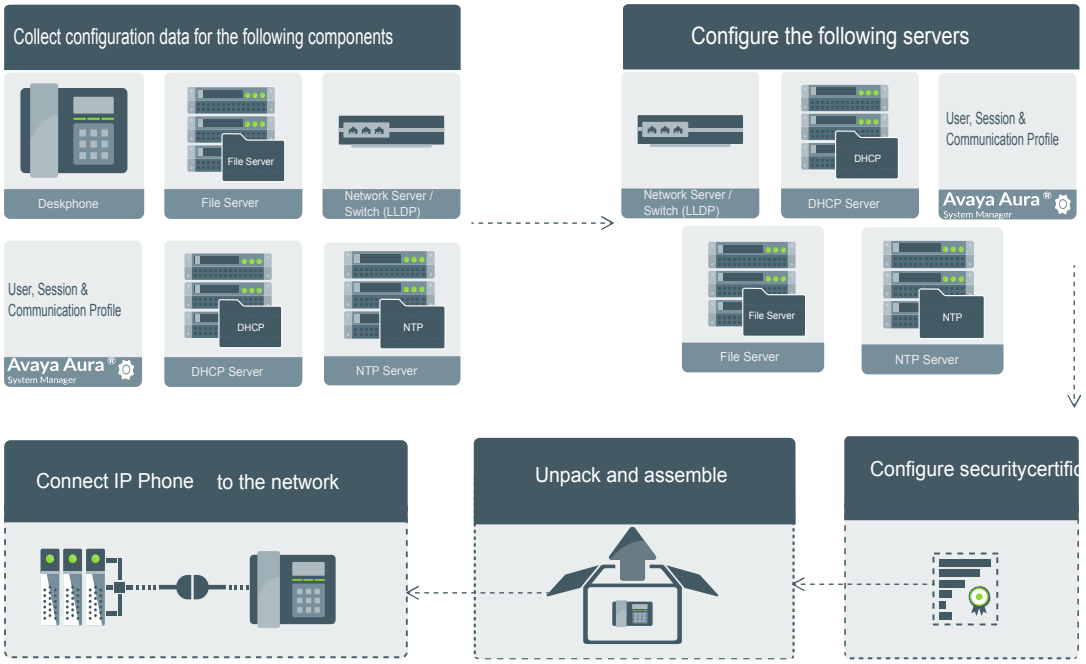
Most of the parameters are configured through multiple methods. If you configure a parameter through more than one method, the device applies the settings of the method that has a higher precedence. The following list shows the precedence of the methods in the highest to lowest order:

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. Avaya Aura® System Manager and IP Office.
3. `46xxsettings.txt` file
4. DHCP.
5. 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.

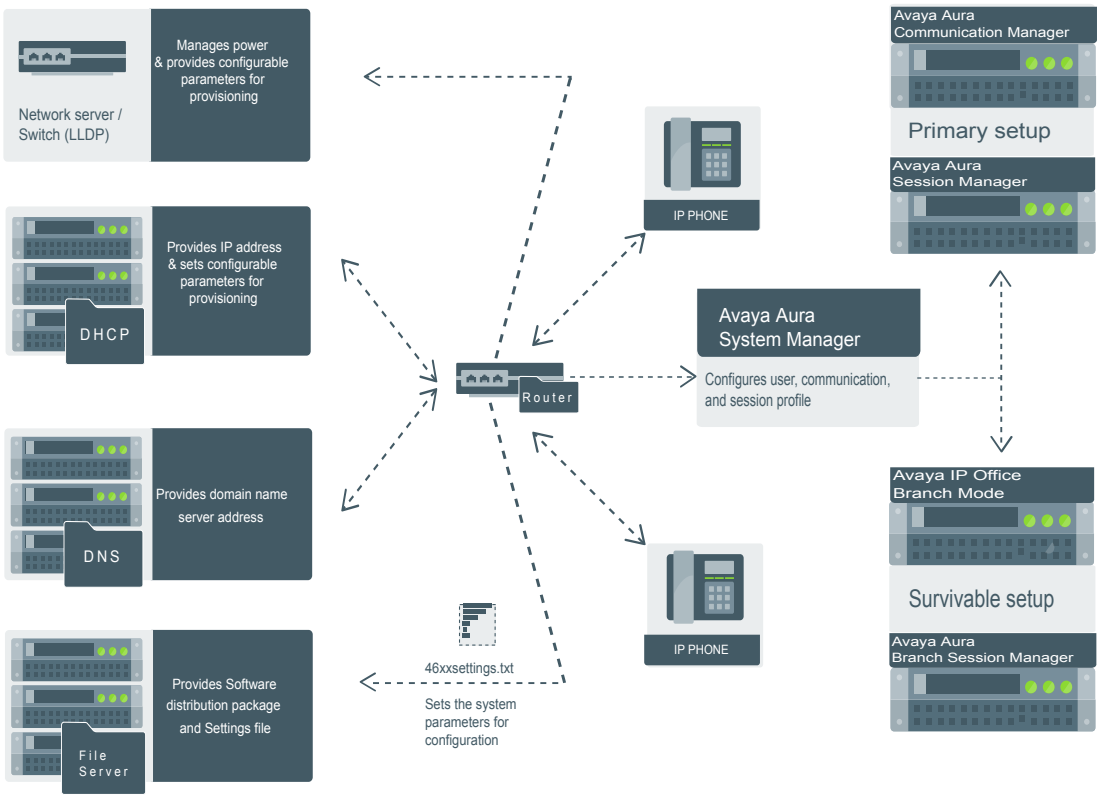
Note:

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

Deployment process



Setup for IP deskphone



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

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 9600 Series IP Deskphones 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 9600 Series IP Deskphones 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 will 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.

Table continues…

Category	TLV Name (Type)	TLV Info String (Value)
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).
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).

Table continues…

System parameter name	TLV name	Impact
		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. • 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).

Table continues…

System parameter name	TLV name	Impact
		• 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	TLSSRVR and HTTPSRVR will be set to the IP addresses in this TLV value.
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.
POE_CONS_SUPPORT	Proprietary - PoE Conservation Level Request TLV	If the value of POE_CONS_SUPPORT is 1, POE_CONS_MODE is set to the level requested in the TLV.

Configuration through DHCP

The 9600 Series IP Deskphones 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 9600 Series IP Deskphones.
- 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.
FILE_SERVER_URL	Specifies the list of URL for downloading image and configuration files. This parameter has higher precedence over HTTPSRVR, HTTPPORT, HTTPDIR, TLSSRVR, TLSDIR, and TLSPORT.
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.

Table continues…

Parameter	Description
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.
PROCSTAT	Controls whether local (Admin menu) procedures can be used to configure the phone.
REUSETIME	Time in seconds for IP address reuse timeout, in seconds. The default is 60 seconds.
SIG	The signaling protocol download flag that indicates the protocol applied as follows: • 0 for Default • 1 for H.323 • 2 for SIP
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.
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 - 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.

Table continues…

Option	Description
	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. • 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 • 26 for interface MTU • 42 for NTP servers
Option 57	Specifies the maximum DHCP message size. Set the value to 1500. Set the value to 1000.

Table continues…

Option	Description
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: • FILE_SERVER_URL • 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
10	L2QVLAN
11	PHY1STAT
12	PHY2STAT
13	PROCSTAT
14	SIG
15	SIP_CONTROLLER_LIST

Extending use of DHCP lease

9600 Series IP Deskphones 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
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

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

Multi Device Access operation in dual-stack mode

When 9600 Series IP Deskphones are 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
- Avaya Aura® System Manager
- Settings File
- DHCP
- LLDP

If you log in with your extension on MDA2 during a call and the signaling mode address 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]` 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

Configuring the File Server

About this task

Use this procedure to configure file server. Examples of File Server are as follows:

- Apache
- Internet Information Services (IIS)
- Avaya Utility Server

Procedure

1. Install the HTTP or HTTPS server software according to the vendor instructions.

2. Download and save the software distribution package and the settings file at the appropriate location on the server.
3. Unzip the distribution package and save the extracted files at an appropriate location on the server.
4. Open and modify the settings file to provision the required device configuration parameters.

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 deskphone 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 deskphone 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 deskphone will continue to interpret the original file.

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

Table continues…

Parameter	Retained
TEAM_BUTTON_RING_T YPE	Y
TPSLIST	N
VLANTEST	Y
WMLHOME	N
WMLPORT	N
WMLPROXY	N

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 9600 Series IP Deskphones are packaged together in a ZIP format. You can download the package from the Avaya support website.

SIP software distribution package contain:

- One or more software files;
- One upgrade file (96x1Supgrade.txt);
- All of the display text language files;
- 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 deskphones based on the value of the TRUSTCERTS parameter. You must also include the System Manager route certificate in the TRUSTCERTS parameter for IM to work.
- File named release.xml that is used by the Avaya Software Update Manager application.

Note:

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

Two configuration files are important to understand. They are:

- The upgrade file, 96x1Supgrade.txt, that tells the deskphone whether the deskphone needs to upgrade software. The deskphones 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 deskphones for your enterprise.

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 `9600 Series IP Deskphones`.
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.

Powering the phone

The 9608, 9608G, 9611G, 9621G, 9641G and 9641GS phones are powered with the Global Single Port PoE Injector (GSPPOE-xx) and power module (DC power jack). The telephone power module is available and sold separately with the Comcode 700511266. Additionally, all phones support IEEE 802.3af-standard LAN-based power.

The power requirements and its class changes when you add peripherals such as button modules and USB devices. The power consumption of the drive varies and when the drive attempts to register with the phone, the phone determines if its current power class setting is adequate. Toggle the IEEE switch between L and H for providing adequate power. An auxiliary power supply is required when the power requirements is greater. Refer to the table below for different power requirements:

The impact of additional devices on power requirements over Ethernet Power Class

Phone Model	Default PoE (Class "L" on IEEE switch)	One BM12 (IEEE switch setting)	Two BM12s (IEEE switch setting)	Three BM12s (IEEE switch setting)	One SBM24 (IEEE switch setting)	Two SBM24s (IEEE switch setting)	Three SBM24s (IEEE switch setting)
9608	Class 1	L	H	H	L	H	H
9608G	Class 1	H	H	H	H	H	H
9611G	Class 1	H	H	H	H	H	H
9621G	Class 2	Not applicable; the 9621G does not support button modules or USB devices.					
9641G/ 9641GS	Class 2	H	H	H	H	H	H

Note:

- 9621G is a PoE Class 2 device with a 10/100/1000 switch and does not have an IEEE power switch.
- The 9601 and 9621G do not support a button module, a USB device, or a Dual Headset Adapter.
- The phone model 9608 does not support USB devices.
- If you set the IEEE switch on the back of the phone to H, the phone registers as a Class 3 device, even if the actual power usage is applicable to Class 1 or 2.
- Use the icons at the back of the phone for locating the correct jacks.
- The system parameter USBPOWER sets the power class for the USB interface.

To learn how to connect cords to the jacks on the phones:

Phone model	Figure:
9608, 9608G, 9611G	Connection jacks on a 9608, 9608G, or 9611G phone **Note:** The 9601 phone has an external power adapter. Use the connections that apply.
9621G, 9641G, 9641GS	Connection jacks on a 9621G, 9641G, or 9641GS phone

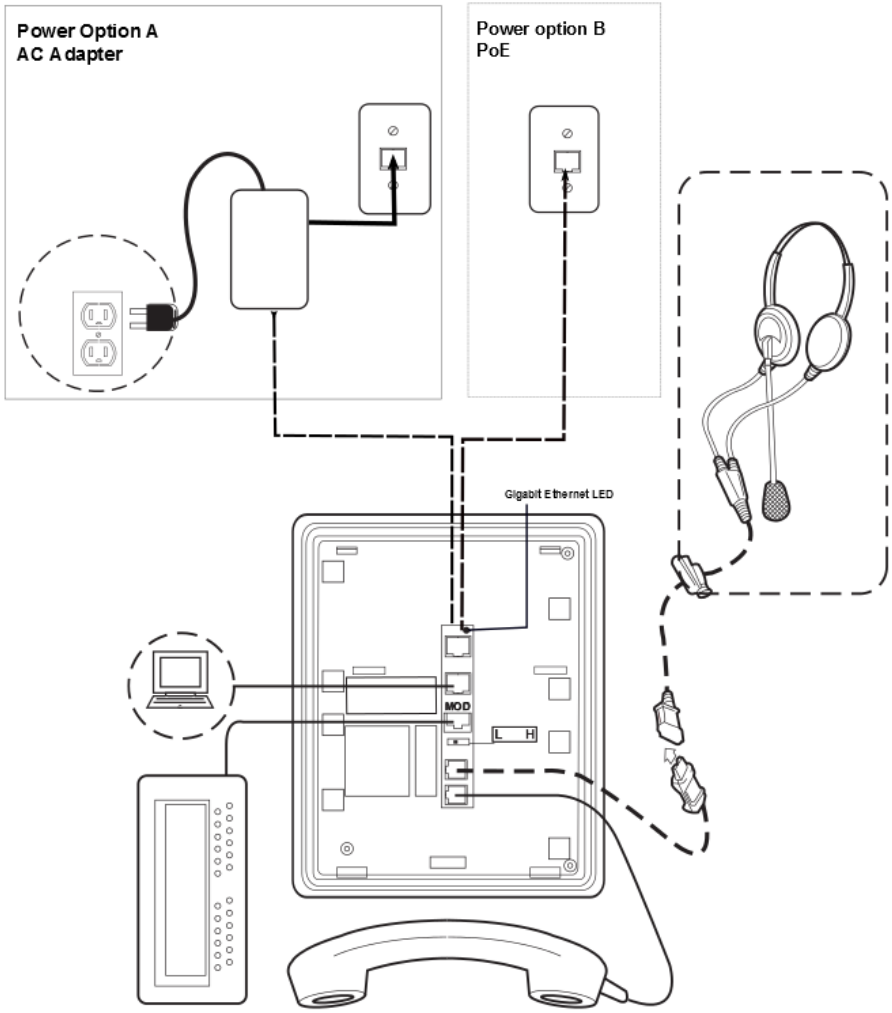
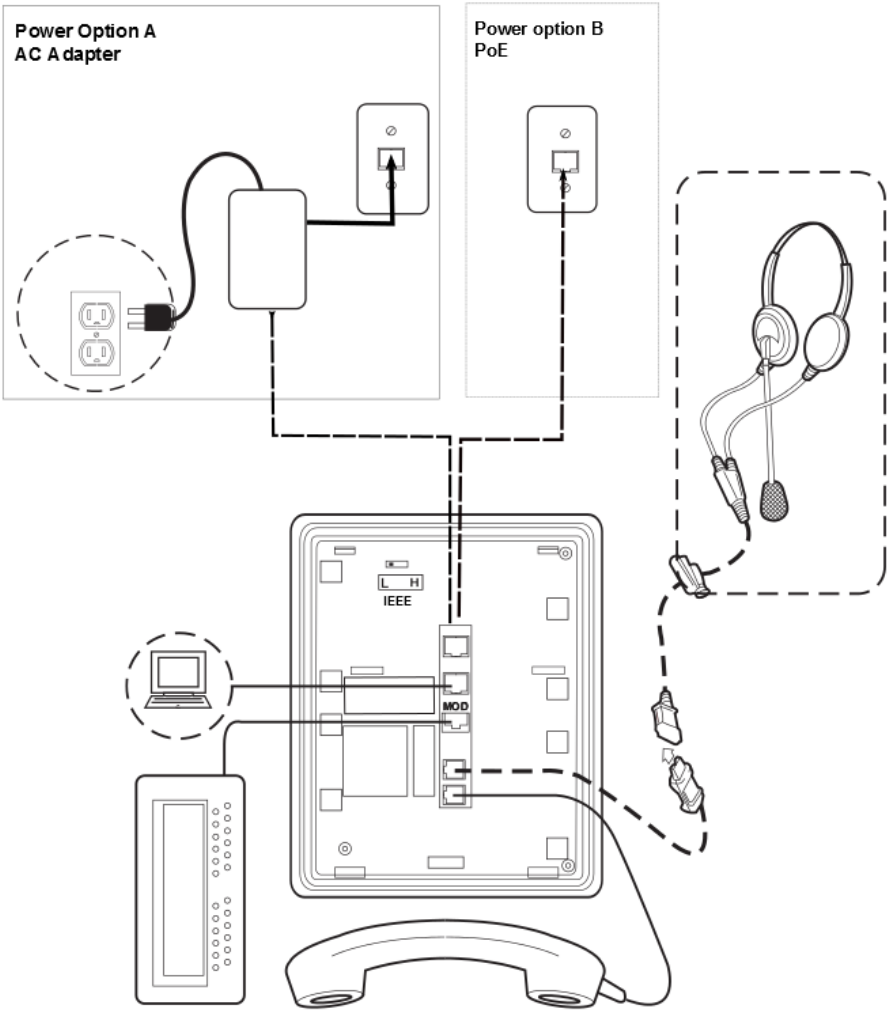


Figure 1: Connection jacks on a 9608, 9608G, or 9611G deskphone

Note:

The Gigabit Ethernet LED indicator is applicable to the 9608G and 9641GS IP phones. This indicator lights up steady green when a link of any speed is established, blinks with any network activity, and turns off upon the loss of network connectivity.

Figure 2: Connection jacks on a 9621G, 9641G, or 9641GS deskphone


Power Option A
AC Adapter
Power option B
PoE
L H
IEEE
MOD

Chapter 4: Security configurations

Security overview

9600 Series IP Deskphones provide several updated security features. For example:

- Supports complex login password for protecting account privacy of both user and administrator. Login password can include both special and alphanumeric characters.
- Protects user logs and data.
- Restricts a user to only receive calls and make emergency calls when the phone is locked.

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.
- Control over USB power using system parameters.
- Display of SSH fingerprint 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.

Device lock management parameters

Parameter name	Default value	Description
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. The options are: • 0: Disabled • 1: Enabled
PHONE_LOCK_IDLETIME	0	Specifies the interval of idle time, in minutes, after which the phone will automatically get locked. Valid values are from 0 to 10080.

Table continues…

Parameter name	Default value	Description
		The options are: • 0: Phone will not lock automatically • 1: Phone will lock automatically

User and account management parameters

Phones support the following parameters for managing the Administration menu for local procedures:

Parameter name	Default value	Description
PROCSTAT	0	Specifies whether local or CRAFT procedures can be used to configure the phone. The options are: • 0: Local procedures can be used • 1: Local procedures cannot be used
PROCPSWD	27238	Specifies an authentication code to access Administration menu. The options are: • 27238: Specifies that the authentication code `27238` is set for accessing local or craft administration procedures • 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 the local or craft procedures.
ADMIN_PASSWORD	27238	Specifies an authentication code for accessing the local (craft) procedures screen. This parameter can be set in System Manager and File Server. When the parameter ADMIN_PASSWORD is not set, then the parameter PROCPSWD is used. PROCPSWD supports only numeric values. ADMIN_PASSWORD supports both alphanumeric and special characters. Hence, for enhanced security, use ADMIN_PASSWORD instead of PROCPSWD.

