Avaya J100 Series SIP Release 4.0.3.0 Readme

This file is the Readme for the Avaya J100 Series SIP Release 4.0.3.0 software (J100 SIP 4.0.3.0). This file describes the contents of the October 2019 (**4.0.3.0.10**) release software distribution package.

J100 SIP 4.0.3.0 software is supported on the Avaya J129, J139, J159, J169, and J179 IP Phones used with Avaya Aura®, Avaya IP Office™, and select OpenSIP platforms. J100 SIP 4.0.3.0 software will not load or operate on any other models.

This release supersedes all previous Avaya J100 Series SIP software releases. Avaya recommends that all customers using Avaya J100 Series SIP software upgrade to this version at their earliest convenience.

The information in this document is accurate as of the issue date and subject to change.



Please refer to the Advisements in this file for important information prior to deploying this software.

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Compatibility

The Avaya J129, J139, J159, J169 and J179 IP Phones using J100 SIP 4.0.3 software are supported with:

- Avaya Aura® Platform 6.2FP4 (Avaya Aura® Communication Manager 6.3.6, Avaya Aura® Session Manager 6.3.8, Avaya Aura® System Manager 6.3.8) and associated service packs
  - ***Refer to the Advisement section for known limitations if not using 7.1.3.3 or above.***
- Avaya Aura® Platform 7.0.0.0 (Avaya Aura® Communication Manager 7.0.0.0, Avaya Aura® Session Manager 7.0.0.0, Avaya Aura® System Manager 7.0.0.0) and associated service packs
  - ***Refer to the Advisement section for known limitations if not using 7.1.3.3 or above.***
- Avaya Aura® Platform 7.0.1.0 (Avaya Aura® Communication Manager 7.0.1.0, Avaya Aura® Session Manager 7.0.1.0, Avaya Aura® System Manager 7.0.1.0) and associated service packs
  - ***Refer to the Advisement section for known limitations if not using 7.1.3.3 or above.***
- Avaya Aura® Platform 7.1.0.0 (Avaya Aura® Communication Manager 7.1.0.0, Avaya Aura® Session Manager 7.1.0.0, Avaya Aura® System Manager 7.1.0.0, Avaya Aura® Presence Services 7.1.0.0) and associated feature/service packs
  - ***Refer to the Advisement section for known limitations if not using 7.1.3.3 or above.***
- Avaya Aura® Platform 8.0.0.0 (Avaya Aura® Communication Manager 8.0.0.0, Avaya Aura® Session Manager 8.0.0.0, Avaya Aura® System Manager 8.0.0.0, Avaya Aura® Presence Services 8.0.0.0) and associated feature/service packs
  - ***Refer to the Advisement section for known limitations if not using 8.0.1.0 or above.***
- Avaya Aura® Platform 8.1.0.0 (Avaya Aura® Communication Manager 8.1.0.0, Avaya Aura® Session Manager 8.1.0.0, Avaya Aura® System Manager 8.1.0.0, Avaya Aura® Presence Services 8.1.0.0) and associated feature/service packs
- IP Office™ 10.0 SP7 / 10.1 SP3 (J129 only)
  - ***Refer to IP Office documentation for specific compatibility.***
- IP Office™ 11.0 or later for J129/J169/J179
  - ***Refer to IP Office documentation for specific compatibility.***
- IP Office™ 11.0 SP1 or later for J129/J139/J169/J179
  - IP Office™ 11.0 FP4 or later for support of Bluetooth on J179
  - ***Refer to IP Office documentation for specific compatibility.***
- IP Office™ 11.0 FP4 SP2 or later for J159
  - ***Refer to IP Office documentation for specific compatibility.***

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- Avaya Aura® Call Center Elite 7.0.1.0[1], 7.1.0.0[1], 8.0.0.0[1], 8.1.0.0[1]
- OpenSIP Platforms
  - Broadsoft Broadworks R22.0
  - Zang Office R1.0
  - Asterisk R16
  - 3CX R15.5
  - FreeSWITCH 1.8.5
  - Netsapiens 40

Refer to https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml?name=J100+-+SIP&version=4.0 for an up-to-date listing of compatible products.

[1] J169/J179 IP Phone are supported with CC Elite. The J129/J139/J159 IP Phones are not supported with CC Elite.

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New Features in J100 SIP 4.0.3

Avaya J100 Series SIP Release 4.0.3 contains the following new features.

New with this release	Description
Support for J159 IP Phone	J100 Release 4.0.3.0 introduces support for the J159 IP Phone.
Configuration of GROUP via PPM	The ability to configure the GROUP parameter via System Manger **Note:** This functionality also requires Avaya Aura® 8.1.1.
LDAP directory access	Support has been added to directly access an LDAP directory.
J139 screensaver/background image	J139 users can now utilize both standard/custom background images and standard/custom screensaver images
Configure half/full width screen via settings file or web	Allow the half/full width setting to be configured via the 46xxsettings.txt file or via the web administration.
Multicast Paging	Support has been added for the sending and receiving of a multicast page on Avaya Aura® and OpenSIP.
Check for new software and configuration files – server-initiated, Admin/Web menu	Support has been added for server-initiated push (SIP INFO) for checking for new software files and new configuration files. Support has also been added for checking for new software files and new configuration files from the Admin/Web menu.
Additional capabilities for 3CX	Support has been added for Config Ringtone, Admin Ringtone, and Shared Parking
Additional capabilities for Broadsoft	Support has been added for Broadsoft Call Center features on the J169/J179

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Documentation for J100 SIP 4.0.3

The following documentation has been updated for this release:

- J100 Series IP Phone Overview and Specifications
- Installing and Administering Avaya J100 IP Phones
- Installing and Administering Avaya J100 Series IP Phones in an Open SIP environment
- Using Avaya J129 IP Phone SIP
- Using Avaya J139 IP Phone SIP
- Using Avaya J169/J179 IP Phone SIP
- Avaya J129 IP Phone Quick Reference
- Avaya J139 IP Phone Quick Reference
- Avaya J169/J179 IP Phone Quick Reference
- Using Avaya J100 series IP Phone for Call Center Agents
- Avaya J129 IP Phone in an Open SIP environment Quick Reference guide
- Avaya J139 IP Phone in an Open SIP environment Quick Reference guide
- Avaya J169/J179 IP Phone in an Open SIP environment Quick Reference Guide
- Using Avaya J129 IP Phone in an Open SIP environment
- Using Avaya J139 IP Phone in an Open SIP environment
- Using Avaya J169/J179 IP Phone in an Open SIP environment

The following documentation has not been updated and is included below for reference:

- Using JBM24 Button Module
- Using Avaya J100 Expansion Module

The following Partner Configuration guides are included below for reference:

- Broadsoft Partner Configuration Guide – J100 Series
- Asterisk Partner Configuration Guide – J100 Series
- 3CX Partner Configuration Guide – J100 Series
- FreeSWITCH Partner Configuration Guide – J100 Series

These documents are available on http://support.avaya.com under "J100 Series IP Phones " -> "SIP 4.0.x" -> Documents. They are also available on https://documentation.avaya.com under "J100 Series Phones".

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J100 SIP 4.0.3.0 (4.0.3.0.10) Package Content

The J100 SIP 4.0.3.0 package (J100-IPT-SIP-R4_0_3_0-101019.zip) contains all the files necessary to upgrade Avaya new or previously installed Avaya J129/J139/J159/J169/J179 IP Phones to the J100 SIP 4.0.3.0 software.

- FW_S_J129_R4_0_3_0_10.bin – application binary file for J129
- FW_S_J139_R4_0_3_0_10.bin – application binary file for J139
- FW_S_J159_R4_0_3_0_10.bin – application binary file for J159
- FW_S_J169_R4_0_3_0_10.bin – application binary file for J169
- FW_S_J179_R4_0_3_0_10.bin – application binary file for J179
- FW_JEM24_R1_0_1_0_10.bin – application binary file for the JEM24
- J100Supgrade.txt – This file is downloaded by the IP Phones and instructs the phone on how to upgrade to this version of software
- Predefined language files for phone display:
    - Mlf_J129_BrazilianPortuguese.xml
    - Mlf_J129_CanadianFrench.xml
    - Mlf_J129_CastilianSpanish.xml
    - Mlf_J129_Chinese.xml
    - Mlf_J129_Dutch.xml
    - Mlf_J129_English.xml
    - Mlf_J129_German.xml
    - Mlf_J129_Hebrew.xml
    - Mlf_J129_Italian.xml
    - Mlf_J129_Japanese.xml
    - Mlf_J129_Korean.xml
    - Mlf_J129_LatinAmericanSpanish.xml
    - Mlf_J129_ParisianFrench.xml
    - Mlf_J129_Polish.xml
    - Mlf_J129_Russian.xml
    - Mlf_J129_Turkish.xml
    - Mlf_J139_Arabic.xml
    - Mlf_J139_BrazilianPortuguese.xml
    - Mlf_J139_CanadianFrench.xml
    - Mlf_J139_CastilianSpanish.xml
    - Mlf_J139_Chinese.xml
    - Mlf_J139_Dutch.xml
    - Mlf_J139_English.xml
    - Mlf_J139_German.xml
    - Mlf_J139_Hebrew.xml
    - Mlf_J139_Italian.xml
    - Mlf_J139_Japanese.xml
    - Mlf_J139_Korean.xml
    - Mlf_J139_LatinAmericanSpanish.xml
    - Mlf_J139_ParisianFrench.xml
    - Mlf_J139_Polish.xml
    - Mlf_J139_Russian.xml
    - Mlf_J139_Thai.xml
    - Mlf_J139_Traditional_Chinese.xml
    - Mlf_J139_Turkish.xml
    - Mlf_J159_Arabic.xml
    - Mlf_J159_BrazilianPortuguese.xml

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  - Mlf_J159_CanadianFrench.xml
  - Mlf_J159_CastilianSpanish.xml
  - Mlf_J159_Chinese.xml
  - Mlf_J159_Dutch.xml
  - Mlf_J159_English.xml
  - Mlf_J159_German.xml
  - Mlf_J159_Hebrew.xml
  - Mlf_J159_Italian.xml
  - Mlf_J159_Japanese.xml
  - Mlf_J159_Korean.xml
  - Mlf_J159_LatinAmericanSpanish.xml
  - Mlf_J159_ParisianFrench.xml
  - Mlf_J159_Polish.xml
  - Mlf_J159_Russian.xml
  - Mlf_J159_Thai.xml
  - Mlf_J159_Traditional_Chinese.xml
  - Mlf_J159_Turkish.xml
  - Mlf_J169_J179_Arabic.xml
  - Mlf_J169_J179_BrazilianPortuguese.xml
  - Mlf_J169_J179_CanadianFrench.xml
  - Mlf_J169_J179_CastilianSpanish.xml
  - Mlf_J169_J179_Chinese.xml
  - Mlf_J169_J179_Dutch.xml
  - Mlf_J169_J179_English.xml
  - Mlf_J169_J179_German.xml
  - Mlf_J169_J179_Hebrew.xml
  - Mlf_J169_J179_Italian.xml
  - Mlf_J169_J179_Japanese.xml
  - Mlf_J169_J179_Korean.xml
  - Mlf_J169_J179_LatinAmericanSpanish.xml
  - Mlf_J169_J179_ParisianFrench.xml
  - Mlf_J169_J179_Polish.xml
  - Mlf_J169_J179_Russian.xml
  - Mlf_J169_J179_Thai.xml
  - Mlf_J169_J179_Traditional_Chinese.xml
  - Mlf_J169_J179_Turkish.xml
- Eight extended Korean ring tone files:
  - KoreanRT1.xml
  - KoreanRT2.xml
  - KoreanRT3.xml
  - KoreanRT4.xml
  - KoreanRT5.xml
  - KoreanRT6.xml
  - KoreanRT7.xml
  - KoreanRT8.xml
- AvayaLanguageTool_SIP.xlsm – Excel tool for creating additional language files
- One certificate file:
  - av_prca_pem_2033.txt – Avaya Product Root CA certificate with an expiration date of 2033
- Avaya-J100IpPhone-MIB.mib – mib file
- release.xml
- A "signatures" subdirectory containing signature files and a certificate file. Both SHA-1 and SHA-256 signature files are included