Table continues…

Parameter name	Default value	Description
		You must provide an authentication code for ADMIN_PASSWORD by using a combination of: • Numbers (0–9) • Alphabets in uppercase (A-Z) • Alphabets in lowercase (a-z) • Special characters, except the double quote character (")
ADMIN_LOGIN_ATTEMPT_ALLOWED	10	Specifies the number of failed attempts allowed for accessing the Administration 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 of lockout when a user reaches the maximum attempts limit to access the Administration menu. Valid values are between 5 to 1440 minutes.

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.

USB port power

This feature is controls the power of the USB port. This feature is controlled by a parameter called USBPOWER. The available values for the parameter are:

- 0: Turns off the USB power.
- 1: Turns on the USB power only when it is powered through Aux.
- 2: Turns on the USB power.
- 3: Turns on the USB power when it is powered through Aux or PoE Class 3.

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.

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.

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.

FIPS mode

The Federal Information Processing Standard, or FIPS 140-2, is a computer security standard used by the U.S. government to approve cryptographic modules. OpenSSL libraries include a set of cryptographic algorithms compliant with FIPS 140-2, which can be invoked when the library is

initiated in FIPS mode. The parameter FIPS_ENABLED controls the usage of OpenSSL FIPS certified cryptographic modules. You can set the parameter through the `46xxsettings.txt` file or DHCP option 242. The description of the parameter is as follows:

Parameter name	Default value	Description
FIPS_ENABLED	0	Specifies whether only FIPS-approved cryptographic algorithms will be supported. The options are: • 0: No restriction on using cryptographic algorithms that are not FIPS-approved. • 1: Use only FIPS-approved cryptographic algorithms using embedded FIPS 140-2 validated cryptographic module.

Ensure that the value of the parameter CONFIG_SERVER_SECURE_MODE is set to 1 when the phone is in FIPS mode.

When you enable FIPS mode, you must disable the following features on the phone:

- SSH Server.
- SCEP certificate enrollment: When a phone runs in FIPS mode, identity certificate enrollment through SCEP is disabled by the software. If identity certificate is generated before FIPS_ENABLED is set to 1, it can still use the existing identity certificate after phone reboot. However, you must not use identity certificates generated using SCEP when FIPS_ENABLED is set to 0 when the phone is configured to work in FIPS mode. The most secure way to install identity certificate is to clear any installed identity certificate and install PKCS#12 file after configuring the phone to FIPS mode. Thereafter, FIPS 140-2 approved cryptographic algorithms can be used to decrypt PKCS#12 file.
- SLA Mon.
- 802.1x with EAP-MD5 or EAP-PEAP authentication. EAP-TLS is allowed.
- WML Browser.
- Push.
- HTTPSRVR. You must use TLSSRVR for file downloading.
- HTTP in OCSP_URI or Authority Information Access (AIA) of a certificate. Ensure that the URI in OCSP_URI or AIA of a certificate is HTTPS.
- Microsoft™ Exchange

When you enable FIPS mode, the phone reboots and runs the OpenSSL FIPS self-test. When the test is completed successfully, the phone displays the message `FIPS mode activated, restarting…`. After reboot, FIPS mode is in effect. If the FIPS-mode self-test fails, the phone displays the message `FIPS self-test failure`. In this case, the phone also displays two options:

- **Program**: The phone prompts for a CRAFT password. After you enter the CRAFT password, the phone boots up in non-FIPS mode.
- **Reboot**: The phone reboots.

Note:

All the logs are stored in SYSLOG. These logs might be referred to for the troubleshooting purpose.

Chapter 5: Phone administration

Introduction

During installation or after you have successfully installed a 9600 Series IP Deskphones, you might be instructed to administer one of the manual procedures described in this chapter. These local administrative procedures are also referred to as Craft Procedures.

Note:

You can modify the settings file to set parameters for deskphones that download their upgrade script and application files from the same HTTP server. Only trained installers or technicians must perform local (craft) procedures. Perform these procedures only if instructed to do so by the system or LAN administrator.

Static administration of these options causes upgrades to work differently than if they are administered dynamically. Values assigned to options in static administration are not changed by upgrade scripts. These values remain stored in the deskphone until you use the local administrative procedures CLEAR or RESET.

Use these option-setting procedures only with static addressing and, as always, only if instructed by the system or LAN administrator. Do not use these option-setting procedures if you are using DHCP.

About local administrative procedures

Craft procedures allow you to customize the IP phone installation for your specific operating environment. This section provides a description of each local administrative option covered in this guide.

Note:

By default, a user can view but not change most of the parameters associated with Craft procedures.

Options	Purpose
802.1X	Specifies the 802.1X operational mode.
ADDR	Specifies the network address information.
AGC	Enables or disables Automatic Gain Control.

Table continues…

Options	Purpose
CALIBRATION	Calibrates the touchscreen (9621G and 9641G models only).
CLEAR	Clears all values to factory defaults.
DEBUG	Enables or disables Debug Mode.
GROUP	Sets the Group Identifier.
HANDSET EQ	Sets the handset equalization.
INT	Specifies the network interface control.
LOG	Enables or disables the event logging.
LOGOUT	Logs off the phone.
RESET VALUES	Resets system initialization values to default.
RESTART PHONE	Restarts the phone.
SIG	Sets the signaling protocol download flag.
SIP	Configures the SIP call settings.
SNTP	Configures the time server settings.
SSON	Sets the Site-Specific Option Number.
VIEW	Shows current parameter values and file names.

Accessing the Administration menu

The Local or the CRAFT procedures can only be invoked if the value of the PROCSTAT parameter in the `46xxsettings.txt` file is set to **0**. Setting the PROCSTAT parameter to **0** provides full access to the local procedures. You can access the Administration menu during phone startup and during normal phone operation.

Note:

You cannot answer a call when the phone displays the Administration password screen.

For all non-touchscreen phones, use the navigation arrows to scroll to and highlight the local procedure you want, then press **Select** or **OK**. Or scroll to the procedure you want and press the corresponding line button. For touchscreen phones, scroll to the local procedure you want if it not already displayed then touch the line on which the local procedure you want appears.

Accessing the Administration menu during phone startup

About this task

You can access the Administration menu during phone startup using the following steps.

Before you begin

The default password to gain access to the local procedures menu is set in PROCPSWD or ADMIN_PASSWORD parameter. The factory-set default password is **27238**. You must not change the default value at the time of initial installation.

Procedure

1. Press **Admin** .
2. Enter the administrator password.
3. Press the **Enter** softkey.

Accessing the Administration menu after phone startup

Procedure

1. Press **Avaya Menu** for button phones or **Avaya Home** for touchscreen phones.
   - For touchscreen phones, tap **Settings** icon.
   - For button phones, scroll down to **Administration** option.
2. Do one of the following:
   - For touchscreen phones, scroll down and tap **Administration**.
   - For button phones, press **Select**.
3. Enter the administrator password on the Enter access code screen.
4. Select **Enter**.

The Avaya Menu administration file

The Avaya Menu administration file contains parameters to customize Avaya Menu and to view Web links on Avaya Menu. You can administer up to eight Web links.

To customize Avaya Menu, you need to:

- Create the Avaya Menu Administration file named AvayaMenuAdmin.txt and configure appropriate parameters.
- Set the AMADMIN parameter in the settings file to point to the HTTP server location where you save the AvayaMenuAdmin.txt file.

The AvayaMenuAdmin.txt file contains the following parameters:

Parameter	Description
AMTYPExx	Specifies the type of Avaya Menu option that you want to configure. For the 9601, 9608, 9608G, and 9611G deskphones, you can assign the following values to this parameter: • 1 for Web link • 2 for the Options & Settings option • 3 for the Network Information option • 4 for the About Avaya one-X option • 5 for the Log Out option For the 9621G and 9641G deskphones, you can assign the following values to this parameter: • 1 for Web link • 2 for the Options & Settings option • 3 for the Network Information option • 4 for the Light Off option • 5 for the Touch Screen Cleaning option • 6 for the About Avaya one-X option • 7 for the Log Out option where, xx is a two-digit integer from 01 to 12. If AMTYPE*xx* is 1 then you must also define the following parameters: • AMLBLxx • AMDATAxx where, xx must be the same for each of the three parameters. If AMTYPExx is 2, 3, or 4, the system ignores any value that you specify for AMLBLxx or AMDATAxx.
AMLBLxx	Specifies the label that the deskphone displays for the Web link. The label must not exceed 16 UTF-16 characters.
AMDATAxx	Specifies the URI for the Web link. The URI must not exceed 255 ASCII characters

You can specify 12 Web links, but the deskphone displays only the first eight Web links. If you have also configured the browser, the deskphone displays eight Web links that includes the browser

The deskphone displays any built-in application for which you set the respective parameter to display the application, even if you do not configure that application in the AvayaMenuAdmin.txt file.

Example

```
##############################################################
## AVAYA MENU CONFIGURATION FILE TEMPLATE ##
##############################################################
## This file is to be used as a template for configuring Avaya Main
## Menu. See the LAN Administrators Guide nd the Avaya one-X™ Deskphone
## Edition for 9600 Series and the Avaya one-X™ Deskphone Edition for
## 9600 Series and the Avaya one-X™ Deskphone Edition for 9600 Series
## IP Telephones Administrator Guide for details.
## Both are available on support.avaya.com
##############################################################

## AMLBLxx=Lable up to 16 unicode character
##
## AMTYPExx=Type 1=WML-Application, 2=local Phone Settings,
## 3=local LogOff Application, 4=local About Avaya Screen
##
## AMDATAxx        URI of up to 255 ASCII-characters
## e.g. http://yy.yy.yy.yy/*.wml
##
## The tags AMLBLxx and AMDATAxx are only used if AMTYPExx = 1
##
## Multiple definitions of local applications (Type 2..4)
## will be supressed. The last tag is valid.
##
## xx describes the sequence in A-Menu and is valid
## from 01 to 12
##

##AMTYPE01=
##AMLBL01=
##AMDATA01=

##AMTYPE02=
##AMLBL02=
##AMDATA02=

##AMTYPE03=
##AMLBL03=
##AMDATA03=

##AMTYPE04=
##AMLBL04=
##AMDATA04=

##AMTYPE05=
##AMLBL05=
##AMDATA05=

##AMTYPE06=
##AMLBL06=
##AMDATA06=

##AMTYPE07=
##AMLBL07=
##AMDATA07=

##AMTYPE08=
##AMLBL08=
##AMDATA08=
```

```
##AMTYPE09=
##AMLBL09=
##AMDATA09=

##AMTYPE10=
##AMLBL10=
##AMDATA10=

##AMTYPE11=
##AMLBL11=
##AMDATA11=

##AMTYPE12=
##AMLBL12=
##AMDATA12=
```

Administering settings at the deskphone

The document Installing and Maintaining Avaya 9601/9608/9608G/9611G/9621G/9641G IP Deskphones SIP explains in details how to use the local administrative procedures on the deskphone for administration. The local procedures that you can administer on 9600 Series IP Deskphones are:

- 802.1X - To set the 802.1X operational mode.
- ADDR - To set the static addresses.
- AGC - To enable or disable Automatic Gain Control.
- CALIBRATION - Applicable to 9621G and 9641G deskphones. To calibrate the touch screen.
- CLEAR - To remove all administered values, user-specified data, option settings, etc. and return a deskphone to its initial "out of the box" default values.
- DEBUG - To enable or disable debug mode for the button module serial port.
- GROUP - To set the group identifier on a per-deskphone basis.
- HANDSET EQ - To set the handset equalization settings of the deskphone.
- INT - To locally enable or disable the secondary Ethernet hub.
- LOG - To enable or disable event logging.
- LOGOUT - To logout the user from the deskphone.
- RESET VALUES - To reset the deskphone to default values including the registration extension and password, any values administered through local procedures, and values previously downloaded using DHCP or a settings file.
- RESTART PHONE - To restart the deskphone in response to an error condition, including the option to reset parameter values.
- SIG **-** To change the default signaling value from SIP to H.323 or vice versa.
- SIP - To configure SIP call settings.

Warning:

The SIP call settings entered through the Administration menu take precedence over other sources for this data, for example- `46xxsettings.txt` file, or PPM. The only way to override these settings is to go into the Administration menu and remove the settings or perform a "Clear" of the deskphone from the Administration menu.

- SNTP - To configure the time server settings.
- SSON - To set the site-specific option number.
- VIEW - To review the system parameters for the deskphone to verify current values and file versions.

Applications and features provisioning

You can configure the following parameters to enable or disable certain applications and general phone features. The following table lists the features and applications and their corresponding parameters that enable

Application or feature	Parameter name	Description
History application	ENABLE_CALL_LOG	If enabled, users can access the list of unanswered and answered calls. If disabled, the History application is not displayed to the user and calls are not logged.
Redial	ENABLE_REDIAL	If enabled, users can redial one to three previously called numbers. If disabled, redialing is not available to the end user.
Redial list	ENABLE_REDIAL_LIST	If enabled, users can select a number to redial from a list. If disabled, only the previously-dialed number can be redialed.
Contacts application	ENABLE_CONTACTS	If enabled, users can access a list of numbers and make calls by selecting a contact name or number. If disabled, the deskphone does not display the Contacts application and users cannot set up or maintain the contact list.
Contacts modification	ENABLE_MODIFY_CONTACTS	If enabled, users can change or update the contact list. If disabled, users cannot change or update the contact list.
Exchange contacts	PROVIDE_EXCHANGE_CONTACTS	If enabled, users can gain access to contacts stored in the Exchange server through the Contact list, by pressing the Exchange contact button. If disabled, users cannot gain access to the Exchange contacts.
Options & Settings menu option	PROVIDE_OPTIONS_SCREEN	If enabled, the deskphone displays the Options & Settings menu option in the Avaya menu or Home Screen. If disabled, the deskphone does not display the Options & Settings menu option. Users cannot change any of the features and ppeonfiasscnted with the Options & Settings menu.
Network Information menu option	PROVIDE_NETWORKINFO_SCREEN	If enabled, the deskphone displays the Network Information menu option in the Avaya menu or Home Screen. If disabled, the deskphone does not display the Network Information menu option.

Table continues…

Application or feature	Parameter name	Description
Logout menu option	PROVIDE_LOGOUT	If enabled, the deskphone displays the Logout menu option in the Avaya menu or Home Screen. If disabled, the deskphone does not display the Logout menu option.
Exchange calendar	PROVIDE_EXCHANGE_CALENDAR	If enabled, users can integrate and gain access to the Exchange calendar on the deskphone. If disabled, users cannot gain access to the Exchange calendar on the deskphone.

Setting the signaling protocol identifier

About this task

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

Note:

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

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. When you select **SIG...** from the Administration procedures screen, the deskphone prompts you to use the Right and Left navigation arrows to select a setting and displays the following text:

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

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

   Note:

   The SIG value "Default" can represent either SIP or H.323 depending on the upgrade file used for the deskphone.
3. To change the setting, press the **Change** softkey until you see the setting you want or use the **Right** or **Left** navigation arrow to cycle through the settings.

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

4. Press **Save** to store the new setting and redisplay the Administration procedures screen.

   The remainder of this procedure depends on the status of the boot and application files.

Configuring SIP settings

About this task

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

Procedure

1. Select **SIP** from the Administration procedures screen.

   The phone displays the following:

   - **SIP Global Settings**
   - **SIP Proxy Server**

2. Press **Select** or **OK** or the corresponding line button to change any of the following SIP global settings:

   - **SIP 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-Aura 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 more than one active controllers.

   - **Failback Policy**: Specifies the fail back policy.

     The two modes are as follows:

     - **Auto**: Active controller automatically recovers after failback.
     - **Administration**: Active controller uses failback policy defined by the administrator.

   - **Proxy Policy**: Specifies whether the settings of SIP proxy servers are taken from the `46xxsettings.txt` file or can be edited by the user.

     The two modes are as follows:

     - **Auto**: The settings are taken from the `46xxsettings.txt` file. The user can only view the settings.

- **Manual**: The user can edit, delete, or create new server properties.

- **Avaya Config Server**: Specifies the IP address of Avaya configuration server, only if PPM is not on the same server as the SIP Proxy server.
- **User ID**: Specifies the user ID of the currently logged in user.
- **Host to Ping:** Checks the host server for response using IP address or DNS.

The SIP call settings entered through the Administration menu take precedence over other sources for this data, for example, `46xxsettings.txt` file or PPM. The only way to override these settings is to remove the settings or use the **CLEAR** option from the Administration menu on phone.

3. Select **SIP Proxy Server** to change SIP proxy server settings.

   The phone displays a list of currently configured servers.

   Caution:

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

4. Do one of the following:
   - To add a new server, select **New**, enter the following values and select **Save**.
     - **SIP Proxy Server**: Specifies the IP address or DNS for Session Manager deployments. The corresponding parameters are SIP_CONTROLLER_LIST for IPv4 and SIP_CONTROLLER_LIST2 for IPv6 or dual mode.
     - **Transport Type**: Specifies the type of transport. The available options are TCP, UDP, or TLS. The corresponding parameter is SIPSIGNAL.
     - **SIP Port**: Specifies the SIP port. If no value is entered, default of 5060 for UDP/TCP or 5061 for TLS is used. If Transport Type is UDP/ TCP, SIP_PORT_ SECURE is used.
   - To edit a SIP proxy server settings, select the server from the list of configured servers and edit fields such as, **SIP Proxy Server**, **Transport Type**, or **SIP Port**. Select **Save**.
   - To delete a SIP proxy server, select **Delete**.

Configuring Time Server settings

About this task

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

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.

2. When you select **SNTP...** from the Administration procedures screen, the deskphone displays the following settings and prompts you to enter the IP Address of the SNTP server:
   - SNTP Server: Specifies the IP address or DNS of the network time server and changes SNTPSRVR or SNTPSRVR_IN_USE parameters.
   - SNTP GMT offset: Specifies the local time difference in hours from Greenwich Mean Time. The corresponding parameter is GMTOFFSET .
   - SNTP Daylight Savings Time Off/On/Auto: Specifies whether the deskphone should recognize Daylight Savings Time (DST) (0=no DST, 1=DST activated as per DSTOFFSET, 2=automatic based on DSTSTART and DSTSTOP values. The corresponding parameter is DAYLIGHT_SAVING_ SETTING_MODE.
3. Press **Save** to store the new setting and returns to the Administration procedures screen.

Setting the date and time on SIP deskphones

9600 Series IP Deskphones need a source of date and time information. This information typically comes from a network time server running the Simple Network Time Protocol (SNTP). The deskphones use several administrative parameters for this functionality. The parameter SNTPSRVR defines the server's IP Address(es). GMTOFFSET defines the offset from Greenwich Mean Time (GMT). DSTSTART and DSTSTOP define the start and end of Daylight Savings Time, respectively. DSTOFFSET defines the Daylight Savings Time offset from Standard Time. Finally, DATETIMEFORMAT defines the format of the date and time display.

About DNS addressing

support DNS addresses, dotted decimal addresses, and colon-hex 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, non-zero, dotted decimal address. Otherwise DNS fails. The text string for the DOMAIN system parameter, Option 15 is appended to the addresses in Option 6 before the phone attempts DNS address resolution. If Option 6 contains a list of DNS addresses, those addresses are queried in the order given if no response is received from previous addresses on the list. As an alternative to administering DNS by DHCP, you can specify the DNS server and or the domain name in the HTTP script file. But first SET the DNSSRVR and DOMAIN values so that you can use those names later in the script.

Note:

Administer Options 6 and 15 with DNS servers and domain names respectively.

Virtual LAN overview

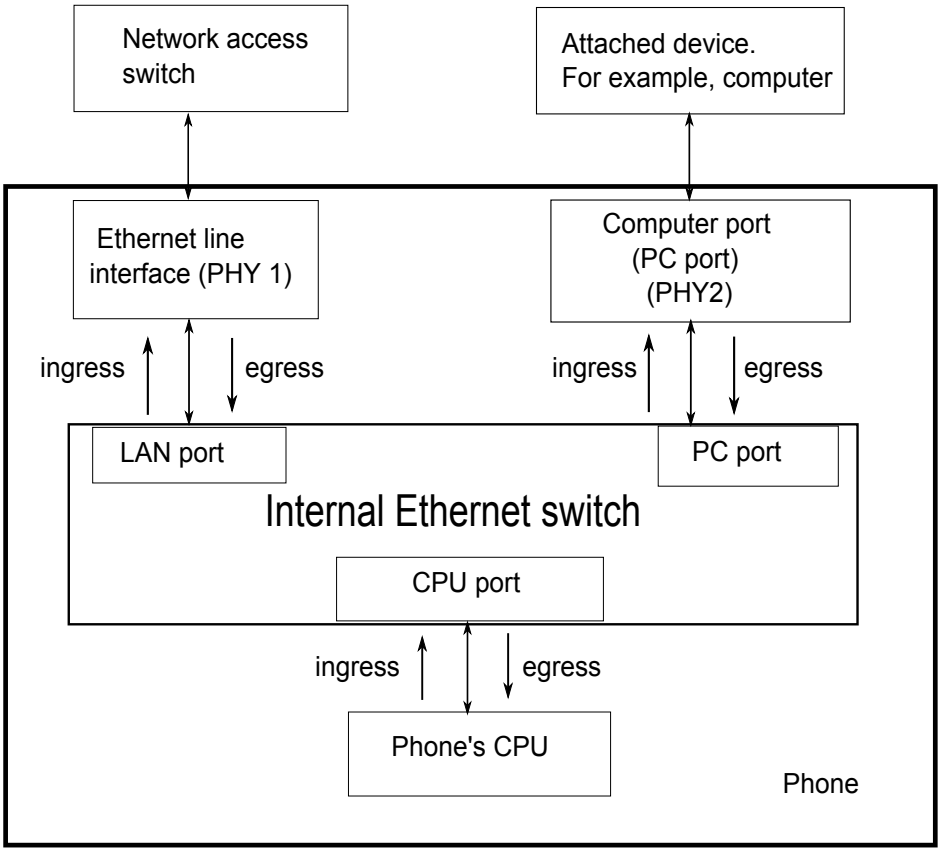
Virtual LANs (VLANs) are used to segregate your network into groups or domains. They can also prioritize the network traffic into each domain. For example, a network can have a Voice VLAN and a Data VLAN. This functionality of grouping devices that have a set of common requirements can simplify network 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 the 802.1Q protocol and the way in which an additional four-byte VLAN tag is inserted at the beginning of an Ethernet frame. This additional VLAN tag describes the VLAN ID to which a particular device belongs and the priority of the VLAN tagged frame. Voice and video traffic get a higher priority in the network because they are subject to degradation caused by network jitter and delay.

VLAN separation

A phone has an internal network switch that can use VLANs to segregate traffic going to 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. Also, you can configure the switch to separate the traffic moving towards the CPU of the phone from the traffic moving towards the PC port.

You can configure the internal switch of the phone by using the `46xxsettings.txt` file, LLDP, or DHCP. However, you must configure the VLAN settings on the internal switch of the phone only through DHCP or LLDP because these protocols are run before and during network initialization. If that is not possible, then use the `46xxsettings.txt` file configuration parameters and start the VLAN 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

VLAN separation modes in 9600 Series IP Deskphones

9600 Series IP Deskphones support three VLAN separation modes:

- No VLAN separation mode: The CPU port of the phone receives untagged and tagged VLAN frames on any VLAN irrespective of whether the phone sends untagged or frames. This traffic is received from the PC port or the LAN port. The CPU filters the frames. To reduce unnecessary traffic to the CPU, you must configure only the necessary VLANs on the external switch port, in particular, voice VLAN and data VLAN. To configure the phone to work in this mode, set VLANSEP=0.
- Partial VLAN separation mode: This is the default mode. In this mode, the CPU port sends tagged packets to voice VLAN only. The CPU port receives tagged packets on voice VLAN or untagged packets from LAN port. PC and LAN port can send any traffic between them. To configure the phone to work in this mode, ensure the following conditions :
  - VLANSEP = 1
  - VLANSEPMODE = 0
  - L2Q = 0 (auto) or 1 (tag)
  - VLANTEST = 0 or timer is less than VLANTEST
  - PHY2VLAN = 0
- Full VLAN separation 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 the CPU port and the PC port. To configure the phone to work in this mode, ensure the following conditions:
  - VLANSEP = 1
  - VLANSEPMODE = 1
  - L2Q = 0 (auto) or 1 (tag)
  - L2QVLAN is not equal to 0
  - PHY2VLAN is not equal to 0
  - L2QVLAN is not equal to PHY2VLAN

The DHCP server on voice VLAN is reachable and the phone receives the IP address on voice VLAN when the phone sends tagged VLAN frames.

Configuring an external switch port

About this task

It is important to restrict the VLAN binding in no VLAN separation mode. This is because the internal phone switch does not filter the frames and the CPU of the phone is subjected to all the traffic going through the phone. In full VLAN separation mode, the internal phone switch filters any tagged VLAN

frames with VLANs other than voice VLAN and data VLAN. However, you must configure only the necessary VLANs on the external switch port.

Procedure

1. Bind VLAN to the voice VLAN, that is L2QVLAN, and the data VLAN, that is PHY2VLAN.
2. Set the default VLAN as the data VLAN.

   This data VLAN is the VLAN assigned by the external switch port to untagged frames received from the phone LAN port.
3. 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.

   When egress voice VLAN frames are sent untagged from the external switch port to the phone LAN port, there is no VLAN separation between the voice VLAN and data VLAN.

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

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.

VLAN parameters

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

Parameter name	Default value	Description
L2Q	0	Specifies whether the VLAN tagging is enabled or disabled. The options are: • 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 the value pair in a DHCPACK message • A SET command in the `46xxsettings.txt` file • DHCP option 43 • LLDP

Table continues…

Parameter name	Default value	Description
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 from 0 to 999. The option is: • 0: The phone continues to attempt a DHCP REQUEST forever. VLANTEST is configured through: • `46xxsettings.txt` file • A name equal to the value pair in the DHCPACK message
VLANSEP	1	Specifies whether VLAN separation is enabled by the built-in Ethernet switch while the phone tags frames with a non-zero VLAN ID. The options are: • 0: Disabled • 1: Enabled if L2Q, L2QVLAN, and PHY2VLAN are set appropriately VLANSEP is configured through the `46xxsettings.txt` file.
VLANSEPMODE	1	Specifies whether full VLAN separation is enabled by the built-in Ethernet switch while the phone tags frames with a non-zero VLAN ID. The options are: • 0: Disabled • 1: Enabled PHY2PRIO is not supported when VLANSEPMODE is 1. VLANSEPMODE is configured through the `46xxsettings.txt` file.
PHY2TAGS	0	Determines whether VLAN tags are stripped on Ethernet frames going out of the computer (PC) port. The options are: • 0: Strips VLAN tags from Ethernet frames leaving the computer (PC) port of the phone.

Table continues…

Parameter name	Default value	Description
		• 1: Does not strip VLAN tags from Ethernet frames leaving the Computer (PC) port of the phone. PHY2TAGS is configured through the `46xxsettings.txt` file.
L2QVLAN	0	Specifies the voice VLAN ID to be used by IP phones. Valid values are from 0 to 4094. L2QVLAN is configured through: • A local admin procedure • A name equal to the value pair in the DHCPACK message • A SET command in the `46xxsettings.txt` 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 from 0 to 4094. PHY2VLAN is configured through: • A SET command in the `46xxsettings.txt` 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, for example, RTP, RTCP, SRTP, and SRTCP. All other frames, except those specified by the L2QSIG parameter, are set to priority 0. Valid values are from 0 to 7. L2QAUD is configured through: • A SET command in the `46xxsettings.txt` file • LLDP

Table continues…

Parameter name	Default value	Description
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, for example, SIP. All other frames, except those specified by the L2QAUD parameter, are set to priority 0. Valid values are from 0 to 7. L2QSIG is configured through: • A SET command in the `46xxsettings.txt` file • LLDP
PHY2PRIO	0	Specifies the layer 2 priority value to be used for frames received on the secondary Ethernet interface when VLAN separation is enabled. Valid values are from 0 to 7. PHY2PRIO is configured through the `46xxsettings.txt` file.

IEEE 802.1X overview

The IEEE 802.1X standard provides specifications on layer 2 network secure access. Phones implement 802.1X Supplicant in unicast and multicast mode, and pass-through mode and proxy logoff feature for the attached device on the PC port.

The following configuration parameters are applicable to this feature:

Parameter name	Default value	Description
DOT1XSTAT	0	Specifies the 802.1X Supplicant operating mode. The options are: • 0: Supplicant disabled. • 1: Supplicant enabled, but responds only to received unicast EAPOL messages. • 2: Supplicant enabled and responds to received unicast and multicast EAPOL messages. The parameter can be configured through the following: • Administration menu

Table continues…

Parameter name	Default value	Description
		• `46xxsettings.txt` file
DOT1X	0	Specifies the 802.1X pass-through operating mode. Using this parameter, a phone can pass through 802.1X traffic to and from the PC port and send an EAP-logoff to the network using the MAC address of the device that gets removed from the PC port. The options are: • 0: EAPOL multicast pass-through enabled without proxy logoff. • 1: EAPOL multicast pass-through enabled with proxy logoff. • 2: EAPOL multicast pass-through disabled. The parameter can be configured through the following: • Admin menu • `46xxsettings.txt` file
DOT1XEAPS	MD5	Specifies the authentication method to be used by 802.1X. The options are: • MD5: 802.1X supplicant uses EAP-MD5 authentication. • TLS: 802.1X supplicant uses EAP-TLS authentication.

Setting the 802.1x operational mode

About this task

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

Before you begin

The DOT1X configuration parameter must be set to "0" or "1" for the deskphone to support 802.1X pass-thru and the DOT1XSTAT configuration parameter must be set to "1" or "2" for the deskphone to support supplicant operation.

Procedure

1. Enter the local procedure (Craft) screen using the password as explained earlier.
2. Select **802.1X** option from the screen.

The following parameters are listed:

- DOT1X (802.1X Pass-Thru Mode)
- DOT1XSTAT (802.1X Supplicant Mode)

two settings shown represent the text strings associated with the current configuration parameter values of DOT1X (802.1X Pass-Thru Mode) and DOT1XSTAT (802.1X Supplicant Mode)

3. Select the option you want to change and press the **Change** softkey or the navigation arrows to cycle through the settings.
   - For the Pass-thru mode:
     - "On" if DOT1X = 0
     - "On & proxy logoff" if DOT1X = 1
     - "Off" if DOT1X = 2
   - For the Supplicant:
     - "Off" if DOT1XSTAT = 0
     - "On" if DOT1XSTAT = 1
     - "On with multicast" if DOT1XSTAT = 2
4. Press **Save** to store the new setting.

   The deskphone restarts if you make any change to the 802.1X data.

Setting Site-Specific Option Number

About this task

Caution:

Do not perform this procedure if you are using static addressing. Perform this procedure only if you are using DHCP and the LAN administrator instructs you to do this. Use the following procedure to set the Site-Specific Option Number (SSON).

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. When you select **SSON** from the Administration procedures screen, the following text displays:

   Setting

   where the ***setting*** is the current system value of DHCP_SSON.
3. To change the setting, press the appropriate softkey(s) and use the dialpad to enter a valid SSON value between 128 and 254.

4. Press **Save** to store the new setting and return to the Administration procedures screen.

Setting the Group identifier

About this task

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

Note:

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

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. Select **GROUP** from the Administration procedures screen.
3. Enter a valid **Group** value (0-999).

   Any changes that you make to the Group value results in the restart of the deskphone when you exit the Administration menu.
4. Press **Save** to store the new setting.

GROUP parameter for customized user groups

You might have different communities of users, all of which have the same deskphone model, but which require different administered settings. For example, you might want to group users by time zones or work activities.

Use the GROUP parameter for this purpose:

- Identify which deskphones are associated with which group, and designate a number for each group. The number can be any integer from 0 to 999, with 0 as the default, meaning your largest group is assigned as Group 0.
- At each non-default deskphone, instruct the installer or user to invoke the GROUP from the CRAFT menu and specify which GROUP number to use. The GROUP System value can only be set on a deskphone-by-deskphone basis.

  For more information, see *Installing and Maintaining Avaya 9601/9608/9608G/9611G/9621G/9641G/9641GS IP Deskphones SIP*
- Once the GROUP assignments are in place, edit the settings file to allow each deskphone of the appropriate group to download its proper settings.

Example

The following is an example of a settings file with Deskphones in three different groups - group "0" (the default), group "1", and group "2":

```
## First check if this phone is in group 1.
## If it is, jump to the tag GROUP1
##
IF $GROUP SEQ 1 goto GROUP1
##
## Now check if this phone is in group 2.
## If it is, jump to the tag GROUP2
IF $GROUP SEQ 2 goto GROUP2
##
## The phone is not in either GROUP 1 or 2 so it is in GROUP 0
## {specify settings unique to Group 0}
GOTO END
# GROUP1
## GROUP 1-only settings go here
## {specify settings unique to Group 1}
GOTO END
# GROUP2
## GROUP 2-only settings go here
## {specify settings unique to Group 2}
# END
## The settings here apply to all three groups
## {specify settings common to all Groups}
```

Using the VIEW administrative option

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

About this task

Note:

Unless otherwise prevented using administration, the user can view but not change most of the parameters associated with Craft Local Procedures. For more information about this option, see the applicable user guide(s). If the View Network Information option is not available due to being disabled by administration, use the **ADDR** option to view IP Addresses. The IP Addresses might have been entered incorrectly. Verify whether you were provided with correct IP Addresses.

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. Select **VIEW** from the Administration procedures screen. See View field descriptions for more details.
3. Press **Back** at any time to return to the Administration procedures screen.

VIEW field description

Setting	Description	Associated Configuration Parameter
Model	The model of the phone that is set by factory procedures.	MODEL
Application File	The name of the Signed Application/Library software package.	
Boot File	The name of the boot file.	
FIPS	The FIPS mode is displayed.	FIPS_ENABLED
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
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.	
Gateway	The primary gateway out of the list of configured ones.	ROUTER_IN_USE
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
SNTP Server	The SNTP server that the phone used before to set or update the date and time.	SNTPSRVR_IN_USE
Protocol	Signaling protocol in effect, such as SIP.	
Phone SN	Phone Serial Number	
Exchange Server	The Microsoft Exchange™ server that the phone uses currently.	EXCHANGE_ SERVER_IN_USE

Push server

The Push feature requires a trusted push server. The push server administration must conform to the requirements in *9600 Series IP Telephone Application Programmer Interface (API) Guide*.

SNMP activation

9600 Series IP Deskphones is compatible with SNMPv2c and SMIv2. The deskphones respond correctly to queries from entities that comply with earlier versions of SNMP, such as SNMPv1. The deskphones respond to queries directed either at the MIB-II or the read-only Custom MIB. Read-only means that the values therein cannot be changed externally by means of network management tools.

You can restrict the IP addresses from which the deskphone accepts SNMP queries with the SNMPADD parameter. You can also customize your community string with the SNMPSTRING parameter. For more information, see Chapter 5: Administering DHCP and HTTP servers .

SNMP is disabled by default. Administrators must initiate SNMP by setting the SNMPADD and SNMPSTRING parameters appropriately.

For more information about SNMP and MIBs, see the IETF website: http://www.ietf.org. The Avaya Custom MIB for the 9600 Series IP Deskphones is available for download in *.txt format on the Avaya support website: http://support.avaya.com.

The SIP software release 2.6 MIB is different than that of release 6.5 MIB. Ensure to download the MIBs applicable to your environment.

Registration and authentication

9600 Series IP Deskphones require an outbound proxy SIP (OPS) extension on Communication Manager and a login and password on Session Manager to register and authenticate.

For the SIP Deskphones to work properly, you must specify the correct domain name on the IP Network Region screen of Communication Manager.

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

- For information about the IP Network Region screen, see *Administering Network Connectivity on Avaya Aura® Communication Manager* at the Avaya Support website: http://support.avaya.com.
- For information about the registration process, see *Maintaining Avaya Aura® Session Manager* and *Troubleshooting Avaya Aura® Session Manager*. Also, see your call server administration documentation.

IP address and settings reuse

After a successful registration with a call server, the IP address of the deskphone and the parameter values are saved in the non-volatile memory of the deskphone. The deskphone can reuse the saved parameters if the DHCP or HTTP/HTTPS server is not available for any reason after the deskphone restarts.

IP Address reuse was added to prevent infinite looping when separate DHCP servers are used for voice and data VLANs, and a response is received from the DHCP server on the data VLAN, but not on the voice VLAN.

Unless indicated otherwise, the values described here during IP address reuse are internally provisioned or set by the process itself and not by manual administration.

- Routers in Use - if no responses are received from the routers indicated in the configuration parameter ROUTER (set using DHCP Option 3 or by a local administrative procedure), and if REUSE = 1, then ROUTER_IN_USE will be set to REUSE_ROUTER_IN_USE. With the exception of the ROUTER configuration parameter, the other router-related parameters are internally set system values.
- VLAN Check - During the VLAN check, if a reset is to be done and VLAN_IN_USE is not zero, VLAN_IN_USE will be added to VLANLIST if it is not already on VLANLIST.