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- Avaya Global Software License Terms 092018.pdf

System specific parameters should be entered into the 46xxsettings.txt file which is available for separate download at http://support.avaya.com. **New/changed configuration parameters with this release of software are shown in** Appendix 3 **.**

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Advisements with J100 SIP 4.0.3 software

J129/J139/J159/J169/J179 – Aura Feature Provisioning Limitations

Avaya Aura® Communication Manager 7.1 and below does not provide native support of the J129/J139/J159/J169/J179 IP Phones. The J129 should be administered as a "9608SIP", the J139 as a "9608SIP", the J159 as a "9608SIP" with single expansion module, the J169 as a "9611SIP" or "9611SIPCC" and the J179 as a "9611SIP" or "9611SIPCC".

Avaya Aura® 8.0.1 provides native support of the J129/J169/J179 IP Phones. The J139 should be administered as J169. The J159 should be administered as a J169 with single expansion module.



When using Avaya Aura® 7.1 (and below) and J100 4.0.0.1 (and below) there are feature administration limitations. **See the table below for further details.**

Service Packs to allow for administration of J100-Series IP Phones on Avaya Aura® 7.1.3.3 have been provided by Avaya. Avaya strongly recommends that all customers using J100-series IP Phones upgrade to these service packs. There are two software components to deliver this solution:

1. A System Manager 7.1.3.3 Service Pack identified via PCN2062Su.
2. A Session Manager 7.1.3.3 Service Pack identified via PCN2068Su.

Session Manager PSN PSN005267u details the main operational changes and a list of things that should be considered when rolling out this solution, including some differences that may be seen when both J100-Series IP Phones and 9600-Series IP Phone are used within the same environment.

Note: Be sure to read the PSN before rolling out the software changes to the servers

NOTE: **For releases prior to Avaya Aura® 8.1.1, it is recommended that all endpoints for a specific user be of the same device family.** A single user can have multiple J100-series IP Phones, or multiple 9600-series IP Phones, but it is not acceptable for a single user to have a mix of the two series associated with the same extension. If this occurs, only the button/profile settings for the configured endpoint type in CM can be modified via the System Manger user interface. **Avaya Aura® 8.1.1 or later allows for a specific user to have a mix of J100-series IP Phones and 9600-series IP Phones.**

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The following two tables highlight the experience that end users and administrators will see when using J100 IP Phones with J100 SIP software on various releases of Avaya Aura®.

Avaya Aura® configuration	**What to expect when a user logs in for the very first time**	**What to expect when SMGR Add, Move Features and Autodials after user has logged in**	**What to expect when user changes a label on their phone**
Avaya Aura® 7.1.3.2 or earlier (J100 aliased as a 9600)	All features and auto dials will show on the phone and button module at the line locations as defined in SMGR. Labels defined in SMGR will NOT show – i.e phone will show default labels.	New Favorite features/pre-configured autodials will NOT show up on Phone screen. Moved Favorite feature/auto-dial will NOT show up in the phone screen and will cause the original key to disappear. NOTE: J100 4.0.0.1 includes a new configuration parameter (SMGR_AUTO_FAVORITE) which will auto populate Features/Autodials if less than Aura 8.0.1 and added/moved keys will not disappear.	Label change will NOT show in SMGR.
Avaya Aura® 7.1.3.3 (J100 aliased as a 9600)	All features and auto dials and labels will show on the phone and button module at the line locations as defined in SMGR.		Label change will show in SMGR
Avaya Aura® 8.0.1 or later (J100 native)	Features and auto dials marked as favorites will show on the phone and button locations. Features not marked as favorites will only show in the Feature menu. Pre-configured Autodials not marked as Favorites will show up only during end user customization. The phone will show the labels defined on keys marked as Favorites in SMGR on the Phone	New Favorite features/pre-configured autodials will show up on Phone screen. Moved Favorite feature/auto-dial will show up in the phone screen.	

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<table>
<tr><th>Avaya Aura® configuration</th><th>What to expect when a user moves/adds a feature / auto dial to a different line location on the phone</th><th>What to expect when a user deletes a feature / auto dial using customization.</th><th>What to expect when user does a Customization -> Restore</th></tr>
<tr><td>Avaya Aura® 7.1.3.2 or earlier (J100 aliased as a 9600)</td><td rowspan="2">Change will not show in SMGR (similar behavior as SIP 9600-series)</td><td rowspan="2">No change in SMGR.</td><td rowspan="2">All features and autodials will show on the phone and button module at the line locations defined in SMGR.

Phone will show default labels.

No change in SMGR.</td></tr>
<tr><td>Avaya Aura® 7.1.3.3 (J100 aliased as a 9600)</td></tr>
<tr><td>Avaya Aura® 8.0.1 or later (J100 native)</td><td rowspan="2">Feature favorite flag will be unchecked in SMGR</td><td rowspan="2">All features and autodials marked as favorites will show on the phone and button module at the line locations.

No change in SMGR.

Phone will show labels as defined in SMGR.</td></tr>
<tr><td>Avaya Aura® 8.1.1 or later (J100 native)</td><td>Changes will be reflected in the SMGR Endpoint editor “phone view” tab.</td></tr>
</table>

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Avaya Aura® – Busy Indicator – does not work on Aura 8.0 to Aura 8.1.0



J100 SIP 4.0.2.0 adds support for Busy Indicator on Avaya Aura® with the J169 and J179 IP Phones. A maximum of 48 Busy indicatiors are supported. This functionality will work on Avaya Aura® 7.x when the J169/J179 are aliased as noted above. This functionality will NOT work on Avaya Aura® 8.0 to Avaya Aura® 8.1.0 when the J169/J179 are natively supported. Avaya Aura® 8.1.1 supports the busy-ind button for J169/J169CC/J179/J179CC.

JEM24 Software – J169/J179 Software Compatiblity



The J100 SIP 4.0.2.0 installation zip file contains JEM24 1.0.1.0.8 software for installations on associated JEM24 modules. The JEM24 will automatically be upgraded from the J169/J179 IP Phone. The JEM24 1.0.1.0.8 software will work with J100 SIP 4.0.2.0 or later and Deskphone H.323 6.8.2 or later. Attempting to use it with earlier versions of J100 SIP / Deskpone H.323 software will result in the JEM24 not being able to connect to the J169/J179.

J169/J179 – Upgrade from J100 SIP 1.5.0 – re-enter configuration



End-users who have customized their J169/J179 when using J100 SIP 1.5.0 software will need to re-do the customization following an upgrade to J100 SIP 2.0.0 or later

3PCC Hardware – cannot be used with Avaya Aura® or Avaya IP Office™



Customers can purchase "3PCC" versions of the J129/J139/J159/J169/J179 hardware which are pre-configued for interworking with Open SIP platforms such as Broadsoft, Zang Office, Asterisk, and 3CX. **When using J100 3.0.0.1 or later software, the "3PCC" hardware cannot be converted for use on Avaya Aura® or Avaya IP Office™.**

J179 with Expansion Modules (JBM, JEM24) modules – 5 volt power supply may be required



There are certain power requirements when connecting either the JBM or JEM24 expansion modules to the phone. Depending upon the amount of power supplied by the power source over Ethernet, it may be necessary to power the phone by a separate 5 Volt power supply. Please see the *Power Specifications* section in the "Installing and Administering Avaya J100 IP Phones".

Public Certificates

J100 SIP software includes 64 built-in public CA certificates from a wide range of vendors which can be used instead of having to explicitly add them via a TRUSTCERTS parameter. The use of these certificates is controlled by the ENABLE_PUBLIC_CA_CERTS parameter. A full list of the certificates is included in Appendix B of "Installing and Administering Avaya J100 IP Phones" and "Installing and Administering Avaya J100 Series IP Phones in an Open SIP environment".

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J129/J139/J159/J169/J179 IP Phones – Minimum software release

Avaya periodically releases new hardware variations of the J100-Series IP Phones typically to address a need to change hardware components. That change may require a new version of software to support the new hardware. This then forces a minimum software release supported on that hardware. ***Attempts to downgrade these models to lower versions of software will be rejected***. Refer to Appendix 1.

J159/J169/J179 IP Phones – Avaya Aura® – Feature Buttons on Main Screen

When deploying a J159/J169/J179 IP Phone on Avaya Aura®, there are three ways to make feature buttons appear on the main screen:

- If using Avaya Aura® 7.1.3.3 or later, tag them as a "Favorite" in SMGR
- Use the SET PHONEKEY parameter in the 46xxsettings.txt file (either for all cases, or on a group basis, or on a MAC-specific basis)
- Use the end-user ability to relayout the screen (Settings -> Phone -> Phone Keys Customization)

Language Localization Software Tool and Localized Language Files

Avaya includes a Language Localization Tool (AvayaLanguageTool_SIP.xlsm) as part of the software download package. This tool allows users to create custom downloadable language files for the J100-series in addition to the built-in language files.

Additional information on the tool as well as already-made localized language files can be found at https://support.avaya.com/downloads/download-details.action?contentId=C2019925105008420_7&productId=P1661

Limitations with IPv6

J100 1.5.0 and later includes support for IPv6 interworking. The following are known limitations of the J100 4.0.0 or later implementation:

- Open SIP Interoperability
- Extended rebind
- LLDP configuration of IPv6 related settings is not supported
- Microsoft Exchange integration over IPv6 must use an FQDN for EXCHANGE_SERVER_LIST. i.e. SET EXCHANGE_SERVER_LIST exch1.myco.com
- The following functionality is only supported via IPv4
    - RTCP
    - Push
    - Avaya Diagnostic Server (ADS / SLAMon)
    - Shared Control / Deskphone Mode
    - Interworking with CC Elite.