  The VLAN detection process described in Automatically detecting a VLAN on page 52 is followed if tagging is off or if tagging is on and L2QVLAN is > 0, and if REUSETIME > 0, and if REUSE_IPADD is not “0.0.0.0”. If VLANTEST expires, the value of VLAN_IN_USE is added to VLANLIST if it is not already on VLANLIST.

If a DHCPOFFER is not received within REUSETIME seconds, or if a DHCPOFFER is received that contains a value of L2QVLAN that is on VLANLIST, REUSE will be set to 1, IPADD will be set to the value of REUSE_IPADD, NETMASK will be set to the value of REUSE_NETMASK, ROUTER will be set to the value of REUSE_ROUTERS, and if the value of REUSE_TAGGING is 1, 802.1Q tagging will be turned on with a VLAN ID equal to the value of L2QVLAN_INIT, DHCP will then enter the “extended” REBINDING state, and operation will proceed as normal.

After a successful registration, the following system values are set:

- REUSE_IPADD will be set to the value of IPADD
- REUSE_NETMASK will be set to the value of NETMASK
- REUSE_ROUTERS will be set to the value of ROUTER
- REUSE_ROUTER_IN_USE will be set to the value of ROUTER_IN_USE
- REUSE_TAGGING will be set to the value of TAGGING
- L2QVLAN_INIT will be set to the value of VLAN_IN_USE
- The MIB object endptVLANLIST will be set to the value of VLANLIST and then the value of VLANLIST will be set to null.

Ping and traceroute

All 9600 Series IP Deskphones respond to a ping or traceroute message sent from the call server switch or any other network source.

For more information, see your call server administration documentation.

TCP and UDP ports

9600 Series IP Deskphones use different protocols, such as TCP, TLS, and UDP to communicate with other equipment in the network. Part of this communication identifies which TCP or UDP ports each piece of equipment uses to support each protocol and each task within the protocol. Depending on your network, you need to know what ports or ranges are used in the operation of the phones.

Received packets (destination = SIP phone)

Destination port	Source port	Use	Protocol UDP or TCP
The number used in the Source Port field of the packets that the HTTP client of the phone sends	Any	Packets that the HTTP client of the phone receives	TCP
The number used in the Source Port field of the TLS/ SSL packets that the HTTP client of the phone sends	Any	TLS/SSL packets that the HTTP client of the phone receives	TCP
68	Any	Received DHCP messages	UDP
SIP messages initiated by the call server should be sent to the port number specified by the value of SIPPORT (TCP) or to the port number specified by the value of SIP_PORT_SECURE (TLS over TCP). Responses to SIP messages initiated by the	Any	Received signaling protocol	TCP

Table continues…

Destination port	Source port	Use	Protocol UDP or TCP
phone should be sent to the number used in the Source Port field of the message from the phone.			
The number used in the Source Port field of the DNS query that the phone sends	Any	Received DNS messages	UDP
The number used in the Source Port field of the SNTP query that the phone sends	Any	Received SNTP messages	UDP
161	Any	Received SNMP messages	UDP

Transmitted packets (source = SIP phone)

Destination port	Source port	Use	Protocol UDP or TCP
53	Any unused port number	Transmitted DNS messages	UDP
67	68	Transmitted DHCP messages	UDP
80, unless explicitly specified otherwise	Any unused port number	Packets transmitted by the HTTP client of the phone	TCP
123	Any unused port number	Transmitted SNTP messages	UDP
The number used in the Source Port field of the SNMP query packet received by the phone	161	Transmitted SNMP messages	UDP
443, unless explicitly specified otherwise	Any unused port number	TLS/SSL packets transmitted by the HTTP client of the phone.	TCP
514	Any unused port number	Transmitted Syslog messages	UDP
The port number specified in the test request message	50000	Transmitted CNA test results messages.	UDP

Table continues…

Destination port	Source port	Use	Protocol UDP or TCP
System-specific	Any unused port number	Transmitted signaling protocol packets	TCP
FEPORT + 1 (if FEPORT is even) or FEPORT -1 (if FEPORT is odd) or the port number specified in a CNA RTP test request plus or minus one, as with FEPORT	PORTAUD + 1 (if PORTAUD is even) or PORTAUD – 1 (if PORTAUD is odd) or the port number reserved for CNA RTP tests plus or minus one, as for PORTAUD, above	RTCP packets transmitted to the far-end of the audio connection	UDP
RTCPMONPORT	PORTAUD + 1 (if PORTAUD is even) or PORTAUD – 1 (if PORTAUD is odd)	RTCP packets transmitted to an RTCP monitor	UDP
System-specific	Any unused port number	Transmitted signaling protocol packets	UDP

Preinstallation checklist for static addressing

Before performing static addressing, verify all the network requirements first. In addition, you must have the values for the following parameters:

- IP address of the deskphone.
- IP address of the router.
- IP subnet mask.
- IP address of the HTTP and/or HTTPS server (applicable for only DHCP server settings).
- IP address of the DNS Server.
- VLAN ID (the L2QVLAN value).
- VLANTEST value.

Assigning static IP address

Use static IP addressing, if the setup does not require a DHCP server.

Procedure

1. In the Administration menu, go to **ADDR**.
2. In the **IP Address** field, enter the IP address. For the description of other fields, see Static addressing field descriptions.

3. Use the navigation arrows to scroll to and highlight the address you want to change. Choose one of the following options:

Option	Description
IP Address	IP addresses have four sets of three digits followed by a period. Pressing ***** followed by three digits causes a period to be placed in the next position, and the cursor to advance one position to the right.
VLAN ID	Use the dialpad to enter the new static VLAN ID of from 0 to 4094, inclusive.
VLANTEST	Use the dialpad to enter the new value of the DHCPOFFER wait period of from 0 to 999, inclusive.
Host to Ping	Use the dialpad to enter the host server IP address or the DNS name of the server you want to check. The deskphone sends four pings and displays the results in a text block screen. If the ping fails, the following message is displayed. `Unable to contact host`

4. Do one of the following:
   - Press **Save** to store the new setting.
   - Press **Cancel** to return to the Administration menu.
   - Press **OK** after setting ping to the host server and return to the Static Address screen.

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

Static addressing field descriptions

Configuration Parameter Name	Description
Use DHCP	Choose one of the following options: • **Yes**: Selects the DHCP option. • **No**: Deselects the DHCP option.
Phone	Specifies the IP address of the deskphone. 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***
HTTPS File Server	Specifies the IP address of the HTTPS file server. The available format is ***nnn.nnn.nnn.nnn***

Table continues…

Configuration Parameter Name	Description
HTTP File Server	Specifies the IP address of the HTTP 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***
802.1Q	Choose one of the following options: • **0**: Automatic mode. • **1**: Turns on the configuration. • **2**: Turns off the configuration.
VLAN ID	Specifies the ID for VLAN. The available format is ***dddd***.
VLAN TEST	Specifies the duration to wait for the DHCP in seconds. The available format is ***ddd***.
Host to ping	Specifies the IP address or the DNS name. The available format are: • ***nnn.nnn.nnn.nnn*** : For IP address. • ***AVohhhhhh*** : For DNS name. ***hhhhhh*** are ASCII characters for the hexadecimal representation of the last three octets of the deskphone's MAC address ***o*** has one of the following values: - “A” if the OID is 00-04-0D - "B" if the OID is 00-1B-4F - "E" if the OID is 00-09-6E - "L" if the OID is 00-60-1D - "T" if the OID is 00-07-3B - "X" if the OID is anything else

Administering display language options

By default, the phone display information is in English. Administrators can specify up to four languages for each phone to replace English. Users can then select the display language.

All downloadable language files contain all the information needed for the phone to present the language as part of the user interface.

The actual character input method does not depend on the languages available from the software download. If the phone does not support a character input method, use ASCII instead.

Use the configuration file and the following parameters to customize the settings for up to four languages

- SYSTEM_LANGUAGE - Contains the name of the default system language file used in the phone. The filename must be one of the files listed in the LANGUAGES parameter. If no filename is specified, or if the filename does not match with one of the LANGUAGES values, the deskphone uses its built-in English text strings. Filename must end in .xml
- LANGUAGES - Specifies the language files to be installed or downloaded to the phone. Filenames may be full URL, relative path name, or filename. (0 to 1096 ASCII characters, including commas). Filenames must end in .xml. For example, to indicate that and Russian, Parisian French, Latin American Spanish, and Korean are the available languages, the setting is **`SET LANGUAGES Mlf_Russian.xml,Mlf_ParisianFrench.xml,Mlf_LatinAmericanSpanish.xml,Mlf_Korean.xml`**

To download a language file or to review pertinent information, go to the Avaya Support website.

Note:

Specifying a language other than English in the configuration file has no impact on Avaya Communication Manager settings, values, or text strings.

Network audio quality

Users can view icons on the deskphone that provide information about:

- Audio quality of a call. A Local Network Quality (LNQ) icon appears whenever the audio quality of a call is below a certain threshold. You can define the threshold in the settings file using the LNQ and QLEVEL_MIN parameters. LNQ is based on a combination of jitter, packet loss, and delay.
- Use of the wide band codec. The HD icon appears if the deskphone uses the wide band codec. You can enable or disable the icon based on the value you assign to the WBCSTAT parameter in the settings file.

For more information about the settings file parameters, see Customizable system parameters for SIP-based 9600 Series IP Deskphones on page 2.

Users can also monitor the network audio performance on a call. The Network Information screen displays the audio network. Users can gain access to the Network Information screen from the Avaya menu. On a touchscreen deskphone, users can gain access to the Network Information screen from the Home screen.

The implication for LAN administration depends on the values the user reports and the specific nature of your LAN, like topology, loading, and QoS administration. This information gives the user an idea of how network conditions affect the audio quality of the current call. Avaya assumes you have more detailed tools available for LAN troubleshooting.

Network progress tones overview

The SIP-based 9600 Series IP Deskphones provide country-specific network progress tones which are presented to the user at appropriate times. The tones are controlled by administering the COUNTRY parameter for the country in which the deskphone will operate. Each Network Progress Tone has the following six components:

- Dialtone
- Ringback
- Busy
- Congestion
- Intercept
- Public Dialtone

All countries listed in this appendix are applicable to the 96xx phones. Some of the dialtone entries have changed from previous releases to be distinctively different than the public dialtone entries.

Alphabetical country list

A:

- Abu Dhabi
- Albania
- Argentina
- Australia
- Austria

B:

- Bahrain
- Bangladesh
- Belgium
- Bolivia
- Bosnia
- Botswana
- Brunei
- Bulgaria

C:

- China (PRC)
- Colombia
- Costa Rica
- Croatia

- Cyprus

D:

- Denmark

E:

- Ecuador
- El Salvador
- Egypt

F:

- Finland
- France

G:

- Germany
- Ghana
- Greece
- Guatemala

H:

- Honduras
- Hong Kong

I:

- Iceland
- India
- Indonesia
- Ireland
- Israel

J:

- Japan
- Jordan

K:

- Kazakhstan
- Korea
- Kuwait

L:

- Lebanon
- Liechtenstein

M:

- Macedonia
- Malaysia
- Mexico
- Moldova
- Morocco
- Myanmar

N:

- Netherlands
- New Zealand
- Nicaragua
- Nigeria Norway

O:

- Oman

P:

- Pakistan
- Panama
- Paraguay
- Peru
- Philippines
- Poland
- Portugal

Q:

- Qatar

R:

- Romania
- Russia

S:

- Saudi
- Arabia
- Serbia
- Singapore
- Slovakia
- Slovenia

- Spain
- South Africa
- Sri Lanka
- Swaziland
- Sweden
- Switzerland
- Syria

T:

- Taiwan
- Tanzania
- Thailand
- Turkey

U:

- Ukraine
- United Arab Emirates
- United Kingdom
- Urugua
- USA

V:

- Venezuela
- Vietnam

Y:

- Yemen

Z:

- Zimbabwe

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. 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. 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.

Table continues…

Parameter name	Default value	Description
		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.

Setting the dial plan on SIP deskphones

Note:

This section only applies to operations with a secondary controller where Communication Manager or Session Manager is not available. In a failover situation, the dial plan is played locally even if a proxy connection is not available. The user might hear a dial tone but cannot make a call.

During manual dialing, a dial plan allows a call to be initiated without using a **Send** button and without waiting for the expiration of a timeout interval. The dial plan consists of one or more format strings. When the dialed digits match a format string in the DIALPLAN configuration parameter, the call is initiated.

Valid characters in a format string, and their meanings, are as follows:

- digits 0 through 9, inclusive = Specific dialpad digits
- * = the dialpad character *
- # = the dialpad character # (but only if it is the first character in the dialed string – see below)
- x = any dialpad digit (i.e., 0-9)
- Z or z = present dial tone to the user (for example, for Feature Access Code (FAC) entry)
- [ ] = any one character within the brackets is a valid match for a dial plan string
- - = any one digit between the bounds within the brackets, inclusive, is a match
- + = the character following the + can repeat 0 or more additional times, for a valid match

An individual valid dial plan is any combination of the above characters. If there are multiple valid dial plans, separate each one from the next using an OR symbol ("|"). If the dial plan text string begins or ends with an OR symbol, that symbol is ignored. Users cannot modify the dial plan.

Dial plan example:

"[2-4]xxx|[68]xxx|*xx|9Z1xxxxxxxxxx|9z011x+"

where:

- **[2-4]xxx**: Four-digit dial extensions, with valid extensions starting with 2, 3, or 4;
- **[68]xxx**: Four-digit dial extensions, with valid extensions starting with 6 or 8;
- ***xx**: Two-digit Feature Access Codes, preceded by a *;
- **9Z1xxxxxxxxxx**: Network Access Code ("9 for an outside line"), followed by dial tone, followed by any string of 10 digits– typical instance of Automatic Route Selection (ARS) for standard US long distance number;
- **9z011x+:** Network Access Code ("9 for an outside line"), followed by dial tone, followed by at least one digit – typical instance of Automatic Route Selection (ARS) for US access to international numbers of unknown, and variable, length.

Additional parameters that affect dialing are as follows:

- COUNTRY - Country of operation for specific dial tone generation.
- PSTN_VM_NUM (PSTN access number for Voice Mail system) - This parameter specifies the telephone number to be dialed automatically when the deskphone user presses the Messaging button under a non-AST controller. The deskphone places a PSTN call out from the local office and back in to the location that houses the voice mail server. Additional codes necessary to reach a specific user's voice-mail box may also be included. Example 1. SET PSTN_VM_NUM 96135550123
- ENABLE_REMOVE_PSTN_ACCESS_PREFIX - When the deskphone is operating with a non-AST controller and the value of the parameter is 0, the PSTN access prefix, defined by the parameter PHNOL, is retained in the outgoing number. If the value is 1, then the PSTN access prefix is stripped from the outgoing number.
- PHNLAC - A string representing the deskphone's local area code. When set, this parameter indicates the endpoint's local area code, which along with the parameter LOCAL_DIAL_AREA_CODE, allows users to dial local numbers with more flexibility. Example: SET PHNLAC 617
- LOCAL_DIAL_AREA_CODE - A flag indicating whether the user must dial the area code for calls within same area code regions. When the parameter is 0, the user does not need to dial the area code; when this parameter is 1, the user needs to dial the area code. When this parameter is enabled (1), the area code parameter (PHNLAC) should also be configured (i.e., not the empty string). Example: SET LOCAL_DIAL_AREA_CODE 1

Example 1 - Setting the parameter configuration:

- **SET ENHDIALSTAT 2**
- **SET PHNOL 27**
- **SET PHNCC 1**
- **SET PHNDPLENGTH 7**
- **SET PHNLDLENGTH 11**
- **SET PHNLD 0**

- **SET PHNIC 001**

Table 2: : Example 2 In the Contacts list, save Contact X with the telephone number 41018989

PHNLAC Parameter Value	LOCAL_DIAL_AREA_CODE Parameter Value	Step to Execute	Result
020	1	Call X from Contacts list	Phone sends an invite message with 2702041018989.
020	1	Call X from Contacts list	Phone sends an invite message with 2741018989 and does not insert the local area code.
Null	1	Call X from Contacts list	Phone sends an invite message with 2741018989 and does not insert the local area code.

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.

Administering audio equalization

Part 68 of the Federal Communication Commission (FAC) rules governs the connection concerning Hearing Aid Compatibility (HAC) and volume control for phones. The HAC feature is an alternative way to provide audio equalization on a handset, from the acoustic standards specified in TIA-810/920 and S004, and may be of benefit to some users of t-coil capable hearing aids.

- Settings File: The administrator can set ADMIN_HSEQUAL. The default value, 1, specifies Handset equalization that is optimized for acoustic TIA 810/920 performance unless otherwise superseded by Local Procedure or User Option. The alternate value, 2, specifies HAC.
- Local Procedure: When users are denied access to Options for administrative reasons, but individual users need an equalization value other than the one in the settings file, the HSEQUAL Craft Procedure provides another method to administer the deskphone with the audio equalization value that you require. Use the Craft password to gain access to the Admin Procedures screen. The default password is **27238**. "Default" uses the settings file value unless superseded by User Option. "Audio Opt." is optimized for TIA-810/920 acoustic performance, and "HAC Opt." is optimized for HAC telecoil performance.
- User Option: The user can select "Default" by which the deskphone uses the settings file value (unless superseded by Local Procedure), "Audio Opt." which uses Handset equalization that is optimized for acoustic TIA 810/920 performance, or "HAC Opt." which uses Handset equalization that is optimized for electrical FCC Part 68 HAC telecoil performance.
- Handset equalization options are effected in the following order:
  - The deskphone uses the User Option value if selected and saved.
  - If a Local Procedure value was selected and saved, the deskphone uses the local procedure value.
  - If a Settings file value is specified and saved the deskphone uses that value.
  - If none of the above options are set, the deskphone uses Handset equalization that is optimized for TIA-810/920 acoustic performance.

Setting the handset audio equalization

About this task

Use the following procedure to configure the Handset Equalization settings:

Procedure

1. On the user menu, navigate to **Settings > Options&Settings > Advanced Options > Handset Equalization**.
2. Using the navigational keys or by tapping the arrow icons on the screen, choose any of the following:
   - Default
   - Audio Opt

- HAC Opt

3. Press **Save**.

Enabling and disabling Automatic Gain Control

About this task

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

Procedure

1. Enter the Administration menu and select **AGC**.
2. Choose one of the following options:
   - Handset AGC: Toggle “On” or “Off” using navigation arrow or **Change** softkey.
   - Headset AGC: Toggle “On” or “Off” using navigation arrow or **Change** softkey.
   - Speaker AGC: Toggle “On” or “Off” using navigation arrow or **Change** softkey.
3. Press **Save** to store the configuration.

Administering headset profiles

The SIP-based 9600 Series IP Deskphones support signaling for wireless headsets that are connected through the analog headset port on the deskphone. Users can make, answer, or disconnect calls through controls on the headset. Users call also control the headset through the buttons available on their deskphone.