SSH – Remote Access (EASG)

J100 SIP software contains an SSH server which is used only by Avaya Services for debugging purposes. The SSH server supports only Avaya Services Logins ("craft" and "sroot"). By enabling Avaya Services Logins, you are granting Avaya access to your system. This is required to maximize the performance and value of your Avaya support entitlements by allowing Avaya to resolve product issues in a timely manner. By disabling Avaya Services Logins, you are preventing Avaya access to your system. This is not

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recommended as it can impact Avaya's ability to provide support for the product. Unless the customer is well versed in managing the product themselves, Avaya Services Logins should not be disabled. The access to the SSH server is protected by EASG (Enhanced Access Security Gateway).

Support for SHA2-signed software files

The software files are signed using both SHA-1 and SHA-256 digital signatures. J100 SIP software is capable of SHA-1 and SHA-256 digital signature verification.

Support for OCSP

J100 SIP software supports OCSP (Online Certficate Status Protocol) for checking whether certificates presented to the phone by servers are good, revoked, or unknown. If a certificate is revoked, the TLS connection will not be established or will be closed (in the case of an ongoing TLS connection). OCSP is supported for 802.1x (EAP-TLS), SIP over TLS, WiFi (EAP-TLS) and HTTPS.

MLPP – Limitations during a server failure

Call override/preemption is not available during a preserved call caused by inability to access Session Manager.

Bi-Directional EHS – Compatible Headsets

Compatibility testing of the Bi-Directional EHS functionality with headsets from 3rd-party vendors is undertaken through the Avaya DevConnect program.

Microsoft Exchange Integration using EWS

If Microsoft Exchange Integration is enabled and the phone is connecting to Exchange Server 2010 or later, Exchange Web Services (EWS) is used for the connection. This connection is secured using HTTPS by default which means that the phone is required to validate the Exchange Server identity certificate. To validate the certificate, the TRUSTCERTS parameter in the settings file must include the root certificate of the Certificate Authority (CA) which issued the Exchange Server identity certificate. This configuration will work if the identity certificate was directly issued by the CA root certificate.

If a public CA such as VeriSign is used to obtain an identity certificate for the Exchange Server, the identity certificate will be issued by an intermediate CA certificate and not by the root. In this case, both the root and intermediate CA certificates must be installed on the phone using TRUSTCERTS or the HTTPS connection will fail. In general, if the Exchange Server identity certificate is issued by an intermediate CA, all certificates from the intermediate CA up to the root must be included in TRUSTCERTS for installation on the phone so that the entire certificate chain is available for validation.

Debug mode

As a general guide, it should be noted that response times could be impacted when debug or syslog is enabled

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SIP_CONTROLLER_LIST

This parameter consolidates SIP controller parameters for IP address, port, and transport protocol into a single configuration parameter. The parameter setting should be a list of controller information where the format for each controller entry is "host:port;transport=xxx" . The host should be specified only by an IP address when interworking with Avaya Aura™. The use of Fully Qualified Doman Names (FQDN) is only supported in IP Office and OpenSIP environments. This applies to all sources of the SIP_CONTROLLER_LIST parameter which includes DHCP, LLDP, Web interface and the 46xxsettings.txt file

Security Certificates – IP Address versus FQDN

There is an industry movement towards the use of a FQDN (Fully Qualified Domain Name) instead of an IP address for the Subject Alternate Name or Subject Common Name for security certificates. J100 software supports a FQDN_IP_MAP parameter which specifies mapping of FQDNs to IP addresses for the purpose of validating an FQDN identity found in a server certificate.

SRTP (Media Encryption)

In order to correctly use SRTP, there are various components within the network that you must correctly configure. For J100 Series IP Phones to function properly with SRTP in an Avaya Aura© environment, you must configure the equivalent parameters in Communication Manager or System Manager. Avaya strongly recommends that the following three parameters on the J100 Series IP Phones and the equivalent Communication Manager parameters must match:

SET ENFORCE_SIPS_URI 1
SET SDPCAPNEG 1
SET MEDIAENCRYPTION X or
SET MEDIAENCRYPTION X,Y or
SET MEDIAENCRYPTION X,Y,Z

J100 software supports AES-256 media encryption. Care must be taken to properly configure the encryption parameter when this is used in conjunction with other devices that do not support AES-256.

EAP TLS

When EAP-TLS is enabled using the CRAFT menu, the phone should be rebooted to allow for proper EAP-TLS authentication.

Multi Device Access

Refer to the "Avaya Aura Multi Device Access White Paper" which is available on http://support.avaya.com for known limitations.

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Language support

The J129 IP Phones does not support a Arabic, Thai or Traditional Chinese user interface.

Ringtone and Ringtone Wave Files

Numeric only naming conventions should be avoided with ringtone names (E.g. 12345.wav). The maximum allowed size of an individual ringtone file is 512 KB. The maximum allowed size of all ringtone files is 5120 KB.

Headset Profiles

J100 SIP 1.5.0.0 and later software supports "Headset Profiles"[2] to provide optimum performance for different brands of headsets. An up-to-date version of the profile <-> vendor cross reference can be found at https://downloads.avaya.com/css/P8/documents/100173755.

Avaya Session Border Controller for Enterprise

For all IP Phones which are remotely connected through an SBCE, please ensure that the following is set in the 46xxsettings.txt file

SET WAIT_FOR_REGISTRATION_TIMER 40

SIP Transport Protocols

TCP or TLS are the recommended transport protocols. UDP transport is not supported with J100 SIP software except in a OpenSIP environment.

Encryption – SHA2 and RSA 2048

J100 software supports RSA 2048 bit length encryption keys and supports the SHA2 (224, 256, 384, and 512) hash algorithm. This has been certified for HTTPS usage for web-based administration of these phone sets. When the TLS server-client handshake is initiated, this IP Phone (operating as the client) is able to send its Identity certificate with an enhanced digital signature (SHA2/2048 key). Additionally, this IP Phone is able to receive and validate server Identity certificates which have an enhanced digital signature (SHA2/2048 key).

Interworking – Avaya Diagnostic Server (ADS)

Avaya J100 SIP Release 2.0.0.0 and later supports the ADS server. The SLMSRVR parameter must be set in the 46xxsettings.txt file for this version of the agent to register with ADS. In addition, a valid certificate file must be downloaded via TRUSTCERTS.

Avaya Diagnostic Server 3.0.3 is the minimum release to support the J129 IPhone, the J169 IP Phone and the J179 IP Phone.

Avaya Diagnostic Server 3.0.4 is the minimum release to support the J139 IP Phone.

[2] J129 does not support a headset

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"Desk Phone" Mode and Lock

Avaya one-X® Communicator, Avaya Equinox and similar UC applications from Avaya support a "Desk Phone" (Shared Control) mode in which the UC application can control an associated IP Phone. An IP Phone supports a "Lock" mode, which can be entered either manually or automatically, which prevents the dialing of any number except for an emergency number using the keypad of the IP Deskphone. If an IP Phone is in Shared Control with a UC application and is also in a "Lock" state, placing a call from the UC application will still result in the call being established from the IP Phone.

Demo Certificates – Avaya Aura® Session Manager 6.3.8 and newer



New installations of Avaya Aura® Session Manager Release 6.3.8 generate SIP and HTTPS (PPM) certificates signed by System Manager CA during installation. Previous versions used a demo Avaya certificate which is deprecated as it does not meet current NIST security standards. The generated Session Manager certificates signed by System Manager CA do not contain all the attributes (SIP domain, IP address, etc.) required by the Avaya IP Phone to correctly validate them. For that reason it is recommended to replace them. To replace the Session Manager certificates signed by System Manager CA to comply with the IP Phone requirements, follow the "Installing Enhanced Validation Certificates for Session Manager" section of the Session Manager Administration Guide. Optionally customers could replace the Session Manager certificates for those signed by a third party CA. For more details, follow the Session Manager Administration Guide.
Upgrading to Avaya Aura® Session Manager Release 6.3.8 or later preserves the demo Avaya certificates used on SIP and HTTPS (PPM) TLS connections. When using J100 Series IP Phones, the demo Avaya certificates MUST be replaced. Refer to the Session Manager Administrator Guide for more details.

Removal of Avaya SIP Root CA Certificate

The Avaya SIP Root CA Certificate for demo certificates (av_sipca_pem_2027.txt) is not included in the installation package. As noted above, the demo certificate has been deprecated as it does not meet current NIST security standards.

Interworking – TLS 1.2

J100 software supports TLS 1.2 and adds includes cipher suites FIPS:!ADH:!DSS:-SSLv3:DHE-RSA-AES256-SHA:AES256-SHA:DHE-RSA-AES128-SHA:AES128-SHA.



J100 software also includes a configuration parameter (TLS_VERSION) which can be used to configure the IP Phone to only use TLS 1.2. Care must be taken to only use this parameter when all components to which the IP Phone will communicate can also support TLS 1.2.

J129/J139 - Presence

The J129 does not display presence in an Avaya Aura® network or have the ability to manually set a presence state. The J129 publishes presence information for other clients that support viewing presence.
The J139 displays presence and publishes presence but does not have the ability to manually set a presence state.

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The J169 and J179 both display presence, publish presence, and have the ability to manually set a presence state

VLAN separation

The J100 software supports 3 versions of VLAN separation; 1) Full VLAN separation, 2) Partial VLAN separation and 3) No VLAN separation. However, the J129 IP Phone does NOT support partial VLAN separation.

Avaya highly recommends that voice and data traffic be separated by VLANs and that voice traffic has its own VLAN.

Features not supported on the J129 Phone

The following features are not supported by the J129 IP Phone with J100 software:

- Exchange integration, WML browser, URI dialing, simultaneous display of caller name and number, redial by list, conference roster list, missed call filtering, displaying presence, downloadable ringtones, Favorites, Personalize labels
- Bridge call appearances (except MDA)
- MLPP, Call Pickup, Hunt Group Busy, Team Button, Enhanced Call Forward, Dial Intercom, Exclusion, LNCC, Priority Calls, Whisper Page, Busy Indicator
- Interworking with Contact Center Elite (CC Elite)

Features not supported on the J139 Phone

The following features are not supported by the J139 IP Phone with J100 software:

- Exchange integration, WML browser, URI dialing, simultaneous display of caller name and number, redial by list, conference roster list, missed call filtering, manually setting presence, downloadable ringtones, Favorites, Personalize labels
- Bridge call appearances (except MDA)
- Hunt Group Busy, Team Button, Enhanced Call Forward, Dial Intercom, Exclusion, LNCC, Priority Calls, Whisper Page, Busy Indicator
- Interworking with Contact Center Elite (CC Elite)

Features not supported on the J159 Phone

The following features are not supported by the J159 IP Phone with J100 software:

- WML browser,
- Interworking with Contact Center Elite (CC Elite)
- Bluetooth

J129 with IP Office – Features supported

The following features are supported by the Avaya IX™ J129 IP Phone when deployed on Avaya IP Office™: Attended transfer, Unattended transfer, transferring a call by selecting a contact or recents, personal directory, voice mail, manual dial mode, conference.

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The following features are supported by the Avaya IX™ J129 IP Phone when deployed on Avaya IP Office™ using a short code: Call Forward, Call Forward Busy, Call Park/Unpark, Do Not Disturb, Automatic Call Back, Private Call, Speed Dial.

J139 with IP Office – Features supported / not supported

The following features are supported by the Avaya J139 IP Phone when deployed on IP Office™:

- Basic call handling on *Call Appearances and Line Appearances only* – Making a call, Call presentation, Answer, Hold, Transfer, Conference, Drop
- IP Office Directory (Personal and System)
- IP Office Call History
- Visual Voice

Include basic operation and call handling feature controls by default via IP Office Features Menu

- DND
- Forwarding
- Mobile Phone Call Twinning (User must first be administered to permit Mobile Twinning by a system Administrator).
- Hot Desking

Allow basic call handling feature controls to be administered as button features by a system Administrator

- Call Park
- Call Pickup
- Call Page
- Call Recording
- Auto Call-back
- Account Code
- Authorisation Code
- User BLF (*NOTE: Requires IP Office 11.0 FP4*)
- Group BLF (*NOTE: Requires IP Office 11.0 FP4*)

Allow basic agent controls to be administered as button features by a system Administrator

- Hunt Group Membership
- Agent Status
- After Call Work
- Coaching Request

The following features are <u>not</u> supported by the Avaya J139 IP Phone when deployed on IP Office™:

Advanced Call Presentation / Handling:

- MADN

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- Bridged Appearances
- Coverage Appearances

IP Office Features/Status Menus:

- Advanced Call Pickup
- Advanced Call Park
- DND exceptions
- Account / Authorisation Code
- Auto Answer Controls
- Withhold Number
- Coverage Ring Controls
- Advanced Hunt Group Controls: (Multi Membership, Group Status, Group Configuration)
- Self-Administration
- System Administration

Button configuration:

- Hands-free Answer
- Automatic Intercom
- Specific Call Dial Types
- Conference Meet-Me
- Self-Administration
- System Administration
- Advanced Hunt Group Controls (Group Status, Group Configuration)
- Agent Supervisor Features: (Call Steal, Call Listen, Call Intrude, Coaching Intrusion)

Others:

- 9600/J100 Push API
- WML Browser
- Exchange Calendar/Contact Integration
- Multicast Paging

J159/J169/J179 with IP Office – Features supported / not supported

The following features are supported by the Avaya J159/J169/J179 IP Phone when deployed on IP Office™:

- Basic call handling on *Call Appearances and Line Appearances only* – Making a call, Call presentation, Answer, Hold, Transfer, Conference, Drop
- IP Office Directory (Personal and System)
- IP Office Call History
- Visual Voice

Include basic operation and call handling feature controls by default via IP Office Features Menu

- DND
- Forwarding

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- Mobile Phone Call Twinning (User must first be administered to permit Mobile Twinning by a system Administrator).
- Hot Desking

Allow basic call handling feature controls to be administered as button features by a system Administrator

- Call Park
- Call Pickup
- Call Page
- Call Recording
- Auto Call-back
- Account Code
- Authorisation Code
- User BLF
- Group BLF

Allow basic agent controls to be administered as button features by a system Administrator

- Hunt Group Membership
- Agent Status
- After Call Work
- Coaching Request

The following features are also supported by the Avaya J159/J169/J179 IP Phone when deployed on IP Office™:

Advanced Call Presentation / Handling:

- MADN
- Bridged Appearances
- Coverage Appearances

IP Office Features/Status Menus:

- Advanced Call Pickup
- Advanced Call Park
- DND exceptions
- Account / Authorisation Code
- Auto Answer Controls
- Withhold Number
- Coverage Ring Controls
- Advanced Hunt Group Controls: (Multi Membership, Group Status, Group Configuration)
- Self-Administration
- System Administration

Button configuration:

- Hands-free Answer
- Automatic Intercom
- Specific Call Dial Types
- Conference Meet-Me
- Self-Administration
- System Administration

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- Advanced Hunt Group Controls (Group Status, Group Configuration)
- Agent Supervisor Features: (Call Steal, Call Listen, Call Intrude, Coaching Intrusion)

The following features are not supported by the Avaya J159/J169/J179 IP Phone when deployed on IP Office™:

- Personalization (i.e. ability to reconfigure the button layout)
- 9600/J100 Push API
- WML Browser
- Exchange Calendar/Contact Integration
- Multicast Paging

Deploying the J129/J139/J159/J169/J179 in OpenSIP Platform

The J129/J139/J159/J169/J179 are supported with Broadsoft Broadworks, Zang Office, Asterisk, 3CX, FreeSwitch, and Netsapiens. IP phone configuration file (settings file) must be deployed from a file server (HTTP or HTTPS). User backup/restore must also be deployed from a file server (HTTP or HTTPS). SIP Transport = TLS is not supported. For these phones to work in an OpenSIP environment, configuration file (settings file) must have following parameter configured with value as given:

  - SET ENABLE_AVAYA_ENVIRONMENT 0
  - SET DISCOVER_AVAYA_ENVIRONMENT 0
  - SET ENABLE_IPOFFICE 0

- See "Installing and Administering Avaya J100 Series IP Phones in an Open SIP environment" for more detail.

Provisioning of File Server Address

Phone can be provisioned using HTTP/S File Server. The HTTP/S File Server address can be provided to the phone through one of the following methods:

- DHCP
- LLDP
- CRAFT/Web Interface
- Device Enrolment Service (DES)

HTTPS file server has priority over the HTTP file server if both configured.

Once provisioned using one of the above methods, HTTP/S file server address can also be changed through settings file by using following parameters:

- For HTTP → HTTPSRVR, HTTPDIR, HTTPPORT
- For HTTPS → TLSSRVR, TLSDIR, TLSPORT

Once File server address is changed through settings file it will override the file server address provided through DHCP or LLDP. Thus, it is advised to use this option only if different server address needs to be provided to override the DHCP.

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If HTTPS file server address is configured in setting file, phone will contact to HTTPS server immediately after the download of settings file without any reboot.

Note:

Please take a note that when HTTPS file server address is configured in settings file, configure SET HTTPSRVR "" in the settings file to override the HTTPSRVR value received from DHCP. Commenting out the HTTPSRVR parameter will not override the value received from DHCP.

Preconfig keys considerations

Improved in Release 4.0.2.0 is the ability to pre-configure keys using the 46xxsettings.txt file or the WebUI.

Due to multiple ways you can configure keys in an Aura environment please be aware if you are configuring keys in System Manager, WebUI, and the 46xxsettings.txt file that conflicts may arise.

In Aura and OpenSIP environments it is reccomended to use either WebUI or 46xxsettings.txt but not both.

Note: MDA groups must use the same preconfig.

For more details please refer to the "Installing and Administering Avaya J100 IP Phones".

3CX Transfer considerations

Blind transfer when registered with a 3CX behaves like a consultative transfer.

The 3CX server intentionally leaves the call held on the transferor phone until the far end answers the call or the call is ended in some other way.

This means before the transfer is answered, the transferor phone shows a Resume softkey. Please ignore the Resume soft key, it will have no effect if selected, the transfer will complete when the transfer is answered or the call is ended.

PHONEKEY customization limitations

CCElite phones are not supported by PHONEKEY customization.

Recents (Call History) – limitations when downgrade from 4.0.3.0 or later

Starting in J100 SIP 4.0.3.0, call logs are encrypted on the phone. If the phone is downgraded from 4.0.3.0 to any previous release then the **call logs will be lost** since the older release will not be able to read the encrypted logs. This will occur in all environments except Avaya Aura® when centalized call logging is enabled. If centralized call logging is enabled, then Recents will be preserved on downgrade from 4.0.3.0 to older firmware versions.

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J100 4.0.3.0 Resolved Issues (since J100 4.0.2.1)

The following table includes issues which are resolved with this release of software compared to J100 4.0.2.1.3

External ID	Internal ID	Issue Description
Avaya Aura®		
	SIP96X1-60449	Phone does not allow Blind Transfer after pressing "Phone" hard button to go to "Select destination" screen and press LED button beside Busy Indicator extension
	SIP96X1-61417	Phone is shared control generates tSharedControlM coredump when cancel login user.
1-14853815173	SIP96X1-61961	J169 SIP Forward call displays as RAI with German
1-14897650562	SIP96X1-63005	J1xx Do not wake up from sleep mode
1-14902889422	SIP96X1-63338	J179 Duplicate answered Call Log
1-14840929201	SIP96X1-63552	J169 BA call cannot be picked up when answered with Handset
Open SIP		
	SIP96X1-61347	The toogle control icons in Shared should show the current state.
	SIP96X1-60451	3PCC 3CX PnP Failed to Barge-in BLF call
	SIP96X1-56316	[Freeswitch 3PCC] Phone cannot detect the state of BLF phone when using an FQDN for the sip proxy list.
	SIP96X1-53711	[Asterisk] phone crashes after consultative transferring by pressing BLF line key
	SIP96X1-62367	In 3pcc mode, the display name can be taken from the Contact header
	SIP96X1-63493	CLID update fails if the remote party after the transfer is a Non-Broadworks User
All Platforms		
	SIP96X1-57796	AUDASYS parameter works incorrectly *work-around: adjust volume to 0 on rocker*
	SIP96X1-31080	Pressing key "0" for the first input in contact group name then press other key, other key is doubled
	SIP96X1-61771	When using WiFi the phone the download of the MAC address file uses the WiFi module MAC address
1-14878008893	SIP96X1-63653	Phone reboots when receiving 180 ringing with long contact number.
Web User Interface		
	SIP96X1-57487	Phone doesn’t enable WEB server automatically in 3CX mode
Bluetooth		
	SIP96X1-59261	After Headset is restarted (after upgrade) or plugged out/in it doesn't auto-reconnect to J179 phone
Button Module (JEM24 and JBM24)		

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External ID	Internal ID	Issue Description
1-14870650488	BUTTONMODULE-505	JEM24 with 1.0.1.0.7/1.0.1.0.8/1.0.1.0.9 cannot be upgraded or downgraded

Unresolved issues in J100 4.0.3.0

The following table includes unresolved issues with this release of software which were known as of the issue date of this document.