Using the headset

- When the deskphone receives a call, the headset produces an audible beep and the user can answer the call by pressing a button on the headset.
- When a call is established, the user can disconnect the call by pressing a button on the headset.
- When a user makes or receives a call using the buttons on the deskphone, the headset automatically connects the speech path without the need for the user to press any buttons on the headset.

Note:

By default, the deskphone displays the name of the headset profiles that Avaya provides. You can use the HEADSET_PROFILE_ NAMES parameter to change the name of the headset profiles.

Calibrating the touch screen

Screen calibration properly aligns the touch screen but should only be used for a significant problem with the touch screen.

Important:

Use a pencil, a pen, or a stylus rather than your finger to touch the calibration points precisely.

Note:

The CLEAR Craft procedure clears any calibration data set using the CALIBRATE SCREEN Craft procedure, but does not affect factory-set calibration data. Use the **Default** softkey to restore factory-set calibration. Calibration results are not saved as part of a backup. The feature is not supported on the 9641GS deskphone.

About this task

Use this procedure to calibrate 9621 and 9641 touch screen phones.

Procedure

1. Enter the Administration menu using the password.
2. Select the **CALIBRATE SCREEN** form the screen.
3. Choose one of the following options:
   - **Cancel**: To returns to the Administration menu screen without calibrating.
   - **Default**: To reset the calibration parameters to the factory default.
   - **Start**: To calibrate the screen.
     - Touch the center of the target with the stylus as soon as it appears.
     - Touch the next target's center with the stylus within 10 seconds of its appearance.
     - Repeat steps to complete all four targets with the following message:`Calibration successful`
4. Press **Save** to save the settings.

Using the Debug Mode

Before you begin

Ensure that the following values are changed:

- Value of PROCPSWD parameter is set to a value other than default. When the ADMIN_PASSWORD is set, then the parameter PROCPSWD is not used.
- Value of SLMCAP parameter is set to “3”.
- Value of SLMCTRL parameter is set to “2”.

Procedure

1. Use the Craft password to gain access to the Administration procedures screen.
2. Navigate to **DEBUG** from the Administration procedures screen. The following debug options are available:

- Debug Mode
- Service mode control
- Service mode record

3. To enable or disable the options, tap or use the appropriate buttons.
4. Press **Save** to store the new setting.

   Restart the deskphone for the DEBUG settings to take effect.

Setting interface control

About this task

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

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. When you select **INT...** from the Administration procedures screen, the following text displays with a prompt to use the Right and Left navigation arrows to select a setting:

   Ethernet: Choice Selector

   PC Ethernet: Choice Selector

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

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

Note:

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

The PHY2STAT text strings are:

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

Note:

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

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

   Depending on the current value, the next sequential text string is selected and displayed as the setting. For example, if the current value is 10 Mbps half (2), pressing the Right navigation arrow changes the value to 10 Mbps full (3). If the current value is 1000 Mbps full (6), pressing the Right navigation arrow changes the value to Auto (1).
4. To change the PC Ethernet setting, select that line and press the Right navigation arrow or **Change** to cycle through the possible settings.
5. Press **Save** to store the new setting(s) and redisplay the Admin Procedures screen.

Enabling and disabling event logging

About this task

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

Procedure

1. Use the Craft password to gain access to the Administration procedures screen. The default password is **27238**.
2. When you select **LOG** rom the Administration procedures screen, the deskphone prompts you to use the Right and Left navigation arrows to select and change a setting and displays the following text:

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

- "Emergencies" when SYSLOG_LEVEL = 0
- "Alerts" when SYSLOG_LEVEL = 1
- "Critical" when SYSLOG_LEVEL = 2
- "Errors" when SYSLOG_LEVEL = 3
- "Warnings" when SYSLOG_LEVEL = 4
- "Notices" when SYSLOG_LEVEL = 5
- "Information" when SYSLOG_LEVEL= 6
- "Debug" when SYSLOG_LEVEL = 7

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

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

4. Press **Save** to store the new setting and redisplay the Administration procedures screen.

   Logging has a direct impact on performance. Only turn on the required categories and turn them off as soon as logging is not required.

History

The History screen displays:

- The calls that the user makes from and receives on the deskphone.
- The instant messages that the user sends from and receives on the deskphone.

The user can select to see one of the following types of call history:

- All Calls
- Answered Calls
- Missed Calls
- Outgoing Calls
- Instant Messages: This option is only available for touch phones.

The user can get information about the Caller ID, Caller number, time and date of the call, and call duration.

Call treatment in a logged out state and busy Call Appearances

If a deskphone gets a call when all Call Appearances are busy, the deskphone does not display the incoming call, but the deskphone records the call as a missed call and displays it on the History screen.

If a deskphone gets a call when the user is not logged in, the system saves the incoming call in a database. When the user logs in again, the deskphone displays the previous call log and the calls received in the logged out state.

You need to configure the Call Log feature through System Manager. The Call Log feature has the following limitations:

- The Call Log feature is available only when the deskphone is connected to the primary Session Manager.
- When a user logs into multiple devices through the Mulitple Device Access (MDA) configuration, the History screens might not be synchronized between devices. If the user deletes Call Logs from the History screen in one MDA device, the other MDA devices might not clear the Call Logs from the History Screen.
- If the user disables the Call Log feature locally on the deskphone, call logs are stored on the server. When the user enables the Call Log feature, all call logs might show up.
- The deskphone does not display the Group Page calls with the name of the initiator of the Group Page, but with the name Group Page.
- Calls that come to deskphones that are redirected might not show the complete set of History details.

Customizing ring tones

There are fourteen (14) standard ring tones, eight (8) classic, and six (6) rich ring tones available for setting. The ring tones are set on the Personal Profile Manager (PPM). Ring tones for external, internal, priority, and intercom calls (distinctive ringing) are combinations of specified frequency, duration and cadence values. You can replace the eight (8) classic ring tones with the Korean ring tones or the customized ring tones.

Korean ring tones

Korean ring tones are part of the SIP software bundle download. To administer all or any of these tones to replace the existing external, internal, priority and intercom call tones, set the EXTEND_RINGTONE parameter in the settings file with the names of the Korean tones you want available to the user. For example, to replace all the Avaya classic ring tones with the Korean ring tones, set the EXTEND_RINGTONE parameter to Korean ring tone XML files as follows:

```
SET EXTEND_RINGTONE = KoreanRT1.xml,KoreanRT2.xml,KoreanRT3.xml,KoreanRT4.xml,
KoreanRT5.xml,KoreanRT6.xml,KoreanRT7.xml,KoreanRT8.xml
```

To administer only the second and fourth Korean ring tones to replace the second and fourth Avaya standard tones, you would specify (without spaces between entries) :

```
SET EXTEND_RINGTONE = KoreanRT2.xml,KoreanRT4.xml
```

Note:

Do not include space between the XML file names.

Customized ring tones

A Ringtone Creation Tool in file `ringtone.xls` is included with the software download package. Use this tool to create XML files for up to eight custom ring tones as described in this section. Then:

- Save the custom ring tones to an HTTP server,
- Set the EXTEND_RINGTONE parameter with the name(s) of the XML file(s) you created,
- Reboot the deskphone to make the custom tone(s) available to the end user through the Avaya (A) Menu -> Options & Settings -> Screen & Sound option.

Important:

When setting up multiple ring tone files using the EXTEND_RINGTONE parameter, be sure that there are no spaces before or after the comma separating the filenames.

To create a custom ring tone, open the Ringtone. XLS spreadsheet and provide a value for each of the cells/fields in Ring tone XLS cell descriptions. A sample spreadsheet follows the table for illustration purposes only.

Ring tone XLS cell descriptions

Cell Name	Description	Comment
Ringer Name	Name of this custom Ring Tone file, for example, Ringtone 1	This filename will be assigned a .XML extension upon completing all required cells and pressing the "Create xml" cell button.
Ringer Index	This numbers the xml file as one of the 8 patterns used in personalized ringing. For example, index 2 will be the second personalized ringing choice a user will have on their deskphone. Eight xml files with indices 1-8 need to be created to customize all the available personalized ringing choices that will be presented on a deskphone. If less than 8 indices/files are set in the settings file, Avaya standard	

Table continues…

Cell Name	Description	Comment
	ringing patterns will be used for the missing indices.	
Type of Wave	Leave empty; this cell is not currently used.	Reserved for future use.
Number of Active Frequencies	Up to four active frequencies can be set. Valid values are 1, 2, 3, or 4.	
Frequency Values	The range of frequency values is from 0 to 3999Hz.	
Number of Notes	Number of notes in this ring tone, from 1 to 3. A note is an interval in which a frequency is used. Currently, a custom ring tone has a 3 note maximum.	
Note 1, 2, and 3	This value represents a collection of frequency intervals that are grouped together and repeated over and over again as the ring tone	
Note Pulse State	The pulse state has two possible settings - On or Off	
Note Frequency	The frequency used for a particular note.	
Note Duration	The duration of the note in milliseconds, from 0 to 2^16.	
Next Note	Leave empty; this cell is not currently used.	Reserved for future use.
Cadence Patterns and States	Cadence patterns are set for internal, external, priority, and intercom calls.	
Cadence 1 to 8	The number of available cadences.	
Cadence Duration	The duration of the cadence in milliseconds, from 0 to 2^16.	
Next Cadence	The next cadence is executed after the current cadence value is completed. This is used to create a loop of notes. For example, if number 1 is used for cadence state 8, when cadence 8 is completed, cadence 1 will follow.	
Cadence Next Index	Leave empty; this cell is not currently used.	Reserved for future use.

Table continues…

Cell Name	Description	Comment
Reserved for future use.	When all applicable cells have been filled in, use this control to create an xml file for this specific tone.	

Downloadable ring tones

You can specify list of audio files that deskphones can download as ring tones. The users can select the required ring tone from the downloaded list.

The audio files list can contain 0 to 1023 UTF-8 characters. Provide the list of audio files as name-value pairs separated by commas without any intervening spaces, where:

- name: Is the display name that you assign to the ring tone. Ensure that you:
  - Do not include a comma, an equals sign (=), or a double quote character in the display name.
  - Quote the entire list if you include spaces in any of the display name.
  - Specify the display name that has the length that your deskphone can display. If the length of the display name exceeds the specified limit of your deskphone, the deskphone truncates the name.
  - Do not specify a display name that contains only numbers.
- value: Is the relative or an absolute URL of the audio file. URLs can include an equals sign (=). If you include the equals sign, the system treats the first equals sign as the separator between the display name and the URL. The system treats any subsequent equals signs after the first sign as a part of the URL. Percent encode a comma if you use it in the URL.

Ensure that audio files must:

- Be single-channel WAV files.
- Have encoding as per ITU-T G.711 A-law or u-law PCM with 8-bit samples at 8 kHz or 16-bit samples at 16 kHz.
- Have a maximum size of 512 KB.
- Have the combined size not more than 5 MB.
- Only contain ASCII characters in their file names.

For example,

```
SET RINGTONES "Steam Whistle=tones/swhistle.wav,Car Horn=tones/chorn.wav,Loud Burp=tones/
lburp.wav"
```

Administering voice mail

Set the MSGNUM parameter for configuring the **Messages** button. Configure the MSGNUM parameter with one the following:

- A standard telephone number the phone should dial to access your voice mail system, such as AUDIX or Octel.
- A Feature Access Code (FAC) that is configured for the Feature "To Voice Mail" will allow the user to transfer the active call directly to voice mail. FACs are supported only for QSIG-integrated voice mail systems like AUDIX or Octel. QSIG is an enhanced signaling system that allows the voice mail system and Communication Manager Call Processing to exchange information.

When the user presses the **Messages** button on the phone, that number or FAC is automatically dialed, giving the user one-touch access to voice mail.

The settings file specifies the phone number to be dialed automatically when the user presses this button. The command is:

```
SET MSGNUM 1234
```

where `1234` is the Voice Mail extension (Communication Manager hunt group or VDN).

However, when the phone is in failover mode, the phone might need to dial a different number to reach the voice mail system. Administer the parameter PSTN_VM_NUM to access the voice mail system through the Message button, while the phone is in failover mode. For example, during failover, the phone dials the public access number `345-555-1234` to access the voice mail at 1234. In this case, the command would be:

```
SET PSTN_VM_NUM 913455551234
```

Note:

If you set two different values for the same parameter through PPM and the `46xxsettings.txt` file, the PPM settings override the settings in the `46xxsettings.txt` file.

Administering Presence

Presence overview

Presence is a feature that indicates the availability status of a contact. The phone periodically publishes its own presence status and retrieves the presence status of a contact from Avaya Aura® Presence Services.

Presence services facilitates aggregation of presence information collected from the following resources:

- Application Enablement ServicesServices

- Microsoft Office™ Communicator Server
- eXtensible Messaging and Presence Protocol (XMPP) Server

Presence services categorizes users into two types:

- **Watcher**: A user who is viewing the presence status.
- **Presentity**: A contact whose presence status is being viewed. Presentity is also referred to as Buddy.

A user can simultaneously be a Watcher and Presentity.

Users can manually set their own presence through the phone menu. You can configure the phone to use the Send All Calls (SAC) feature when the user sets the Do Not Disturb (DND) status.

Access Control List

The Access Control List specifies whether other users on the network can view the presence of another user. By default, users can automatically see the presence of other users. You can disable the automatic viewing by setting the `PRESENCE_ACL_CONFIRM` parameter in the `46xxsettings` file.

The phones that are in 1XC Shared Control Mode with a 1XC soft client can control automatic viewing. The 1XC softclient displays a popup window to the presentity that prompts whether to allow or to restrict the display of the presence information.

Instant Messaging

Instant Messaging (IM) is available to the user if the Presence feature is operational, and the SIP signaling is set to TLS. Users can communicate through text messages by using this feature. The phone stores the IM conversations in the History log. Each phone can store IMs up to 2 MB. If a user logs out of a phone and logs back into a different phone, the IM history is not available.

Presence profile

Presence profile is a part of the communication profile of a user. Presence profile is configured in **User Profile Management** of Avaya Aura® System Manager. In cluster deployment of multiple presence servers, presence profile explicitly associates a user with a presence server instance. The `46xxsettings.txt` file supports the use of only one presence server address. Using presence profile is an efficient method to ensure that a user is connected to the appropriate presence server. For information about configuring a presence profile, see *Avaya Aura® Presence Services*.

Avaya Aura® Call Center Elite features

The 9600 SIP software supports Avaya Aura® Call Center Elite features on 9600 Series IP Deskphones models 9608, 9608G, 9611G, 9621G, 9641G, and 9641GS. The Avaya Aura® Call Center Elite features are not supported on the 9601 deskphones Avaya Aura® Call Center Elite 6.2 is the minimum requirement to support the following Call Center Elite features:

- Agent login/logout

- After call work
- Auxiliary work
- Auto and Manual in
- CC-Info agent event package
- Third party MWI
- Stroke counts
- Call work codes
- Display active VDN name
- VuStats
- Accept and display ASAI UUI information
- Forced logout override
- Supervisor assist
- Service observe
- QStats
- Interruptible AUX work

For detailed information on all the Call Center features, see *Using Avaya 96X1 SIP Agent Deskphones with Avaya Aura® Call Center Elite*, and *Administering Avaya Aura® Call Center Elite*, available on the Avaya support Website www.avaya.com/support.

You can assign the **Call Center** button features through PPM. The system displays these buttons in the Feature screen, button module, Quick Touch Panel, Phone screen, and the Home screen.

You can administer the Call Center features through the settings file using the following parameters:

Parameter name	Parameter Description
SKILLSCREENTIME	Sets the duration for which the deskphone displays the Skills screen
ENTRYNAME	Sets the entry name as the calling party name or the VDN or Skills name.
UUIDISPLAYTIME	Specifies the duration for which the deskphone displays the user to user information (UUI) screen.
CC_INFO_TIMER	Sets the CC-Info event package timer.
BUTTON_MAPPINGS	Specifies a list of Button=Status pairs that change the operation of some of the buttons on the phone. Valid Button values are "Forward", "Speaker", "Hookswitch", and "Headset". Valid Status values are "na" and "cc-release". The default is null.

Team Button overview

The Team Button feature provides the facility to watch or monitor the phone of another user. The feature is configured on Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

By pressing the **Team** softkey, a user can do the following:

- Make a speed dial call to the monitored phone when it is free.
- Transfer an active call to the monitored phone.
- Monitor whether the phone redirects calls to any other phone.
- Monitor whether the phone has an active call.
- Answer a call that rings on the monitored phone.

The Team Button feature is configured on the Avaya Aura® Communication Manager and Avaya Aura® Session Manager. You can customize the ring type for Team Buttons on a monitored phone through a system parameter: TEAM_BUTTON_RING_TYPE.

Team Button parameters

Parameter	Default value	Description
TEAM_BUTTON_REDIRECT_INDICATION	0	Controls if the redirection indication is shown on a Team Button on a monitoring station. Valid values are 0 and 1.
TEAM_BUTTON_REDIRECT_OVERRIDE	0	Specifies whether the monitoring deskphone can override the SAC, CFWD, ECF features set by the monitored deskphone. Valid values are: • 0: Monitoring deskphone cannot override the features. • 1: Monitoring deskphone cannot override the features by placing the call to the monitored deskphone. • 2: Monitoring deskphone displays a message to the user asking if the call should be placed to the monitored deskphone.

Table continues…

Parameter	Default value	Description
		Note: The value of this parameter is stored in PPM.
TEAM_BUTTON_RING_TYPE	3	Specifies the default ring tone for all **Team Buttons**. Valid values are 1 to 8.
TEAM_BUTTON_RING_TYPE_PER_BUTTON	NULL	Specifies a list of name-value pairs that indicate default ring tones.

Team Button override

The monitored phone might have the call redirection feature active through one of the following features:

- Send All Calls
- Call Forward
- Enhanced Call Forward

You can configure Avaya Aura® Communication Manager and Avaya Aura® Session Manager so that the monitoring phone overrides the call redirection active on the monitored phone. If the call redirection override is set, the monitored phone rings for 30 seconds. If no one answers the call, the call is automatically sent to the redirection number.

Direct Transfer

The Direct Transfer feature provides the facility to transfer an active call to the monitored station.

When a call is answered at a monitoring phone, the user can begin the process of transferring the call to the monitored station by pressing the line corresponding to the Team button. The monitoring phone puts the existing call on hold and places a call to the monitored station. A speech path is established, and the monitoring station and monitored station can interact.

To complete the transfer between the original call and the monitored phone, the user at the monitoring station can disconnect the call by going on-hook or pressing the **Complete** softkey.

Direct Transfer is applicable only to the Team Button feature, and you need not configure the Direct Transfer feature on Avaya Aura® System Manager.

Enhanced Call Forward

The Enhanced Call Forward (ECF) feature provides flexibility to forward incoming calls to different destinations, based on the following different conditions:

- Caller type: Whether the call is coming from internal number or an external number.
- Phone status: Based on the phone status a user can choose to forward the calls in the following scenarios:
  - All incoming calls.
  - When the phone is busy.
  - When there is no answer on the phone.

You can set the **ECF** feature button for the phone to display on the Feature Screen, Quick Touch Panel, SBM24, BM12, Favorite Feature on the Phone Screen, or the Home Screen. Use the **ECF** feature button to access and enable or disable ECF.

Advanced call conference

In collaboration with the Avaya Aura® Conferencing server and Communication Manager, SIP release 6.5 or later provides advanced call conference features to the end user.Avaya Aura® Conferencing 7.0 is the minimum requirement to support advanced call conference. See the following table for a comparison of the advanced call conference features with different conference servers:

Feature	AST	AAC7
Participant list	No	Yes
Participant drop	Last party drop	Selective drop
Add participant	Single dial	Dial group
Voicemail	No filter	Voicemail
Number of participants	6	Unlimited
What is displayed	Number of participants, on the call appearance line, or on the Top Line.	The Details screen displays up to 25 participant names and their presence status. The Phone screen or the Top Line displays the total number of participants.
Selective mute	No	Yes

Note:

The 9601 phone does not support the conference details screen.

To enable advanced call conference features, you must first set the following parameters through the settings file:

- CONFERENCE_FACTORY_URI
- CONFERENCE_SERVER_ADDRESS
- CONFERENCE_SERVER_PORT
- ENABLE_SECURE_HTTP_FOR_CONFERENCING_SERVICE

Note:

SIP 6.5 does not support non-AST servers as primary controller for conferencing.

Assured services SIP

The Assured services SIP (AS-SIP) contains the following features:

Multi-Level Precedence and Preemption

The Multi-Level Precedence and Preemption (MLPP) feature is provided for the “dsn” and “uc” network of the U.S. DoD and Canadian DND. With this feature, users can place calls at various levels of precedence. Users are notified with precedence tone upon receiving incoming precedence calls. Higher precedence calls preempt the lower precedence calls when a user has no idle call appearance. Five precedence levels are supported for the “dsn” and “uc” network domains: Routine, Priority, Immediate, Flash, and Flash Override.

DSCP

The DSCP feature classifies outgoing traffic from SIP phones by marking each outgoing packet with a DSCP value. User signaling packets are marked according to the value of the DSCPSIG configuration parameter. Audio packets are marked with a DSCP value that is converted from the Precedence or Priority level of each call when the MLPP feature is enabled. OA&M packets are marked with the value of the DSCPMGMT configuration parameter.

Blind Transfer

With the Blind transfer feature, users can transfer an active call to another party without consultation.

Multiple Device Access

9600 Series IP Deskphones 6.3 and later support Multiple Device Access (MDA) with which you can register up to 10 SIP devices with your extension.

With MDA, you can do the following:

- 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.

Setting a large font size for the display

About this task

Use the following procedure to display the English language text in a larger font on the phone display. When you configure the large fonts feature, the phone displays the **Text Size** option in the **Options & Settings** menu. The large text size setting applies to the phone display as well as any attached button modules.

The 9601 phone does not support the large text size feature.

Before you begin

Ensure that the file `Mlf_Englarge.xml` resides on the web root of the HTTP Server.

Procedure

Modify the settings file to set the LANGLARGEFONT parameter as follows:

```
SET LANGLARGEFONT "Mlf_Englarge.xml"
```

Setting the background logo

About this task

Use the following procedure to set a customized background logo. The phone models 9601, 9608, and 9608G do not support a background logo.

Before you begin

Ensure that you have set the parameter LOGOS in the settings file.

Procedure

Modify the settings file to set the CURRENT_LOGO parameter to one of the labels specified in the LOGOS parameter.

When the logo file is not configured correctly or the HTTP server containing the logo file is not reachable, the phone might be temporarily unavailable.

In Avaya Aura® System Manager6.3.8 or later, you cannot use this parameter if you are using the Expanded Template support for SIP Endpoints feature.

Background logo specifications

The following table specifies the information on the maximum size, color depth and format of a logo supported by the phone models 9611G, 9621G and 9641G. The phone models 9601, 9608, and 9608G do not support a background logo.

Models	Maximum size (pixels)	Color depth (bit)	Format
9611G	217 x 130	16	JPG
9621G	232 x 140	16	JPG
9641G	232 x 140	16	JPG

WML browser overview

WML browser feature comparison

The phones have an built-in WML browser application. Users can see the browser application listed in the **Home** screen of Avaya menu, if the related system parameters are configured in the `46xxsettings.txt` file.

The following table shows a comparison of different features of the WML browser across the phone models.

Feature	9601	9608/9608G	9611G	9621G	9641G/9641GS
Top line	Yes	Yes	Yes	Yes	Yes
Application lines	2	4	4	4	5
Line buttons	Yes	Yes	Yes	No	No
Selectable objects per line	1	1	1	1	1

Table continues…

Feature	9601	9608/9608G	9611G	9621G	9641G/9641GS
Application line height (in pixels)	14	15	31	38	38
Softkeys per screen	3	4	4	5	5
Softkey height (in pixels)	14	14	31	38	38
Navigation buttons	Yes	Yes	Yes	No	No
Text input	Yes	Yes	Yes	Yes	Yes
Color support	No	No	Yes	Yes	Yes
Supported image format	No	JPEG	JPEG	JPEG	JPEG
Max image width (in pixels)	NA	175	300	430	430
Click to dial	Yes	Yes	Yes	Yes	Yes
Add to phonebook	Yes	Yes	Yes	Yes	Yes
Characters per line (normal font)	26	31	40	39	39
Characters per line (large font)	NA	25	22	26	26
Characters per softkey (normal font)	8	8	8	8	8
Characters per softkey (large font)	NA	6	6	8	8

Microsoft Exchange Server integration

Phones can connect with Microsoft Exchange Server. With this connection, users can do the following:

- View calendar reminders to make a call that users can dismiss or snooze. Users can also use the reminders to make a call.
- Dial into conference calls without entering a conference call number and access code.
- View exchange contacts.

Note:

Exchange Web Services is supported for Microsoft Exchange Server 2010 or later.

You can secure the link between the phones and the Exchange server. The phone adds the Exchange contacts of a user to the contact list and displays it in the Exchange Contacts screen. Users can save the Exchange contacts to PPM using the +Local key. To gain access to the Exchange contacts, calendar, and reminder information, users must use Avaya Menu to specify the following information:

- Credentials
- Display preferences
- Calender date

Users can modify or delete contacts that are saved in PPM. Avaya SIP phones do not support the presence tracking feature for the Exchange contacts.

Microsoft™ Exchange parameters

Parameter name	Default value	Description
PROVIDE_EXCHANGE_CALENDAR	1	Specifies whether menu items for Exchange Calendar are displayed. Value operation: • 0: Not displayed • 1: Displayed
PROVIDE_EXCHANGE_CONTACTS	1	Specifies whether menu items for Exchange Contacts are displayed. Value operation: • 0: Not displayed • 1: Displayed
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_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.

Table continues…

Parameter name	Default value	Description
		You can change the value by using the Admin menu on the phone.
ENABLE_EXCHANGE_REMINDER	0	Specifies whether or not exchange reminders will be displayed. Value Operation: • 0: Not displayed • 1: Displayed
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_SNOOZE_TIME	5	Specifies the number of minutes in which a reminder must be displayed again after it is temporarily dismissed. Valid values are 0 through 60.
EXCHANGE_NOTIFY_SUBSCRIPTION_PERIOD	180	Specifies the number of seconds between re-syncs with the Exchange server. Valid values are 60 through 3600.
EXCHANGE_EMAIL_DOMAIN	Null	Specifies the Exchange email domain. The value can contain 0 to 255 characters.
EXCHANGE_SERVER_SECURE_MODE	1	Specifies if HTTPS should be used to contact Exchange servers. Value Operation • 0: Use HTTP • 1: Use HTTPS
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 (default)

Resetting system values

About this task

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

Caution:

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

Procedure

1. Select RESET VALUES from the screen. The phone displays the following text:

```
Are you sure you want to reset the phone's initialization values?
```

2. Press **No** to return to the screen without resetting the phone.

   Press **Yes** to start the phone reset.

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

   - All system values and system initialization values except AUTH and AUTH_ONLY to default values.
   - The 802.1X ID and Password to their default values.
   - Call server values to their defaults.
   - Any entries in the Redial buffer.
   - Does not affect user-specified data and settings like Contacts data or the phone login and password.

Clearing the phone settings

About this task

Sometimes, you might want to remove *all* administered values, user-specified data, and option settings and return a phone to its factory settings. You might have to remove all administered values when you give a phone to a new, dedicated user and when the **LOGOFF** option is not sufficient. For example, a new user is assigned the same extension, but requires different permissions than the previous user.

The **CLEAR** option erases all administered data—static programming, HTTP and HTTPS server programming, and user settings including Contact button labels and locally programmed Feature button labels, and restores all such data to default values. Using the **CLEAR** option does not affect:

- The software load. If you upgrade the phone, the phone retains the latest software. After you clear a phone of the settings, you can administer the phone normally.

Caution:

This procedure erases all administered data without any possibility of recovering the data. Neither the boot code nor the application code is affected by this procedure.

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

Procedure

1. **CLEAR** from the menu.

   The phone displays the message,

   ```
   Are you sure you want to reset the phone to factory defaults? This operation
   cannot be undone.
   ```

2. Tap **Yes** to clear all values to use initial default values.

   Tap **No**. If you do not want to clear all values and to terminate the procedure and retain the current values.

   The phone displays the following text:

   ```
   Clearing values...
   ```

   The phone is reset to the default factory settings.

   - All system values and system initialization values.
   - 802.1X identity and password.
   - User options, parameter settings, identifiers, and password.

   After clearing the values, the phone resets.

Restarting the phone

About this task

Use the following procedure to restart the phone.

Procedure

1. Use the CRAFT password to gain access to the Administration procedures screen. The default password is *27238*.
2. When you select **RESTART PHONE** from the Administration procedures screen, the phone displays a confirmation screen with the following message:

   ```
   Are you sure you want to restart the phone?
   ```

3. Press **No** to return to the Administration procedures screen without restarting the phone. Press **Yes** to proceed with the registration steps

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

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

Chapter 6: Failover and survivability

Redundancy with IP phone and Avaya Aura®

Avaya IP phones and Avaya Aura® Communication Manager can be configured to provide optimal redundancy support. The phones can be configured to register simultaneously with the following:

- Two Avaya Aura® Session Manager SIP proxies
- Two Session Manager instances and one Branch Session Manager
- One Session Manager and one Branch Session Manager

If the connection is lost to the primary Session Manager, the phone establishes communications with the second Session Manager. Similarly, if the second Session Manager is unavailable, then the phone establishes communication with the third Session Manager. The third Session Manager can only be a Branch Session Manager.

Alternatively, a non-Avaya Aura proxy can be used as a survivable proxy. In this case, when the connection is lost between the phone and the Session Manager, the phone again registers with the non-Avaya Aura proxy and attempts to continue the service with little disruption. The two possible non-Avaya Aura configurations are as follows:

- One Session Manager and one non-Avaya Aura proxy
- Two Session Manager instances and one non-Avaya Aura proxy

If connection between a phone and Session Manager is lost during a call, then the phone attempts to preserve the call by sustaining the audio path between the two parties. This is called call preservation. In spite of this best effort service, the audio path might be lost. Further, in a preserved call, the phone does not support call features, for example, conference call, call transfer, and call forward.

Detection of loss of connection

The three methods to detect a loss of connection between the phone and the SIP proxy are as follows:

- Loss of TCP connection between the phone and the SIP proxy: If the TCP socket closes, or if the TCP keep alive timer times out, then there is a loss of connection. The TCP keep alive timer is set to a default value of 45 seconds but can be modified by using the TCP_KEEP_ALIVE_TIME parameter in the `46xxsettings.txt` file.

- Failure of the proxy to respond to a SIP INVITE message within a specified time: If the phone sends a SIP INVITE message to the proxy and the proxy does not reply within a specified time, then there is a loss of connection. The response time is set to a default value of 5 seconds in the `46xxsettings.txt` file but can be modified by using the FAST_RESPONSE_TIMEOUT parameter. TheAvaya Aura® System Manager parameter, TIMER B, takes precedence over the `46xxsettings.txt` file parameter.
- Failure of the proxy to respond to a SIP registration method: After the initial registration, the phone sends a re-registration message periodically to the proxy. If the proxy fails to respond to the re-registration message, the phone starts a failover. The parameter REGISERWAIT in the `46xxsettings.txt` file defines the period of re-registration. However, the Avaya Aura® System Manager parameter Registration Expiration Time takes precedence over the `46xxsettings.txt` file parameter.

Failover to a backup proxy

When a loss of connection occurs, the phone continues the service with the secondary Session Manager. If the secondary Session Manager is unavailable, the phone uses the survivable proxy.

Restoring the phone to the primary proxy

When the link between the phone and the primary Session Manager is restored, the phone might re-establish communication and revert to the primary Session Manager. This process is referred to as failback.

After a failover occurs, the phone waits for a period of time defined by the RECOVERYREGISTERWAIT parameter and then the phone attempts to register back to the primary proxy. You can modify the time in the Reactive Monitoring parameter on System Manager. This parameter takes precedence over the `46xxsettings.txt` file parameter. After this timer expires, the phone attempts to connect to the primary Session Manager. If the attempt is successful, the phone sends a new SIP registration message to the Primary Session Manager. At this point, another timer starts that is defined by the parameter WAIT_FOR_REGISTRATION_TIMER. If there is no response to the registration message from the proxy by the time it expires, then it waits for the RECOVERYREGISTERWAIT time.

This process maps to the `46xxsettings.txt` file parameter FAILBACK_POLICY being set to automatic. If the parameter is set to manual, then the administrator must send a message to the phone through System Manager to force it to re-register with the primary Session Manager.

Proxy determination when the connection to the primary proxy is lost

A list of all proxies is provided to the phone during initial configuration. This list serves two purposes:

- Specifies the SIP proxies that are used by the phone.
- Prioritizes the list of proxies into primary, secondary, and survivable proxies.

Initially, DHCP, LLDP, or the `46xxsettings.txt` file provides this list of prioritized proxies. After the phone connects to Session Manager, it receives a new prioritized list of proxies specified by System Manager. This list takes precedence over other sources. The list provided by System Manager is derived from the following three fields:

- **Primary Session Manager**
- **Secondary Session Manager**
- **Survivability server**

When a phone detects a loss of connection with the primary proxy, the phone fails over to the secondary proxy. If both the primary and secondary proxies are unreachable, then the phone fails over to the survivable proxy.

Simultaneous registration

Phones can register simultaneously with more than one proxy. This makes the method of redundancy quick and deterministic. While configuring the phones for redundancy with Avaya Aura®, set the parameter SIPREGPROXYPOLICY to Simultaneous. In fact, when the phone registers for the first time, the parameter SIPREGPROXYPOLICY is forced to simultaneously register. Also, you can use the parameter SIMULTANEOUS_REGISTRATIONS to specify the number of proxies required to support simultaneous registration.

Note:

All Session Manager and Branch Session Manager instances support simultaneous registration while non-Avaya Aura proxies do not support simultaneous registration. For example, if your configuration is two Session Manager instances and a non-Avaya Aura proxy, then the value of SIMULTANEOUS_REGISTRATIONS is 2.

Limitations during failover or failback

Limitations of the phone when the phone is in the process of failover or failback are as follows:

- Held calls are dropped.

- Calls that are in the middle of the conferencing or transfer set up are dropped.
- Calls in the dialing or ringing state might not be completed.
- Emergency calls might not work depending on the stage of failover and the functionality available on the alternate server.
- Incoming calls might not be completed, or they might get diverted to voicemail.
- Message Waiting Indicator is cleared.

Preserved call

When there is a call in progress and a loss of connection occurs between the phone and the proxy, an attempt is made to preserve the audio path between the phone and the far end. This is called Call preservation. In most cases, call preservation is successful. However, there are conditions when the audio path is lost. This loss of audio might happen when there is no direct path between the phone and the far end. The entity that connects the media between the two ends is also affected by the loss. Further, there are limitations to modify a preserved call.

Limitations of a preserved call

In a successful call preservation, the audio path is preserved. However, in a preserved call, the phone does not support call features, for example, conference call, call transfer, and call forward . The reason is loss of signaling between the phone and the SIP proxy that was used when the call was initially established.

The loss of signalling between the phone and the originating proxy also limits the call control between the preserved calling parties. For example,

- If one of the calling parties disconnects from a preserved call, the other end might not be disconnected.
- Communication Manager cannot perform mid-call operations, such as MLPP features, when the target phone is in a connection preserved state.

Supported non Avaya Aura® proxies for redundancy

The supported non Avaya Aura® proxies for redundancy are as follows:

- Avaya Secure Router 2330 and 4134
- Avaya IP Office
- Audiocodes MediaPack™ 11x series and Mediant™ series gateways

Limitations after a successful failover

Failover to a Session Manager

instance

After a phone successfully fails over to a secondary Session Manager, all features and functionality work properly for new calls. However, there are limitations to modify a preserved call.

Failover to a Branch Session Manager

After a phone successfully fails over to Branch Session Manager, the value of the parameter FAILBACK_POLICY changes to Admin. In this case, you must go to the System Manager and manually re-register the phone with Session Manager.

Note:

Administration of Session Manager and Branch Session Manager nodes are explicitly required in the System Manager user record.

Failover to a proxy other than Avaya Aura®

The limitations after a phone fails over to a proxy other than Avaya Aura® are:

- A conference is limited to three parties and is hosted by the phone.
- Contacts can be used and new contacts can be saved on the phone. New contacts are cached on the phone, and after failback to Avaya Aura®, the new contacts are synchronized with Avaya Aura®.
- The dial plan for Avaya Aura® is unavailable. Instead, the dial plan configured in the `46xxsettings.txt` file is used.
- The following Avaya Aura® features are unavailable:
  - Last party drop
  - Send All Calls (Do Not Disturb)
  - Presence
  - Calling party block/unblock
  - Call park/unpark
  - All forms of call pickup
  - Priority calls
  - MLPP functionality
  - Auto callback
  - Malicious call trace
  - EC500 on/off
  - Transfer to voicemail
  - Paging
  - Call recording

- Bridge Line Appearance
- Extend call
- Hold recall
- Transfer recall
- Busy Indicator
- Message Waiting Indicator
- Team button
- Call Center Elite

Indications of redundancy

The following indications are given to the user when the phone has connection issues:

Acquiring service

When a phone does not have a communication channel established with any SIP proxy and a call is in progress, then the phone displays the `Limited Phone Service` message. The message either disappears by itself or can be cancelled by the user. Also, an icon indicating Acquiring Service is displayed on the top line of the phone. This icon does not go away until a communication channel is established with a SIP proxy. The icon is in the form of an exclamation mark within a triangle similar to the following:

If there is no ongoing call and there is no communication channel between the phone and the proxy, then the phone displays the message `Acquiring Service`.