External ID	Internal ID	Issue Description
Avaya Aura®		
	SIP96X1-30262	Remote Control icon is not getting displayed for logged out J169 phones
	SIP96X1-31097	Parameter PKCS12_PASSWD_RETRY works incorrectly.
	SIP96X1-31177	J100 phones take time to perform 'hold/transfer' SK operation if these SKs are pressed after recording an active call.
	SIP96X1-44264	After reset to defaults, dual phone displays 'Acquiring service' icon although phone can login extension successfully *work-around:reboot phone*
	SIP96X1-44373	Popup message "SIP proxy list is empty" is displayed after changing SIP Proxy Policy from Manual to Automatic *work-around: Navigate to Admid->SIP->SIP Proxy list*
	SIP96X1-52864	Crisis Alert/call-pickup/team tone continues with auto-answer after answer
	SIP96X1-53719	J129 phone does not changed feature when replace this feature on SMRG
	SIP96X1-55691	J100 phone displays roster list incorrectly in Equinox Conferencing
	SIP96X1-58928	Exclusion green led is still ON although exclusion is not in enabled state
	SIP96X1-66398	Phone doesn't store history of a participant which are not in the same Aura system after end conference.
1-14987362572	SIP96X1-67387	Push function not working on J179
IP Office™		
	SIP96X1-35056	J169/J179 phones require additional reboot to function normally in IPO-CCMS environment if they get incorrect SIP proxy IP in previous reboot.
	SIP96X1-36922	J129 – IPO - Phone A displays "Acquiring service" and Phone B cannot login to phone A when phone B tries to log into phone A's extension

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External ID	Internal ID	Issue Description
1-14886845842	SIP96X1-63526	J100 CCMS phones with IP Office send in-band dtmf
CCElite / Contact Center		
	SIP96X1-31004	Phone does not display full AgentID when system language set to English and phone set to Arabic
	SIP96X1-40958	Phone does not display UU-info when moving line CA from phone screen to button module.
	SIP96X1-36054	LED state of Auto-in/Manual-in feature button flutters instead of flash
	SIP96X1-59045	Aux Work's LED keeps flashing although the agent-ID is logged out. *work-around: login agent-id then press the Aux work button*
	SIP96X1-66565	Phone does not remove vu-stats information follow value of displaytimeout parameter.
Open SIP		
	SIP96X1-43220	[Broadsoft] Unable to login as BroadWorks user with error of "acquiring service" when application server FQDN addre ss is given in SIP_CONTROLLER_LIST2 *work-around: Open SIP over IPv6 is not supported, use SIP_CONTROLLER_LIST*
	SIP96X1-41164	HTTP redirect to HTTPS fails certificate validation, connection fails *work-around: reconfigure phone URL to HTTPS*
	SIP96X1-45650	Phone does not use XSI SIP User ID which is set from Web UI for XSI authentication on fresh login *work-around: manually input SIP user ID*
	SIP96X1-44653	[Broadsoft] The led of some features (call unpark, call waiting, call forward,..) blinking incorrectly when phone reestablished Broadsoft XSI event channel
	SIP96X1-49886	[Broadsoft] J100 phones send cancel request for early conference if XSI is configured on phone.
	SIP96X1-49887	While creating conference call J100 phones displayed toast 'select a participant' second time after pressing 'Join' SK.
	SIP96X1-53662	[3CX] Blind transfer Server limitations Phone-A shows "Resume" soft key till the time Phone-C rings Once Phone-C answers the call , Phone-A turns to idle & the transfer completes. Transfer while ringing
	SIP96X1-53909	Red led still glow after replacing feature with application by Web pre-configuration

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External ID	Internal ID	Issue Description
	SIP96X1-57840	[Broadsoft 3PCC] No voice speech path when the phone receives the call transferred. (Meet-me conference) *work-around: Phone-B transfers (attended transfer) the call to phone-C.* *OR* *On phone-C, press "hold/resume" the call.*
	SIP96X1-59537	[PreConfigKeys] 3PCC Autodials: User customization lost if a new PHONEKEY definition is added or removed at a smaller line key
	SIP96X1-61421	Cannot add new BLF/Autodial key after removing BLF keys on Web *work-around: reset phone to default*
	SIP96X1-65303	[Broadsoft] -Phone can not log in to Sencondary server when Primary server down if receive 423 *work-around:* increase REGISTERWAIT value so phone does not need to handle 423 (brief interval).
	SIP96X1-65658	Auto-Call-Back is not working in metaswitch
	SIP96X1-66339	Phone can not register to server when configuration NAPTR record.
	SIP96X1-66593	Customize key and PHONEKEY of Multicast Paging configured on J100 phone do not display on Web UI - Key Configuration
	SIP96X1-66640	Do not to use Web UI>Environment Settings>3PCC Server Mode = Netsapiens even though it shows, use generic
	SIP96X1-66799	BLF line key isn't displayed after Guest login feature is activated
	SIP96X1-67048	[Broadsoft 3PCC]The phone is stuck while pressing Get updates if set PRIMARY_LINE_TYPE=0 in settings file
All Platforms		
	SIP96X1-29607	[DES] Auto Config prompt is not displayed if DES is manually invoked from Admin Menu
	SIP96X1-30718	Phone display message " Contact already exists" when edit ringtone
	SIP96X1-30948	Phone displays number of BCA incorrectly after moving in customize key.
	SIP96X1-33023	Unable to gracefully recover phone when lost WIFI on active call
	SIP96X1-34870	When adding a new group in Contact groups, the "NAME" prompt is not translated to some languages.
	SIP96X1-31991	Phone does not process SCEP request with new change for MYCERTCN or MYCERTURL *work-around: Remove old identity certificate (SET DELETE_MY_CERT 1) then install new cert.*
	SIP96X1-41860	J129 "Redial" softkey disappears on dialing screen after calling to phone which actived feature "CFW all"

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External ID	Internal ID	Issue Description
	SIP96X1-33445	J100 gets locked in the first time after Phone Lock Idle Time is changed to 0
	SIP96X1-37721	Phone displays restaring forever after clearing phone *work-around: restart the phone*
	SIP96X1-41905	Change Menu>Administration>IP Configration>Auto Provisioning>Service to Active and reboot fails and connects to incorrect HTTPSRVR
	SIP96X1-51611	[WML - Call pickup] Phone doesn't display main screen after exiting WML idle page
	SIP96X1-43649	[Intermittent] Speaker LED may stay on when the call is dropped by far end *work-around: reboot the phone*
	SIP96X1-64828	Phone does not respond to call pickup and Team calls while screensaver in ON
	SIP96X1-66702	Phone should add LDAP contacts to FIPS exception list when TCP is used for searching contacts
	SIP96X1-66716	J100 phone can make call on calendar event when it is receiving a Multicast Page
	SIP96X1-57525	SLAmon cannot remote control agent J159 phone.
Web User Interface		
	SIP96X1-41670	Last extension is hidden when logout user with Arabic language is changed from Web UI.
	SIP96X1-39592	Values configured from web are displayed in Phone UI source for parameter IPV4 Address
	SIP96X1-41142	Icon presence does not work after when enable presence on Web UI
	SIP96X1-42434	Web UI - Phone does not use new SIP user ID which is set on Web for XSI authentication
	SIP96X1-44209	Phone does not update configuration parameter of SCAs after importing the settings file to the WEB server
	SIP96X1-45911	Phone can not upgrade firmware on WEB UI via IPv6.
	SIP96X1-51515	Phone is rebooting after save if some parameters on wifi web page are changed
	SIP96X1-52583	Changing "No Digit Dial Timer" value on WEB UI doesn't apply immediately on phone
	SIP96X1-53165	Save Extension and show last extension do not save from WebUI *work-around:save it from Phone UI*
	SIP96X1-59350	can add certificate but cannot delete if the certificate's name has a space on Web browser.

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External ID	Internal ID	Issue Description
	SIP96X1-60930	[WebPreConfigKeys] [Firefox|Edge]Cannot delete keys after "Prevent this page from creating additional dialogs" is checked in dialog *Work-around: Ways to regain alerts after selecting the "prevent additional dialogue":* *- Mozilla firefox – Simply refresh the page* *- Microsoft edge –Open the webpage in a new tab*
	SIP96X1-61251	[WebPreConfigKeys] Label updates incorrectly after deleting Label, then updating Label on webUI *work-around: restore phone customization*
	SIP96X1-61421	[WebPreConfigKeys] Cannot add new BLF/Autodial key after removing BLF keys on Web *work-around:reset phone to defaults*
	SIP96X1-61433	Cannot relabel BLF on Web UI *work-around:use settings file*
	SIP96X1-63655	Reboot of phone is required for setting parameter SCREENSAVER_IMAGE on web UI
Wifi		
	SIP96X1-32364	Phone does not automatically connect when switching to WIFI with WEP secured method *work-around: Reboot phone, phone connects to new WIFI successfully*
Bluetooth		
	SIP96X1-53498	Phone can still perform the call from speed dial when Bluetooth device is turned off.
	SIP96X1-59557	[3PCC] BT device can't resume the transfer call after making release and answer
	SIP96X1-59637	[Intermittent - Bluetooth] Phone shows toast message "Pairing Bluetooth device interrupted. Passkey mismatch" when pairing BT headset. *work-around: Press "Back" SK to return Bluetooth menu => go back to Bluetooth devices menu again and Pair*
	SIP96X1-64785	[BT_CCMS] - BT devices work wrong when phone switch the active call.
	SIP96X1-65840	[BT - CCMS] – Phone can not drop and hold the call when phone gets the call of the bridge call appearance.
	SIP96X1-65841	[BT - CCMS] – Phone can not hear ringing tone and answer the call when phone have the call of the bridge call appearance.
	SIP96X1-66311	[BT - CCMS] – BT devices answer the unattend conference automatically.
	SIP96X1-66941	Intermittent - Phone resumes the held call automatically when there is a new call and it is canceled
Button Module (JEM24 and JBM24)		
	SIP96X1-43092	JEM24: Power cycle required message dissappears unexpectedly

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External ID	Internal ID	Issue Description
	SIP96X1-52618	JEM24 not upgrading automatically to 1.0.1.0.7 or reporting wrong JEM24 version *work-around: Power cycle the phone*
	SIP96X1-52950	[Intermittent] J169 phone detects 2 JEM24s although phone is attached 3 JEM24s
	BUTTONMODULE-426	JEM24 doesn't report upgrade failure in case of invalid fw file
	SIP96X1-66400	Phone can exit an incoming multicast page by pressing buttons on JEM24

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Appendix 1 – Supported Hardware and Minimum Software Release

J100 SIP 4.0.3.0 software is supported on the following models of IP Phones. Models may ship from the factory with a different load of software pre-installed. As such, they should be upgraded to this release on first installation.

Note: Comcodes indicated with an asterisk (*) are either end-of-sale or pending end-of-sale and include a link to the corresponding end-of-sale document.

Comcode	Short Description	Model(s)	Replaced by
700512392	J129 IP PHONE	J129D01A	700513638
700513638	J129 IP PHONE NO PWR SUPP	J129D02A	700514813
700512969	J129 IP PHONE 3PCC W/O PWR SUPP	J129D01A	700513639
700513639	J129 IP PHONE 3PCC W/CERT	J129D02A	700514814
700514813	J129 IP PHONE 5V	J129D03A	
700514814	J129 IP PHONE 5V 3PCC	J129D03A	
700513916	J139 IP PHONE	J139D01A	
700513917	J139 IP PHONE 3PCC	J139D01A	
700513918	J139 IP PHONE GSA	J139D01A	
700512394	J159 IP PHONE	J159D01A	
700512395	J159 IP PHONE GSA	J159D01A	
700512970	J159 IP PHONE 3PCC	J159D01A	
700513634	J169 IP PHONE NO PWR SUPP	J169D01A	
700513635	J169 IP PHONE GSA	J169D01A	
700513636	J169 IP PHONE 3PCC	J169D01A	
700514468	J169 IP PHONE GLBL NO PWR SUPP WH	J169D01A	
700514757	J169 IP PHONE NO BEZEL	J169D01A	
700513569	J179 IP PHONE NO PWR SUPP	J179D02A	
700513629	J179 IP PHONE GSA	J179D02A	
700513630	J179 IP PHONE 3PCC	J179D02A	
700514469	J179 IP PHONE GLBL NO PWR SUPP WH	J179D02A	

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Avaya periodically releases new hardware variations of the J100-Series IP Phones typically to address a need to change hardware components. That change may require a new version of software to support the new hardware. This then forces a minimum software release supported on that hardware. ***Attempts to downgrade these models to lower versions of software will be rejected***.

The following table provides a matrix of the different models of J100 Series IP Phones including hardware generations and any limitation on supported software version. The "Model" information can be found on the label on the outside of the shipping box, on the label on the back of the IP Phone, within the Information menus available from the screen of the phone, remotely via LLDP, remotely via the Web Interface (SIP software), and remotely via SNMP. The "Hardware Revision" can be found on the label of the box on the same line as the "Model" information, and also found on the label on the back of the phone on the first row.

Model	Hardware Revision(s)	Minimum SIP Software	Minimum H.323 Software
J129D01A	all	1.0.0.0.43	N/A
J129D02A	01 to 07	2.0.0.0.45	N/A
J129D02A	08 and greater	3.0.0.1.6	N/A
J129D03A	all	4.0.2.0.8	N/A
J139D01A	01 to 05, 07, 08	3.0.0.0.20	N/A
J139D01A	06, 09 to 11	3.0.0.1.6	N/A
J139D01A	12 and greater	4.0.0.0.21	N/A
J159D01A	all	4.0.3.0.9	6.8.3.01
J169D01A	01 to 03	1.5.0.0.15	6.7.0.02
J169D01A	04 to 07	3.0.0.1.6	6.8.0.03
J169D01B	08 and greater	4.0.0.0.21	6.8.0.03
J179D02A	01 to 03	1.5.0.0.15	6.7.0.02
J179D02A	04 to 08	2.0.0.0.45	6.7.0.02
J179D03A	09 and greater	4.0.1.0.11	6.8.2.02

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Appendix 2 – Release History

The following table provides a history of the J100 SIP software releases. The "ID" column shows the identifier of this software which is seen in the "About" menu item.

Release	ID	Date	Link to Readme file
1.0.0.0	1.0.0.0.43	Dec 2016	https://support.avaya.com/css/P8/documents/101033485
1.1.0.0	1.0.0.0.15	Mar 2017	https://support.avaya.com/css/P8/documents/101037079
1.1.0.1	1.0.0.1.3	Aug 2017	https://support.avaya.com/css/P8/documents/101042514
1.5.0.0	1.5.0.0.15	Mar 2018	http://support.avaya.com/css/P8/documents/101047039
2.0.0.0	2.0.0.0.45	April 2018	https://support.avaya.com/css/P8/documents/101048016
3.0.0.0	3.0.0.0.20	July 2018	https://support.avaya.com/css/P8/documents/101050223
3.0.0.1	3.0.0.1.6	Aug 2018	https://support.avaya.com/css/P8/documents/101051793
3.0.0.2	3.0.0.2.2	Nov 2018	https://support.avaya.com/css/P8/documents/101053115
4.0.0.0	4.0.0.0.21	Dec 2018	https://support.avaya.com/css/P8/documents/101054005
4.0.0.1	4.0.0.1.2	Mar 2019	https://support.avaya.com/css/P8/documents/101056162
4.0.1.0	4.0.1.0.11	Apr 2019	https://support.avaya.com/css/P8/documents/101056525
4.0.2.0	4.0.2.0.8	July 2019	https://support.avaya.com/css/P8/documents/101058668
4.0.2.1	4.0.2.1.3	July 2019	https://support.avaya.com/css/P8/documents/101059981
4.0.3.0	4.0.3.0.10	Oct 2019	https://support.avaya.com/css/P8/documents/101060975

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Appendix 3 – New and changed 46xxsettings.txt parameters

The latest version of the 46xxsettings.txt file can be downloaded from

https://support.avaya.com/downloads/download-details.action?contentId=C201773928555860_8&productId=P0553

New parameters.

```
## SCREENSAVER_IMAGE_SECONDARY specifies a list of screensaver images for the secondary display. The default value is "".
## This parameter is supported by:
##      J100 SIP R4.0.3.0 and later (J159 only)
##                                      Up to 5 background images are supported. Only jpeg/jpg files are supported.
##                  The maximum size of any jpeg file is 256 KB. The filenames are case insensitive.
##                  J159 secondary screen resolution is 240 pixels x 320 pixels.
##                                      The files shall be stored in the same directory defined by HTTPDIR / TLSDIR.
## SET SCREENSAVER_IMAGE_SECONDARY "secondary_screensaver_example1.jpg,secondary_screensaver_example2.jpeg"
##
## SCREENSAVER_IMAGE_DISPLAY_SECONDARY specifies the administrator choice of secondary screensaver image.
## The filename shall be one of the filenames listed in SCREENSAVER_IMAGE_SECONDARY.
## If SCREENSAVER_IMAGE_SELECTABLE_SECONDARY is set to 1 then the end user may override this setting.
## The default value is "".
## This parameter is supported by:
##      J100 SIP R4.0.3.0 and later (J159 only)
## SET SCREENSAVER_IMAGE_DISPLAY_SECONDARY secondary_screensaver_example1.jpg
##
## SCREENSAVER_IMAGE_SELECTABLE_SECONDARY specifies whether end users are allowed to choose secondary screensaver images
## (and overrides administrator choice as configured using SCREENSAVER_IMAGE_DISPLAY_SECONDARY parameter).
##  Value                Operation
##    0       End user is not allowed to choose secondary screensaver image and will not see the secondary screensaver image selection in the Settings -> Display menu.
##    1       End user is allowed to choose the secondary screensaver image from the Settings -> Display menu (Default)
## This parameter is supported by:
##      J100 SIP R2.0.0.0 and later (J159 only)
## SET SCREENSAVER_IMAGE_SELECTABLE_SECONDARY 0
##

###################### CUSTOM RING TONES ####################
##

## PROVIDE_PRIMARY_RINGTONE specifies whether the primary ringtone option is provided to the user.
##  Value  Operation
##    0    the Primary ringtone option is not provided
##    1    the Primary ringtone option is provided (default)
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##          The user option is displayed in "Settings > Audio > Personalize ringing > Primary".
##   Note: The parameter is supported in Aura and OpenSIP environments only.
## SET PROVIDE_PRIMARY_RINGTONE 0
##
## PROVIDE_RING_REMINDER_RINGTONE specifies whether the ring reminder ringtone option is provided to the user.
```

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```
## Value  Operation
##   0   the ring reminder ringtone option is not provided
##   1   the ring reminder ringtone option is provided (default)
## This parameter is supported by:
##     J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##       The user option is displayed in "Settings > Audio > Personalize ringing > Ring Reminder".
##  Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_RING_REMINDER_RINGTONE 0
##
## PROVIDE_CALL_PARK_RINGTONE specifies whether the call park ringtone option is provided to the user.
## Value  Operation
##   0   the call park ringtone option is not provided
##   1   the call park ringtone option is provided (default)
## This parameter is supported by:
##     J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##       The user option is displayed in "Settings > Audio > Personalize ringing > Call Park".
##  Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_CALL_PARK_RINGTONE 0
##
## PROVIDE_BLF_INCOMING_CALL_RINGTONE specifies whether the BLF incoming call ringtone option is provided to the user.
## Value  Operation
##   0   the BLF incoming call ringtone option is not provided
##   1   the BLF incoming call ringtone option is provided (default)
## This parameter is supported by:
##     J159, J169, J179 SIP R4.0.3.0 and later.
##       The user option is displayed in "Settings > Audio > Personalize ringing > BLF Incoming Call".
##  Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_BLF_INCOMING_CALL_RINGTONE 0
##
## PROVIDE_BLF_CALL_PARK_RINGTONE specifies whether the BLF call park ringtone option is provided to the user.
## Value  Operation
##   0   the BLF call park ringtone option is not provided
##   1   the BLF call park ringtone option is provided (default)
## This parameter is supported by:
##     J159, J169, J179 SIP R4.0.3.0 and later.
##       The user option is displayed in "Settings > Audio > Personalize ringing > BLF Call Park".
##  Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_BLF_CALL_PARK_RINGTONE 0
##
## PROVIDE_PRIORITY_ALERT_RINGTONE specifies whether the priority alert ringtone option is provided to the user.
## Value  Operation
##   0   the priority alert ringtone option is not provided
##   1   the priority alert ringtone option is provided (default)
## This parameter is supported by:
##     J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##       The user option is displayed in "Settings > Audio > Personalize ringing > Priority Alert".
##  Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_PRIORITY_ALERT_RINGTONE 0
##
## PROVIDE_ALTERNATE_NUM1_RINGTONE specifies whether the alternative number 1 ringtone option is provided to the user.
## Value  Operation
##   0   the alternative number 1 ringtone option is not provided
```

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```
##    1    the alternative number 1 ringtone option is provided (default)
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##          The user option is displayed in "Settings > Audio > Personalize ringing > Alternative Number 1".
##   Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_ALTERNATE_NUM1_RINGTONE 0
##
## PROVIDE_ALTERNATE_NUM2_RINGTONE specifies whether the alternative number 2 ringtone option is provided to the user.
##  Value  Operation
##    0    the alternative number 2 ringtone option is not provided
##    1    the alternative number 2 ringtone option is provided (default)
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##          The user option is displayed in "Settings > Audio > Personalize ringing > Alternative Number 2".
##   Note: The parameter is supported in OpenSIP environment only.
## SET PROVIDE_ALTERNATE_NUM2_RINGTONE 0
##
## RINGTONE_DEFAULT_CALL_FWD_RING specifies the index of the default ringtone to be used when an incoming call has been forwarded
from another phone.
##  Valid values 1-18. Default is 1.
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##   Note: The parameter is supported in OpenSIP environment only.
## SET RINGTONE_DEFAULT_CALL_FWD_RING 2
##
## RINGTONE_DEFAULT_PRIMARY specifies the index of the default ringtone to be used when an incoming call is received which is not
associated with any
##   other ringtone based on the call type or a contact association.
##  Valid values 1-18. Default is 1.
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##   Note: The parameter is supported in OpenSIP environment only.
## SET RINGTONE_DEFAULT_PRIMARY 2
##
## RINGTONE_DEFAULT_RING_REMINDER specifies the index of the default ringtone to be used when an incoming call is identified as a
ring reminder.
##   Applies only to OpenSIP servers which support this feature.
##  Valid values 1-18. Default is 18.
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##   Note: The parameter is supported in OpenSIP environment only.
## SET RINGTONE_DEFAULT_RING_REMINDER 2
##
## RINGTONE_DEFAULT_CALL_PARK specifies the index of the default ringtone to be used when a notification that a call has been parked
on the logged in extension.
##   Applies only to OpenSIP servers which support this feature.
##  Valid values 1-18. Default is 1.
##  This parameter is supported by:
##      J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
##   Note: The parameter is supported in OpenSIP environment only.
## SET RINGTONE_DEFAULT_CALL_PARK 2
##
```