Preserved call

When a failover occurs and a call is preserved, the call appearance line of the phone displays the following preserved call Indicator: :

Parameters for redundancy provisioning

SIP connection parameters

Parameter name	Default value	Description	System Manager parameter name
CONTROLLER_SEARCH_INTERVAL	16	Specifies the number of seconds the phone will wait to complete the maintenance check for monitored controllers.	NA

Table continues…

Parameter name	Default value	Description	System Manager parameter name
		Valid values are from 4 to 3600.	
DISCOVER_AVAYA_ENVIRONMENT	1	Specifies dynamic feature set discovery. Value operation are: • 0: Non-Avaya environment. Does not auto-discover Avaya SIP Telephony (AST) support . • 1: Avaya environment. Auto-discovers AST support. The SIP proxy server or controller might not support AST.	NA
FAILBACK_POLICY		Specifies the policy in effect for recovery from failover. Value operation: • Admin: The phone waits for administrative intervention before failing back to a higher priority controller. • Auto: The phone periodically checks the availability of the primary controller and fails back to the primary controller if available.	Failback Policy The value set in System Manager overwrites the value in the `46xxsettings.txt` file.
FAST_RESPONSE_TIMEOUT	4	Specifies the number of seconds the phone will wait before terminating an invite transaction if no response is received. Valid values are from 0 to 32. Value operation: • 0: Timer is disabled	Timer B This parameter is mandatory in System Manager and the default value is 2 seconds. The value set in System Manager overwrites the value in the `46xxsettings.txt` file.
RECOVERYREGISTERWAIT	60	Specifies the number of seconds. If no response is received by WAIT_FOR_REGISTRATION_TIMER to a REGISTER request within the specified number of seconds, the phone tries again after a randomly selected delay	Reactive Monitoring

Table continues…

Parameter name	Default value	Description	System Manager parameter name
		of 50% to 90% of the value of RECOVERYREGISTERWAIT. Valid values are from 10 to 36000.	
REGISTERWAIT	900	Specifies the number of seconds for next re-registration to the SIP proxy. Valid values are from 30 to 86400 seconds.	Registration Expiry Timer The value set in System Manager overwrites the value in the`46xxsettings.txt` file.
WAIT_FOR_REGISTRATION_TIMER	32	Specifies the number of seconds the phone will wait for a response to a REGISTER request. If no response message is received within this time, the phone tries to register again based on the value of RECOVERYREGISTERWAIT. Valid values are from 4 to 3600.	NA
SIP_CONTROLLER_LIST	Null	Specifies a list of SIP controller designators, separated by commas without any intervening spaces. When this parameter has multiple IP addresses, the list order defines the priority of the controllers for selection during a failover. The first element of the list has the highest priority, and the last element has the lowest priority.	Primary Session Manager, Secondary Session Manager and Survivability server
ENABLE_PPM_SOURCED_SIPPROXYSRVR	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 used.	NA
SIP_CONTROLLER_LIST_2	Null	Replaces SIP_CONTROLLER_LIST for IPv4 and IPv6 phones. It is used	Primary Session Manager, Secondary Session Manager, and Survivability server.

Table continues…

Parameter name	Default value	Description	System Manager parameter name
		to select the registration address.	
SIMULTANEOUS_REGISTRATIONS	3	Specifies the number of simultaneous Session Manager and Branch Session Manager registrations that the phone must maintain. Valid values are from 1 to 3. The value of this parameter must not be less than the number of core Session Manager instances in SIP_CONTROLLER_LIST.	NA
SIPREGPROXYPOLICY	alternate	Specifies whether the telephone will attempt to maintain one or multiple simultaneous registrations. Value operation: • Alternate: The phone registers only to the first controller in the list. If the phone cannot reach the first controller, the phone registers to the second controller . • Simultaneous: The phone simultaneously registers to more than one SIP proxy controller at the same time.	NA

The primary, secondary, and survivable server settings for a phone must be configured in System Manager. This enables the phone to access the full list of assigned servers after the phone logs in. You must provide at least one primary and secondary server to the phone to make the initial login connection. You can provide the servers by using DHCP, LLDP, or the `46xxsettings.txt` file parameters SIP_CONTROLLER_LIST or SIP_CONTROLLER_LIST_2. Ideally, the full list of servers must be provided. However, when a survivable server is location specific, you must only include the survivable server in DHCP, LLDP or the `46xxsettings.txt` file if the correct survivable server for the location can be provided. This ensures that the phone always receives the correct survivable server address. A DHCP server local to a branch is one such method in which this could be done. However, if you cannot provide the correct location-specific survivable server reliably in DHCP, LLDP, or the `46xxsettings.txt` file, then you must not include it. In this case, the phone gains access to it after login.

Dial Plan parameters for use when failing over to a proxy other than Avaya Aura

Parameter name	Default value	Description
ENABLE_REMOVE_PSTN_ACCESS_PREFIX	0	Enables the removal of the PSTN access prefix from the collected dial strings when the phone communicates with a non-AST controller. Value operation: • 0: PSTN access prefix digit is not removed. • 1: PSTN access prefix digit is removed from the collected digit string before formulating the INVITE for delivery to the controller. The parameter has no effect if you enable this parameter when the phone communicates with an AST-capable controller.
PSTN_VM_NUM	Null	Specifies a phone number or Feature Access Code to be used by the messaging application in a non-Avaya or failover server environment. This dialable string is used to call into the messaging system, for example, when you press the Message Waiting button.
INTER_DIGIT_TIMEOUT	5	Specifies the timeout that takes place when a user stops entering digits. The timeout is treated as digit collection completion, and when it occurs, the application sends out an invite. Valid values are from 1 to 10.
ENABLE_REMOVE_PSTN_ACCESS_PREFIX		Enables the phone to perform digit manipulation during failure scenarios. This parameter enables removal of the PSTN access prefix from the outgoing number. Value operation: • 0: PSTN access prefix is retained in the outgoing number • 1: PSTN access prefix is stripped from the outgoing number.
PHNLAC		Indicates the local area code of the phone. . PHNLAC is a string that enables users to dial local numbers with more flexibility when used together with the LOCAL_DIAL_AREA_CODE parameter .
LOCAL_DIAL_AREA_CODE		Specifies whether a user must dial the area code for calls within the same area code regions. Value operation: • 0: Users do not need to dial an area code.

Table continues…

Parameter name	Default value	Description
		• 1: Users need to dial an area code.
DIALPLAN	Null	Specifies the dial plan used in the phone. It accelerates dialing by eliminating the need to wait for the INTER_DIGIT_TIMEOUT timer to expire.

Redundancy in a non-Avaya proxy environment

In an Avaya environment, the SIP proxy list is defined using dotted decimal notation to define the proxy addresses. In a non-Avaya environment, where FQDNs are used to define the SIP proxy, there can only be one proxy. In this case, redundancy is not supported.

Chapter 7: Backup and restore

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.

User profile parameters for backup

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

Parameter	Default value	Description
BAKLIGHTOFF	120	Specifies the timer to switch off the backlight of the display.
CLICKS	1	Specifies if the phone button can generate click sounds.
CALL_PICKUP_RING_TYPE	1	Specifies the default call pickup ring type.
OUTSIDE_CALL_RING_TYPE	1	Specifies the default outside call ring type.
FORWARDED_CALL_RING_TYPE	1	Specifies the default forwarded ring type that the user selects.
CALL_PICKUP_INDICATION	3	Specifies the following call pickup indication types: • Audio • Visual • None
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.
CONTACT_NAME_DISPLAY	1	Specifies how contact names are displayed.

Table continues…

Parameter	Default value	Description
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.
AUDIOPATH	1	Specifies whether the default audio path is speaker or headset.

Chapter 8: Phone upgrade

Device upgrade

Before upgrading the device, ensure that you download the latest software, the distribution package and the settings file, on the file server. You can perform the device upgrade in the following ways:

- Automatic: You can configure the device to poll periodically for a newer version of the software in the file server and automatically download the software and upgrade itself.
- Manual: You can upgrade the device without the device waiting for a polling interval by:
  - Using the update option in the Settings app on the device. With the update option, the device immediately downloads and installs the software if an updated version is available.
  - Rebooting the device from the Settings app or from System Manager. With rebooting, the device might upgrade immediately or later based on the upgrade policy configured for the device.

Device upgrade process

The upgrade event is logged under NOTICES level in the Syslog file. During boot up, the 9600 Series IP Deskphones performs the following tasks:

1. The phone receives the file server address from DHCP, LLDP, or the device interface.
2. The phone connects to the file server and searches for the upgrade file depending on the SIG parameter value.
   - 0: Default, `96x1Supgrade.txt`
   - 1: H.323, `96x1Hupgrade.txt`
   - 2: SIP, `96x1Supgrade.txt`
3. The phone compares its software version with the version specified in the upgrade file.
   - For Sonic and other hardware phones, HWVER value is also checked when the MODEL4 match is found.

4. The phone then downloads the upgrade file for parsing. The parameter UPGRADE_FILE_EXECUTION_STATUS is updated with the following values upon parsing:
   - 0: Upgrade file is downloaded and parsed.
   - 1: Upgrade file is downloaded but not parsed.
   - 2: Upgrade file is not downloaded and not parsed.
5. The upgrade gets triggered depending on the parameter UPGRADE_FILE_EXECUTION_STATUS value.
6. The phone starts downloading files depending on the parameter APPNAME value contained in the upgrade file.

   Tip:

   The software files contain three binaries BootA, BootB, and System RFS. Each binary contains a version number and a signature header which is processed in sequence and is stored in the appropriate flash memory location.
7. The phone downloads the software files and upgrades itself if no fatal error occurs.

   Tip:

   Fatal error occurs when the file size is too large, missing signature or if the signature validation fails, file is not found, file download failure, fails to write to the flash memory, file is incompatible with the hardware, or during any parsing error.

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 `9600 Series IP Deskphones`.
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.

Manual upgrade

About this task

Use the Avaya-provided upgrade script files and the application files that are included in the zip files to upgrade the phones. Ensure that all the files are together on the file server. Do not modify the files. Use this procedure to download the latest version of the software to the file server.

Procedure

1. Stop the file server.
2. Specify the port settings for HTTP or TLS in the HTTPPORT or TLSPORT settings respectively.
3. Perform a back up of all the current file server directories.
4. Copy the `46xxsettings.txt file` to a backup location.
5. Remove all the files in the download directory. This ensures that you do not have an inappropriate binary or configuration file on the server. The only system values that can be used in the conditional statement are: BOOTNAME, GROUP, MACADDR, MODEL, and SIG.
6. Download the self-extracting executable file or the corresponding zip file.
7. Extract all the files.
8. Copy the `46xxsettings.txt file` to the download directory.
9. Check the `Readme` file for release-specific information.
10. Modify the `46xxsettings.txt file` as required.
11. Restart the HTTP/HTTPS server.
12. Reset the phone.

Downloading text language files

Language files contain the language name (as it should be presented to a user for selection), an indication of the preferred character input method, text string replacements for the built-in English text strings, where each replacement string may contain up to 120 characters. Each has a unique index that associates it with the corresponding built-in English string, an indication as to whether

Chinese, Japanese or Korean glyphs should be displayed for the Unicode "Unified Han" character codes, and a Language Identification Tag for the language of the text contained in the file. A downloadable language file may also contain a translation of the language name into any or all of the languages for which a language file is included in the software distribution package. Language files must be stored in the same location as the 46xxsettings.txt file.

You can download a new language file version only if the filename differs from the language file previously downloaded. Alternately, you can remove the old language file using an empty **SET LANGUAGES** command in the 46xxsettings file before downloading a language file with the same filename.

Note:

Language files for SIP deskphones have a **.xml** filename extension whereas language files for 9608, 9608G, 9611G, 9621G, or 9641G IP deskphones set to H.323 have a .txt filename extension.

Changing the signaling protocol

About this task

For enterprises requiring both H.323-based and SIP-based protocols, there are two ways to specify the protocol to be used by all or specific deskphones:

Procedure

1. The SIG parameter can be set in DHCP Option 242 (Site-Specific Option Number) or in the 46xxsettings.txt file. This setting will apply to all deskphones except those for which SIG has been manually configured to a value of H.323 or SIP using the SIG Craft procedure.
2. The SIG parameter can be set on each phone.

   The 9601 deskphone supports only the SIP signaling.

The GROUP parameter

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

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

Then, at each non-default deskphone, invoke the **GROUP** Local (Craft) Administrative procedure and specify which GROUP number to use. Once the GROUP assignments are in place, edit the configuration file to allow each deskphone of the appropriate group to download its proper settings.

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

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

Chapter 9: 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*

Error conditions

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

- Check both the power and Ethernet wiring for the following conditions:
  - Whether all components are plugged in correctly.
  - Check LAN connectivity in both directions to all servers - DHCP, HTTP, HTTPS, Avaya Communication Manager, and/or SIP Proxy server.
  - If the deskphone is supposed to be powered from the LAN, ensure that the LAN is properly administered and is compliant with IEEE 803.3af.

- If you are using static addressing:
  - Use the **VIEW** Craft procedure to find the names of the files being used and verify that these filenames match those on the HTTP/HTTPS server.
  - Use the **ADDR** Craft procedure to verify IP Addresses.
- If the 9600 Series IP Deskphones are not communicating with the system (DHCP, HTTP, or Communication Manager call server), make a note of the last message displayed. Consult the system administrator.
- If you expect the deskphone to be IEEE-powered, verify with the LAN administrator that IEEE power is indeed supported on the LAN.

Related links

DTMF tones on page 132
Power interruption on page 132

DTMF tones

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

Related links

Error conditions on page 131

Power interruption

If power to 9600 Series IP Deskphone is interrupted while the deskphone is saving the application file, the HTTP/HTTPS application can stop responding. If this occurs, restart the phone.

Related links

Error conditions on page 131

Installation error and status messages

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

Most of the messages described in the table appears only for about 30 seconds or less, and then the deskphone resets. The most common exception is

```
Extension in Use
```

, which requires manual intervention.

Possible error and status messages during installation of 9600 Series IP Deskphones

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

Operational errors and status messages

The tables described identifies some of the possible operational problems that might be encountered after successful installation of 9600 Series IP Deskphones. The user guide for a specific deskphone model also contains troubleshooting for users having problems with specific deskphone applications.

Possible operational error conditions for 9600 Series IP Deskphones

Condition	Cause/Resolution	
During Craft procedure access, display freezes at prompt "Press * to program"	Cause: Craft access has failed; deskphone cannot operate. Resolution: Unplug the deskphone, then plug it in again to reset.	
After Login, the progress bar shows just a few completed bars and stops moving.	Cause: Login has failed. Resolution: Check that the LAN and File servers are operating correctly. Re-attempt login.	
The message light on the deskphone turns on and off intermittently, but the deskphone never registers.	Cause: This is a hardware fault. Resolution: The deskphone must be returned to Avaya for repair.	
The deskphone stops working in the middle of a call.	No lights are lit on the deskphone and the display is not lit.	Cause: Loss of power. Resolution: Check the connections between the deskphone, the power supply, and the power jack.
	Deskphone might have gone through the restarting sequence.	Cause: Loss of path to the call server or the other party's deskphone, DHCP Lease expired, or DHCP server not available when deskphone attempts to renegotiate DHCP lease. Resolution: Check the connections between the deskphone, the power supply, and the power jack.
The deskphone was working, but does not work now.	No lights are lit on the deskphone and the display is not lit.	Cause: Loss of power. Resolution: Check the connections between the deskphone, the power supply, and the power jack.
	Power to the deskphone is fine, but there is no dial tone or the call appearances or feature buttons do not work.	Cause: Loss of communication with the call server. Resolution: Check LAN continuity from the call server to the deskphone using ARP or trace-route and from the deskphone to the call server by invoking a

Table continues…

		Feature button. Verify that administration has not changed for the LAN equipment (routers, servers, etc.) between the call server and the deskphone. Verify no one changed the deskphone settings locally using the **View** and **ADDR** craft procedures, as described earlier in this guide.
	The deskphone was recently moved.	Cause: Loss of communication with the call server. Resolution: As above, but pay particular attention to the possibility that the deskphone is being routed to a different DHCP server, or even a different proxy server. If so, the new server might need to be administered to support the deskphone.
	The network was recently changed to upgrade or replace servers, re-administer the Communication Manager call server, add or change NAT, etc.	Cause: Loss of communication with Session Manager. Resolution: As above.
The deskphone works, but the audio quality is poor.	The user hears echo when speaking on a handset.	Cause: Echo from digital-to-analog conversion on your Communication Manager call server trunk. Resolution 1: Try a different Call Quality setting under the Audio Parameters section. Resolution 2: Check whether packet loss, or jitter delay is causing this problem, by eliminating or minimizing both. Resolution 3: Verify which trunk is causing the echo, and check the trunk's Trunk Termination parameter on the call server.
	The user hears echo on a headset, but not on a handset.	Cause: Improper headset adapter. Resolution: Replace adapter with Avaya's M12LU or 3412-HIC adapters. We recommend the M12LU, since it supports Automatic Gain Control.

Table continues…

	The user is on Speaker and hears no echo, but the far-end hears echo.	Cause: Room acoustics. Resolution: Ensure that there are six inches or so of blank space to the right of the deskphone. If that is insufficient, use the handset.
	the user experiences sudden silences such as gaps in speech, or static, clipped or garbled speech, etc.	Cause: Jitter, delay, dropped packets, etc. Resolution: You can have the user provide diagnostic data by invoking the Network Information feature under the **A** (Avaya) button on the deskphone. One or more Quality of Service (QoS) features should be implemented in the network. Cause: Improper non-Category 5 wiring. Resolution: Replace non-Category 5 wiring with Category 5 wiring.
	The user hears fluctuations in the volume level which are worse when the Speaker is on, or at the beginning of a call, or when a call goes from no one talking abruptly to a loud voice.	Cause: The user has changed the Automatic Gain Control (AGC) or environmental acoustics are not consistent with the current audio settings. Resolution: Try different on/off settings for the AGCHAND, AGCHEAD, and AGCSPKR parameters.
The deskphone works properly except for the Speaker.	Cause: The Speaker was disabled in the settings file. Resolution: Check the settings file and re-enable the Speaker if appropriate.	
The deskphone works properly, except incoming DTMF tones are not received.	Cause: The TN2302AP board does not pass in-band DTMF tones. Resolution: None; the board is operating as designed.	
When a line is selected, a short dial tone burst sounds followed by a reorder/fast busy tone.	Cause: The extension is provisioned on Session Manager and some Communication Manager forms, but not on the off-pbx-telephone station-mapping form. Communication Manager is unable to map back to Session Manager, and rejects the line reservation. Resolution: Map the extension on the off-pbx-telephone station-mapping form. Cause: Possible error in SIG group configuration on Communication Manager, which indicates the default region for the SIP trunk to Communication Manager.	

Table continues…

	Resolution: On the IP-network-region form, ensure that the region pointed to is configured with an **authoritative domain** that is the same as the Session Manager SIP domain. also verify that the station in question has not been redirected to a different network region on the ip-network map.	
The HTTP/HTTPS script file and settings file are ignored (not being used by the deskphone).	Cause: The system value AUTH is set to 1 (HTTPS required) but no valid address is specified in TLSSRVR. Resolution: Change AUTH to 0 (zero), or enter a valid address for TLSSRVR.	
The HTTP/HTTPS script file is ignored or not used by the deskphone.	The HTTP/ HTTPS server is a LINUX or UNIX system.	Cause: UNIX and LINUX systems use case-sensitive addressing and file labels. Resolution: Verify the file names and path in the script file are accurately specified.
	The deskphone administration recently changed.	Cause: The ***96xxSupgrade.txt*** file was edited incorrectly, renamed, etc. Resolution: Download a clean copy of the ***96xxSupgrade.txt*** file from the Avaya support siteand do not edit or rename it. Customize or change only the ***46xxsettings*** file.
The MS Exchange contacts take too long to load	Cause: The correct Exchange server is not specified in the parameter EXCHANGE_SERVER_LIST in the ***46xxsettings*** file. Resolution: Verify that the MS Exchange server being used is specified in the settings file. To view the Exchange server in use, go to: **Outlook** > **Tools>Options** > **Mail Setup** > **E-mail Accounts** > **Change** .	
Some settings in the settings file are being ignored while other settings are being used properly.	Cause: Improper settings file administration. Resolution: Verify that customized settings are correctly spelled and formatted.	
	The setting being ignored is one or more of the AGC settings.	Cause: The user changed the AGC setting(s). Resolution: Have the user reset the AGC value(s) back to the desired setting(s).
	The setting being ignored is the TIMEFORMAT setting.	Cause: The time format was changed using the Avaya Menu Options & Settings. Resolution: If the time disappears, Reboot the phone.
Deskphone power is interrupted while the deskphone is saving the	Cause: The HTTP/HTTPS application stops responding if power is interrupted while a deskphone is saving the application file.	

Table continues…

application file and the HTTP/ HTTPS application stops responding.	Resolution: Restart the phone.
The user indicates an application or option is not available.	Cause: The ***46xxsettings*** script file is not pointed to accurately, or is not properly administered to allow the application. Resolution: Assuming the user is meant to have that application, verify the 46xxsettings script file is properly specified for your system, including case if your file server is UNIX or LINUX, and extension. Then, verify all the relevant parameters.
User data disappeared when the user logged off one deskphone and logged into another deskphone.	Cause: Possible PPM problem. Resolution: Contact the Session Manager administrator.
The deskphone displays "User logged in at another location".	Cause: The extension entered by the user during login is currently in use on another phone. Resolution: Instruct user to log in with a different extension. Tell the user to press the ‘Retry’ softkey, then enter new extension and password. Or, have the user log in with the original extension, while unregistered the extension from the other phone.
Login fails	Cause: Invalid provisioning on Communication Manager or Session Manager. Resolution: Session Manager needs to point to Communication Manager's PROCR interface for the "Media Server Admin Address." Session Manager must point to a specially-provisioned PPM Administration account on Communication Manager. The PPM Administration account on the Communication Manager side must have several specific parameters set. Specifically: login group must be "susers" additional group must be "prof18" or equivalent shell access must be "no shell access".
Multiple call appearances on incoming call.	Cause: Provisioning problem. Resolution: On the off-pbx-telephone station-mapping form, set the Bridged Calls field to "none".
A blank screensaver appears and the phone does not immediately respond to pressing the Phone button	Cause: The server IP Address in the LOGO parameter is invalid or unavailable. Resolution: Correct/change the LOGO parameter in the settings file.

SRTP provisioning

SRTP is now supported (with TLS). To use SRTP, the network region codec set must have media encryption set up for each region that calls may traverse.

When SRTP is provisioned in Communication Manager, the default cryptosuite used is ‘aescm128-hmac80’. The 9601, 9608, 9608G, 9611G, 9621G, 9641G, or 9641GS IP deskphone also assumes

that no encryption is an option provisioned in Communication Manager. If Communication Manager is provisioned with the cryptosuite aescm128-hmac80, then the following entry must be in the 46xxsettings.txt file:

SET MEDIAENCRYPTION "1,9"

If some other encryption set is required, the string must be set appropriately in the 46xxsettings.txt file.

Chapter 10: Resources

Documentation

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

Title	Use this document to:	Audience
Overview		
9600 Series IP Deskphones Overview and Specifications	See characteristics and capabilities, including feature descriptions, interoperability, performance specifications, security and licensing requirements of the 9600 Series IP Deskphones.	For people who want to gain a high-level understanding of the 9600 Series IP Deskphones features, functions, capacities, and limitations.
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.

Table continues…

Title	Use this document to:	Audience
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® 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 9601/9608/9608G/9611G/9621G/ 9641G/9641GS IP Deskphones SIP	See information about how to perform 9600 Series IP Deskphones 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 9600 Series IP Deskphones system administration tasks such as backing up and restoring data and managing users.
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		

Table continues…

Title	Use this document to:	Audience
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.
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. This parameter is not supported in IP Office environment as there is no subscription to Avaya-cm-feature-status.
AUDIOENV		Specifies the audio environment index and enables you to customize the phone's audio performance. Valid values are 0 through299. This parameter affects settings for AGC dynamic range, handset and headset noise reduction thresholds, and headset transmit gain. Always consult Avaya before changing this parameter.
AUDIOSTHD		Specifies the level of sidetone in the headset.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Normal level for most users • 1: One level softer than normal • 2: Two levels softer than normal • 3: Three levels softer than normal • 4: Off which means inaudible • 5: One level louder than normal
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.

Table continues…

Parameter name	Default value	Description
AUTO_UNMUTE	0	Specifies whether the call is not in a muted state when the transducer is changed. This feature is applied on all types of calls. Value Operation: • 0: Disabled. Call is in mute state. • 1: Enabled. Call is not in mute state.
B		
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.
BLUETOOTHSTAT	1	Specifies whether the user is given an option to enable the bluetooth. Value Operation: • 0: Bluetooth is disabled and the user is not given an option to enable it. • 1: The user is given an option to enable the bluetooth.
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
BUTTON_MAPPINGS	Null	Specifies a list of Button and Status pairs that change the operation of some of the buttons on the phone.

Table continues…

Parameter name	Default value	Description
		Button and Status pairs are separated by commas without any intervening spaces. Valid button values are Forward, Speaker, Hookswitch, and Headset. Valid Status values are na and cc-release. Value Operation: • na: The corresponding button is disabled. • cc-release: Button invokes the cc-release feature. • null: All buttons operate normally.
C		
CALL_TRANSFER_MODE	0	Determines the call transfer mode in 3rd party environments. Valid value is 0 or 1.
CALLFWDADDR	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		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	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.
CC_INFO_TIMER	8	Specifies the duration, in hours, of the subscription to the SIP CC-Info event package. Valid values are 1 through 24.
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.

Table continues…

Parameter name	Default value	Description
		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.
CLDISPCONTENT	1	Specifies whether the name, the number, or both will be displayed for Call Log entries. Value Operation • 0: Both the name and the number will be displayed. • 1: Only the name will be displayed.
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 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.

Table continues…

Parameter name	Default value	Description
		If the parameter is set to a value that is outside the range then default value is selected.
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. 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: • 1: Enabled. The phone does not open a listening socket and will maintain and re-use the sockets it creates with the outbound proxies.
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.

Table continues…

Parameter name	Default value	Description
		• 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. 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.

Table continues…

Parameter name	Default value	Description
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. 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

Table continues…

Parameter name	Default value	Description
		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.
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.

Table continues…

Parameter name	Default value	Description
		• 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.
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.

Table continues…

Parameter name	Default value	Description
		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. 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

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 ifr 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_MLPP	0	Specifies that whether the Multiple Level Precedence and Preemption (MLPP) is enabled or not. Value Operation: • 0: Disabled. • 1: Enabled.

Table continues…

Parameter name	Default value	Description
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.
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_PPM_SOURCED_SIPPROXYSRVR The parameter is only available in an Avaya Aura® environment.	1	Enables PPM as a source of SIP proxy server information.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: Proxy server information received from PPM is not used. • 1: Proxy server information received from PPM is not used.
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.

Table continues…

Parameter name	Default value	Description
		Value Operation; • 0: PSTN access prefix is retained in the outgoing number. • 1: PSTN access prefix is removed from the outgoing number.
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.

Table continues…

Parameter name	Default value	Description
		Value Operation: • 0: SRTCP is disabled. • 1: SRTCP is enabled.
ENFORCE_SIPS_URI	1	Specifies if a SIPS URI must be used for SRTP. 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.
EXCHANGE_AUTH_USERNAME_FORMAT	0	Specifies the necessary format of the username for http authentication. Value operation: • 0: Current format. Username= <ExchangeUserDomain\ExchangeUserAccount> or Username= <ExchangeUserAccount> if <ExchangeUserDomain> is empty. • 1: Office 365 format. Username= <ExchangeUserAccount@ExchangeUserDomain

Table continues…

Parameter name	Default value	Description
		> 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. Valid values are 0 through 60.

Table continues…

Parameter name	Default value	Description
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. You can change the value by using the Admin menu on the phone.
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.

Table continues…

Parameter name	Default value	Description
		The purpose of this parameter is to support cases where the server certificate Subject Common 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 craft procedure. The default value of GROUP in each phone is 0, and the maximum value is 999.
H		
HEADSET_PROFILE_DEFAULT	1	Specifies the number of the default headset audio profile. Valid values are 1 through 20.
HEADSET_PROFILE_NAMES	Null	Specifies an ordered list of names to be displayed for headset 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 is displayed for the corresponding profile. Names can contain spaces, but if any do, the entire list must be quoted. There is no way to prevent a profile from being displayed.

Table continues…

Parameter name	Default value	Description
HEADSYS	0	Specifies whether the phone goes on-hook if the headset is active when the disconnect message is received. Value Operation: • 0: The phone goes on-hook if the disconnect message is received when the headset is active. • 1: Disconnect messages are ignored when the headset is active. This is used for Call Center setting.
HOMEIDLETIME	10	Specifies the number of minutes of idle time after which the **Home** screen is displayed. Valid values are 0 through 30. A value of 0 means that the **Home** screen is not displayed automatically when the phone is idle.
HOTLINE	Null	Specifies zero or one hotline number. Valid values can contain up to 30 dialable characters ranging from 0 to 9, *, and #.
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

Table continues…

Parameter name	Default value	Description
		• DNS name The parameter can be set by using LLDP. Valid values contains 0 to 255 ASCII characters.
I		
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.

Table continues…

Parameter name	Default value	Description
		Value operation: • 0: IPv6 will not be supported. • 1: IPv6 will be supported.
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

Table continues…

Parameter name	Default value	Description
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. **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.

Table continues…

Parameter name	Default value	Description
		Value operation: • 0: Disabled • 1: Enabled • 2: Enabled, but only begins transmitting if an LLDP frame is received.
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.

Table continues…

Parameter name	Default value	Description
		• 7: Debug, Informational, Notice, Warning, Error, Critical, Alert and Emergency events are logged **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).
LOGOS		Specifies a list of tuples describing the logo or the wallpaper to be used as the phone display background. Only full path URLs are supported. For 9611G, 9621G and 9641G, the maximum size in pixels are: 217 x 130, 232 x 140 and 232 x 140 respectively with color depth 16 bit and JPG file type. GIF is presented without animation.

Table continues…

Parameter name	Default value	Description
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 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

Table continues…

Parameter name	Default value	Description
		• 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 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.
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
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 • DSN: DSN network. • UC: UC network.

Table continues…

Parameter name	Default value	Description
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.
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. 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.

Table continues…

Parameter name	Default value	Description
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.
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		

Table continues…

Parameter name	Default value	Description
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. 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.

Table continues…

Parameter name	Default value	Description
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).
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, *, #).

Table continues…

Parameter name	Default value	Description
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.
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. Valid values are from 0 to 10080. Value operation: • 0: Phone will not lock automatically.

Table continues…

Parameter name	Default value	Description
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 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.

Table continues…

Parameter name	Default value	Description
		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
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

Table continues…

Parameter name	Default value	Description
		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.
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.
PROCPSWD	27238	Specifies an access code to access the local (craft) procedures. Valid values contain 0 through 7 ASCII numeric digits. The default value is 27238 unless indicated

Table continues…

Parameter name	Default value	Description
		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 whether local (craft) procedures can be used to configure the phone. 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.

Table continues…

Parameter name	Default value	Description
		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. Value Operation: • 0: No • 1: Yes
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.
PSTN_VM_NUM		Specifies the dialable string that is used to call into the messaging system. For example, when you press the **Message Waiting** button. **Note:** This parameter is supported when the phone is failed over.
PUSHCAP	0000	Controls the modes of individual push types.

Table continues…

Parameter name	Default value	Description
		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. • ||+--- 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.

Table continues…

Parameter name	Default value	Description
		• 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.
QTP_BUTTON_COMPRESS	0	Specifies the range of features which can be assigned to Quick Touch Panel on phone screen. Features and Autodials configured on SMGR buttons 4 through 24 will show up on the QTP, up to a maximum of 8 buttons. Moreover, Call Appearances, and Bridged Call Appearances will be excluded from QTP. Value Operation: • 0: Buttons will be compressed and all features depicted in SMGR buttons 4 to 11 will be assigned to QTP without blanks. • 1: Buttons will be compressed and blanks removed from the QTP panel.
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. 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.

Table continues…

Parameter name	Default value	Description
		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.
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.

Table continues…

Parameter name	Default value	Description
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 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**.

Table continues…

Parameter name	Default value	Description
		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
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 challengePassword 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.

Table continues…

Parameter name	Default value	Description
		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.
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 96x1Supgrade.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.
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].
SIP_PORT_SECURE	5061	Specifies the destination TCP port for SIP messages sent over TLS. Valid values are 1024 through 65535. The default value is 5061.

Table continues…

Parameter name	Default value	Description
		The parameter is used in non-Avaya environment. In Avaya environment, this parameter is overwritten by PPM configuration.
SIPCONFERENCECONTINUE	0	Specifies if a conference call continues after the host hangs up. Value Operation: • 0: Drop all parties. • 1: Continue conference **Note:**
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.

Table continues…

Parameter name	Default value	Description
SLMCAP	0	Specifies if the SLA Monitor agent is enabled for packet capture. 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 CRAFT menu - Allows you to enable or disable of RTP packets capture using local CRAFT procedures.
SLMCTRL	0	Specifies whether the SLA Monitor agent is enabled for phone control. Value Operation: • 0: Disabled • 1: Enabled • 2: Controlled from craft 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.

Table continues…

Parameter name	Default value	Description
		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. 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.

Table continues…

Parameter name	Default value	Description
SNTPSRVR	Null	Specifies a list of addresses of SNTP servers. 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.
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.
SP_DIRSRVRPORT	389	Sets the TCP port number of the LDAP Directory Server. The default port number is 389.
SP_DIRTOPDN	Null	Sets the Directory Topmost Distinguished Name. This value must be set to a non-null value to enable the LDAP application. The default is null, but DIRTOPDN should be set to the LDAP root entry.
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 craft 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_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. 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.

Table continues…

Parameter name	Default value	Description
		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.
SNTP_SYNC_INTERVAL	1440	Specifies the time interval in minutes when the phone attempts to synchronize its time with the configured NTP servers. Valid values are from 60 min. to 2880 min.
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 it is not a redirect destination of the monitored station.

Table continues…

Parameter name	Default value	Description
		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.
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. Valid values are 5 through 60.
V		
VLANSEP		Specifies whether VLAN separation is enabled by the built-in Ethernet switch while the phone is tagging frames with a non-zero VLAN ID.

Table continues…

Parameter name	Default value	Description
		When VLAN separation is enabled, only frames with a VLAN ID that is the same to the VLAN ID used by the phone (as well as priority-tagged and untagged frames) is forwarded to the phone. Also, if the value of PHY2VLAN is non-zero, only frames with a VLAN ID that is the same to the value of PHY2VLAN (as well as priority-tagged and untagged frames) is forwarded to the secondary (PHY2) Ethernet interface. Tagged frames received on the secondary Ethernet interface will have their VLAN ID changed to the value of PHY2VLAN and their priority value changed to the value of PHY2PRIO. Value Operation 0: Disabled. 1: Enabled if L2Q, L2QVLAN and PHY2VLAN are set appropriately (default).
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

Table continues…

Parameter name	Default value	Description
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.
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.
WBCSTAT	1	Specifies whether a wideband codec indication is displayed when a wideband codec is used. Value Operation: • 0: Disabled • 1: Enabled
WEBLMSRVR	Null	Sets the IP address or Fully-Qualified Domain Name (FQDN) of the licensing server.

Table continues…

Parameter name	Default value	Description
		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.
WMLEXCEPT	Null	Specifies zero or more IP addresses or domains for which the HTTP proxy server specified by WMLPROXY will not be used. The values are separated by commas without any intervening spaces. The value can contain up to 255 characters.
WMLHELPSTAT	1	Specifies if a web application help item will be displayed on the Home screen, if no WML applications are administered and if the value of WMLHOME is null. Value Operation: • 0: A web application help item is not displayed. • 1: A web application help item is displayed.
WMLHOME	Null	Specifies the URL of a WML page to be displayed by default in the WML browser, and whenever the Home softkey is selected in the browser. The value can contain zero or one URL of up to 255 characters. If the value is null, the WML browser is disabled.
WMLIDLETIME	10	Specifies idle time in minutes of after which the web page specified by the value of WMLIDLEUR is displayed. If WMLIDLEUR is null, a web page is not displayed when the phone is idle. Valid values are 1 through 999.
WMLIDLEURI	Null	Specifies zero or one URL for a WML page to be displayed when the phone has been idle for the number of minutes specified by the value of WMLIDLETIME. The value can contain up to 255 characters.
WMLPORT	808	Specifies the TCP port number of the HTTP proxy server specified by WMLPROXY. Valid values are 0 through 65535.
WMLPROXY	Null	Specifies zero or one address for an HTTP proxy server that is used by the WML browser, and by

Table continues…

Parameter name	Default value	Description
		the Weather and World Clock applications on the 9621, 9641 and 9670. The address can be in dotted-decimal (IPv4), or DNS name format, separated by commas without any intervening spaces. The value can contain up to 255 characters.

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