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RINGTONE_DEFAULT_BLF_INCOMING_CALL specifies the index of the default ringtone to be used when an incoming call has been forwarded from another phone.
Applies only to OpenSIP servers which support this feature.
Valid values 1-18. Default is 1.
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only.
SET RINGTONE_DEFAULT_BLF_INCOMING_CALL 2
##
RINGTONE_DEFAULT_BLF_CALL_PARK specifies the index of the default ringtone to be used when a notification is received of a parked call on a BLF.
Applies only to OpenSIP servers which support this feature.
Valid values 1-18. Default is 1.
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only.
SET RINGTONE_DEFAULT_BLF_CALL_PARK 2
##
RINGTONE_DEFAULT_PRIORITY_ALERT specifies the index of the default ringtone to be used when an incoming priority call is received.
Valid values 1-18. Default is 15.
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only.
SET RINGTONE_DEFAULT_PRIORITY_ALERT 2
##
RINGTONE_DEFAULT_ALTERNATE_NUM1 specifies the index of the default ringtone to be used when an incoming call is from the first alternate number.
Applies only to Broadsoft servers.
Valid values 1-18. Default is 16.
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only.
SET RINGTONE_DEFAULT_ALTERNATE_NUM1 2
##
RINGTONE_DEFAULT_ALTERNATE_NUM2 specifies the index of the default ringtone to be used when an incoming call is from the second alternate number.
Applies only to Broadsoft servers.
Valid values 1-18. Default is 17.
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only.
SET RINGTONE_DEFAULT_ALTERNATE_NUM2 2
##
RINGTONE_ALERTINFO specifies a comma separated list of specific ring tone assigned to a call queue (QueueID:SelectedRingtoneID).
"QueueID" is a string matching the call queue name, as configured in the 3CX server.
"SelectedRingtoneID" is a numerical index of one of the ringtones known to the phone.
When the 3CX server is not configured for distinct ringtones per queue, the only supported QueueID is "queue".
The default is "".
This parameter is supported by:
J129, J139, J159, J169, J179 SIP R4.0.3.0 and later.
Note: The parameter is supported in OpenSIP environment only and when 3PCC_SERVER_MODE is set to 2 (3CX).
SET RINGTONE_ALERTINFO international:4 --> If INVITE contains Alert-Info...info=international then play Classic Tone 4

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SET RINGTONE_ALERTINFO q802:CustomTone1 --> If INVITE contains Alert-Info...info=q802 then play CustomTone1 as per SET RINGTONES downloaded ringtone
##
ADMIN_CHOICE_RINGTONE specifies the administrator choice of ringtone to be used for incoming calls.
The default value is "Default" which represents Avaya built in ringtone.
Otherwise, one of the ringtones name shall be configured.
User may choose to use its own ringtone.
The parameter can also be used to choose ringtone out of downloaded ringtones (using RINGTONES parameter supported in Avaya Vantage Devices SIP R1.1.0.0 and later).
This parameter is supported by:
Avaya Vantage Connect Application SIP R1.0.0.0 and later
SET ADMIN_CHOICE_RINGTONE "Titania"
##

##################### MULTICAST PAGING SETTINGS #####################
Note: The parameters are supported in Aura and OpenSIP environments.
##
MP_ENABLED specifies whether multicast paging feature is enabled or not.
Value Operation
0 Disabled (default)
1 Enabled
This parameter is supported by:
J100 SIP R4.0.3.0 and later (J129, J139, J159, J169, J179)
SET MP_ENABLED 1
##
MP_CODEC specifies codec, which is used for multicast paging transmissions.
Value Operation
1 G.729 (default)
2 G.711u
3 G.711a
This parameter is supported by:
J100 SIP R4.0.3.0 and later (J129, J139, J159, J169, J179)
SET MP_CODEC 2
##
MP_PACKET_SIZE specifies size of page RTP packets in milliseconds. Valid values 10-80. Default value is 20.
This parameter is supported by:
J100 SIP R4.0.3.0 and later (J129, J139, J159, J169, J179)
SET MP_PACKET_SIZE 10
##
MP_GROUPS_TO_LISTEN specifies a list of multicast paging groups that the phone is monitoring.
Format of a record for a group is: “IP:port:priority:label”.
IP – IP address of a multicast paging group
Port – IP port of a multicast paging group
Priority – group priority in 1-16 range. Lower numbers indicate higher priority with 1 being the highest. Priority for regular calls is hardcoded to 3.
Label – a group label.
Maximum 10 listening groups are allowed. Default value is "".
This parameter is supported by:
J100 SIP R4.0.3.0 and later (J129, J139, J159, J169, J179)
SET MP_GROUPS_TO_LISTEN "224.0.0.0:1208:1:Security,224.0.0.0:1209:4:Sales,224.0.0.0:1210:5:Team"
##
MP_GROUPS_TO_SEND specifies the list of multicast paging groups that the phone can send a page to:
Format of a record for a group is: “IP:port:label”.

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```
##  IP – IP address of a multicast paging group
##  Port – IP port of a multicast paging group
##  Label – a group label.
##  Maximum 10 listening groups are allowed. Default value is "".
##  This parameter is supported by:
##     J100 SIP R4.0.3.0 and later (J129, J139, J159, J169, J179)
## SET MP_GROUPS_TO_SEND "224.0.0.0:1208:Security,224.0.0.0:1209:Sales,224.0.0.0:1210:Team".
##

##################### DISPLAY SETTINGS #####################
##

## BACKGROUND_IMAGE_SECONDARY specifies a list of background images for the secondary display. The default value is "".
## This parameter is supported by:
##     J100 SIP R4.0.3.0 and later (J159 only)
##                                  Up to 5 background images are supported. Only jpeg/jpg files are supported.
##                The maximum size of any jpeg file is 256 KB. The filenames are case insensitive.
##                J159 secondary screen resolution is 240 pixels x 320 pixels.
##                                  The files shall be stored in the same directory defined by HTTPDIR / TLSDIR.
## SET BACKGROUND_IMAGE_SECONDARY "secondary_background_example1.jpg,secondary_background_example2.jpeg"
##
## BACKGROUND_IMAGE_DISPLAY_SECONDARY specifies the administrator choice of secondary background image.
## The filename shall be one of the filenames listed in BACKGROUND_IMAGE_SECONDARY.
## If BACKGROUND_IMAGE_SELECTABLE_SECONDARY is set to 1 then the end user may override this setting.
## The default value is "".
## This parameter is supported by:
##     J100 SIP R4.0.3.0 and later (J159 only)
## SET BACKGROUND_IMAGE_DISPLAY_SECONDARY secondary_background_example1.jpg
##
## BACKGROUND_IMAGE_SELECTABLE_SECONDARY specifies whether end users are allowed to choose secondary background images
## (and overrides administrator choice as configured using BACKGROUND_IMAGE_DISPLAY_SECONDARY parameter).
##  Value              Operation
##   0      End user is not allowed to choose secondary background image and will not see the secondary background image selection in
the Settings -> Display menu.
##   1      End user is allowed to choose the secondary background image from the Settings -> Display menu (Default)
## This parameter is supported by:
##     J100 SIP R4.0.3.0 and later (J159 only)
## SET BACKGROUND_IMAGE_SELECTABLE_SECONDARY 0
```

Changed parameters.

```
###############################################################################
##
##                      COMMON SETTINGS
##
## Settings in this section will be processed by all telephones,
## but not all parameters are supported by all telephones or all software releases.
## Settings for parameters that are not supported will be ignored.
## For more information, see the Administrator's Guide available at support.avaya.com
```

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```
##
###############  LAYER 2 VLAN AND QOS SETTINGS  ##############
##
## L2Q specifies whether layer 2 frames generated by the telephone will have IEEE 802.1Q tags.
##  Value  Operation
##    0    Auto - frames will be tagged if the value of L2QVLAN is non-zero (default).
##    1    On - frames will always be tagged.
##    2    Off - frames will never be tagged.
##  Note: This parameter may also be set via DHCP or LLDP.
##  This parameter is supported by:
##       J100  SIP  R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
- if L2QVLAN == 0, L2Q is treated as 2 (disabled).
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later. Note: Value 1 has the same behavior as
value 0.
##       J169/J179 SIP R1.5.0 - if L2QVLAN == 0, L2Q is treated as 2 (disabled).
##       J129  SIP  R1.0.0.0 (or R1.1.0.0)
##       H1xx  SIP  R1.0 and later. Note: Value 1 has the same behavior as value 0.
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later; R7.1.0.0 and later, if L2QVLAN == 0, L2Q is treated as 2
(disabled).
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET L2Q 0
##
## L2QVLAN specifies the voice VLAN ID to be used by IP telephones.
##  Valid values are 0 through 4094; the default value is 0.
##  Note: This parameter may also be set via DHCP or LLDP.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET L2QVLAN 5
##
## L2QAUD specifies the layer 2 priority value for audio frames generated by the telephone.
##  Valid values are 0 through 7; the default value is 6.
##  Note: This parameter may also be set via LLDP and H.323 signaling,
##         which would overwrite any value set in this file.
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.1.0.0 and later (This parameter may also be set
via AADS or LLDP which would overwrite any value set in this file)
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx  SIP  R1.0 and later
```

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```
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET L2QAUD 7

## L2QSIG specifies the layer 2 priority value for signaling frames generated by the telephone.
##  Valid values are 0 through 7; the default value is 6.
##  Note: This parameter may also be set via LLDP or H.323 signaling,
##         which would overwrite any value set in this file.
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.1.0.0 and later (This parameter may also be set
via AADS or LLDP which would overwrite any value set in this file).
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET L2QSIG 7
##
## VLANSEP specifies whether VLAN separation will be enabled by the built-in Ethernet switch
##  while the telephone is tagging frames with a non-zero VLAN ID. When VLAN separation is
enabled,
##  only frames with a VLAN ID that is the same as the VLAN ID being used by the telephone
##  (as well as priority-tagged and untagged frames) will be forwarded to the telephone.
##  Also, if the value of PHY2VLAN (see below) is non-zero, only frames with a VLAN ID that is
##  the same as the value of PHY2VLAN (as well as priority-tagged and untagged frames) will be
##  forwarded to the secondary (PHY2) Ethernet interface, and 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 (see below).
##  Value  Operation
##    0    Disabled.
##    1    Enabled if L2Q, L2QVLAN and PHY2VLAN are set appropriately (default).
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169, J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##      Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch; see comments for H1xx  SIP  R1.0 and later. All K155 devices have embedded Ethernet
switch.
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       H1xx  SIP  R1.0 and later; VLAN separation supported on H1xx have the following
exceptions:
##        1. Priority-tagged and untagged frames from the network port will be forwarded to the
PC port only when VLANSEP==1,
```

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```
##           H1xx sends tagged packets (L2Q==0 or 1 and VLANTEST==0 or timer < VLANTEST) and
L2QVLAN<>0, else to both phone and PC ports.
##        2. No enforcement of PHY2VLAN and PHY2PRIO on tagged VLAN packets received from PC
port. If VLANSEP==1,
##           H1xx sends tagged packets (L2Q==0 or 1 and VLANTEST==0 or timer < VLANTEST) and
0<>PHY2VLAN<>L2QVLAN<>0 then:
##           a. Untagged packets from PC port will be tagged with PHY2VLAN and priority==0.
##           b. Tagged packets will be forwarded as tagged packets only if their VLAN equal to
PHY2VLAN.
##              Otherwise the packets from PC will be sent unmodified.
##           Only in case of VLANSEP==1,H1xx sends tagged packets (L2Q==0 or 1 and VLANTEST==0 or
timer < VLANTEST) and 0<>PHY2VLAN<>L2QVLAN<>0,
##           there will be full separation between PC and phone traffic. In all other cases, PC
traffic can reach the phone.
##        3. When VLANSEP ==0, H1xx sends untagged packets even if L2Q==0 or 1 and VLANTEST==0 or
timer < VLANTEST.
##       16xx H.323 R1.0 and later
## SET VLANSEP 0
##
## VLANSEPMODE 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. This VLAN separation is
enabled when:
##  VLANSEP=1, L2QVLAN<> PHY2VLAN (and both has value different than 0), L2Q is auto (0) or (1)
tagging.
##  In this new VLAN separation scheme:
##  - Untagged packets from PC port will be forwarded to network port only as untagged packets.
##  - Tagged packets from PC port will be forwarded to network port only as tagged packets only
in case
##    their VLAN is equal to PHY2VLAN.
##    In this mode, tagged and untagged packets from  PC port will never reach phone's port.
##  - Untagged packets from the network will be sent to the PC port only.
##  - Tagged packets from the network port will be sent to the PC port if their VLAN is equal to
PHY2VLAN
##    and to the phone if their VLAN is equal to L2QVLAN.
##  - 802.1x/LLDP and Spanning tree packets are supported as in previous releases in this new
mode.
##  When VLANSEPMODE is 0, then the VLAN separation is based on previous releases where untagged
packets
##  from PC port can reach the phone.
##  Please note that PHY2PRIO is NOT supported when VLANSEPMODE is 1.
##  Value  Operation
##    0    Disabled
##    1    Enabled
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later, Default is 0.
##       J169/J179 SIP R1.5.0 , J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later, Default is 0.
##       96x1  SIP  R7.1.0.0 and later, Default is 0.
##       96x1 H.323 R6.6 and later, Default is 0.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later (J129 only); Default is
1. VLANSEP is not supported by J129. The conditions for VLAN separation mode are
##             as described above (except no support for VLANSEP). If one the conditions is not
fulfilled then J129
##              will get any tagged/untagged unknown/broadcast/multicast/known DA equal to CPU
MAC address packets from the network or PC port.
## SET VLANSEPMODE 1
##
```

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```
## PHY2VLAN specifies the VLAN ID to be used by frames forwarded to and from the secondary
##  (PHY2) Ethernet interface when VLAN separation (see VLANSEP above) is enabled.
##  Valid values are 0 through 4094; the default value is 0.
##  Note: This parameter may also be set via LLDP.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch, All K155 devices have embedded Ethernet switch.
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET PHY2VLAN 1
##
## PHY2PRIO specifies the layer 2 priority value to be used for frames received on the secondary
##  (PHY2) Ethernet interface when VLAN separation (see VLANSEP above) is enabled.
##  Valid values are 0 through 7; the default value is 0.
##  The parameter is not supported when VLANSEPMODE is 1.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET PHY2PRIO 2
##
## PHY2TAGS specifies whether or not tags will be removed
##  from frames forwarded to the secondary (PC) Ethernet interface.
##  Value  Operation
##    0    Tags will be removed (default)
##    1    Tags will not be removed
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch, All K155 devices have embedded Ethernet switch.
##       H1xx SIP R1.0 and later
##       96x1 SIP R6.3 and later
##       96x1 H.323 R6.6 and later
## SET PHY2TAGS 1
##
####################  LAYER 3 QOS SETTINGS  ##################
##
## DSCPAUD specifies the layer 3 Differentiated Services (DiffServ) Code Point
##  for audio frames generated by the telephone.
##  Valid values are 0 through 63; the default value is 46.
##  Note: This parameter may also be set via LLDP or H.323 signaling,
##         which would overwrite any value set in this file.
```

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```
##  This parameter is supported by:
##       Avaya Equinox R3.5.5 and later used in IP office environment only (for Aura environment
##             DSCPAUD is taken from PPM and configured using SMGR). The parameter can be
configured from AADS and LLDP which have higher precedence than this file.
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later; used in IP office and OpenSIP
environments only (for Aura environment
##             DSCPAUD is taken from PPM and configured using SMGR). The parameter can be
configured from AADS and LLDP which have higher precedence than this file.
##       H1xx SIP R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET DSCPAUD 43

## DSCPSIG specifies the layer 3 Differentiated Services (DiffServ) Code Point
##  for signaling frames generated by the telephone.
##  Valid values are 0 through 63; the default value is 34.
##  Note: This parameter may also be set via LLDP or H.323 signaling,
##         which would overwrite any value set in this file.
##  This parameter is supported by:
##       Avaya Equinox R3.5.5 and later used in IP office environment only (for Aura environment
##             DSCPAUD is taken from PPM and configured using SMGR). The parameter can be
configured from AADS and LLDP which have higher precedence than this file.
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later; used in IP office and OpenSIP
environments only (for Aura environment
##             DSCPSIG is taken from PPM and configured using SMGR). The parameter can be
configured from AADS and LLDP which have higher precedence than this file.
##       H1xx SIP R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET DSCPSIG 41
##
######################  CALL QUALITY INDICATION SETTINGS  ######################
##
## WBCSTAT and QLEVEL_MIN configuration parameters related to the LOCAL network quality (MAY not
be end to end indication).
##
## WBCSTAT specifies whether a wideband codec indication will be displayed when a wideband codec
is being used
##  Value  Operation
##    0    Disabled
##    1    Enabled (default)
##  This parameter is supported by:
```

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```
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.4 and later
##       96x1  SIP  R6.4 and later
##       H1xx  SIP  R1.0 and later
## SET WBCSTAT 0
##
## QLEVEL_MIN specifies the minimum quality level for which a low local network quality
indication will not be displayed
##  Value  Operation
##    1    Never display icon (default)
##    2    Packet loss is > 5% or round trip network delay is > 720ms or jitter compensation
delay is > 160ms
##    3    Packet loss is > 4% or round trip network delay is > 640ms or jitter compensation
delay is > 140ms
##    4    Packet loss is > 3% or round trip network delay is > 560ms or jitter compensation
delay is > 120ms
##    5    Packet loss is > 2% or round trip network delay is > 480ms or jitter compensation
delay is > 100ms
##    6    Packet loss is > 1% or round trip network delay is > 400ms or jitter compensation
delay is > 80ms
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.4 and later
##       96x1  SIP  R6.4 and later
##       H1xx  SIP  R1.0 and later
## SET QLEVEL_MIN 4
##
######################  DHCP SETTINGS  ########################
##
## DHCPSTD specifies whether DHCP will comply with the IETF RFC 2131 standard and
##  immediately stop using an IP address if the lease expires, or whether it will
##  enter an extended rebinding state in which it continues to use the address and
##  to periodically send a rebinding request, as well as to periodically send an
##  ARP request to check for address conflicts, until a response is received from
##  a DHCP server or until a conflict is detected.
##  Value  Operation
##    0    Continue using the address in an extended rebinding state (default).
##    1    Immediately stop using the address.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET DHCPSTD 1
##
```

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```
## DHCPSTDV6 specifies whether DHCPv6 will comply with the IETF RFC 3155 standard and immediately
stop using
## an IPv6 address if the address valid lifetime expires, or whether it will enter an extended
rebinding state
## in which it continues to use the address and to periodically send a rebinding request, as well
as to periodically send
## a NS (Neighbor Solicitation) request to check for address conflicts, until a REPLY response is
received from
## a DHCPv6 server (either a new address, or zero lifetimes, or error status codes) or until a
DAD conflict is detected.
## If the address is duplicated, DHCPv6 client transitions into STOPPED state and the phone
reboots.
##  Value  Operation
##    0    Enter proprietary extended rebinding state (continue to use IPv6 address , if DHCPv6
lease expires) (default)
##    1    Comply with DHCPv6 standard (immediately release IPv6 address, if DHCPv6 lease
expires)
##  This parameter is supported by:
##       J100  SIP  R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
##  SET DHCPSTDV6 1
##
## VLANTEST specifies the number of seconds that DHCP will be attempted with a
##  non-zero VLAN ID before switching to a VLAN ID of zero (if the value of L2Q is 1)
##  or to untagged frames (if the value of L2Q is 0).
##  Valid values are 0 through 999; the default value is 60.
##  A value of zero means that DHCP will try with a non-zero VLAN ID forever.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later (only J169/J179), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later - if L2QVLAN == 0, L2Q is treated as 2 (disabled).
##       Avaya Vantage Devices SIP R1.0.0.0 and later. Note: L2Q==1 has the same behavior as
L2Q==0.
##       J129  SIP  R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later. Note: L2Q==1 has the same behavior as L2Q==0.
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later. R7.1.0.0 and later, if L2QVLAN == 0, L2Q is treated as 2
(disabled).
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET VLANTEST 90
##
## REUSETIME specifies the number of seconds that DHCP will be attempted with a VLAN ID of
##  zero (if the value of L2Q is 1) or with untagged frames (if the value of L2Q is 0 or 2)
##  before reusing the IP address (and associated address information) that it had the last
##  time it successfully registered with a call server, if such an address is available.
##  While reusing an address, DHCP will enter 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 (i.e., no reuse).
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##          J100 SIP R4.0.0.0 and later: REUSETIME specifies the number of seconds that DHCPV4 or
DHCPv6 discovery will be attempted before either:
```

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```
##              reusing the previously cached value of IPv4 address (and associated address
information) that the Phone had the last time successfully
##              registered with a call server, if such an address is available, or continue
discovery DHCPv6 server: IPv6 does not support reuse,
##              so there is no corresponding parameter to IPv4's REUSE_IPADD.
##              While reusing an address, DHCPV4 will enter the extended rebinding state
described for DHCPSTD.
##              A value of zero means that DHCP or DHCPv6 will be tried forever (i.e., no reuse).
##       H1xx  SIP  R1.0 and later (REUSE mechanism is supported on Ethernet interface only (not
Wi-Fi))
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R2.5 and later
## SET REUSETIME 90
##
######################  DNS SETTINGS  ######################
##
## DNSSRVR specifies a list of DNS server addresses.
##  Addresses can be in dotted-decimal (IPv4) or colon-hex (IPv6, if supported)
##  format, separated by commas without any intervening spaces.
##  A value set in this file will replace any value set for DNSSRVR via DHCP.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET DNSSRVR 198.152.15.15
##
## DOMAIN 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 ("").
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET DOMAIN mycompany.com
##
```

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```
##
#####################  LOGIN SETTINGS  ######################
##

## SHOW_LAST_EXTENSION specifies whether extension is presented after logout.
##  Value  Operation
##    0    Extension is not presented after logout (Default)
##    1    Extension is presented after logout
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET SHOW_LAST_EXTENSION 1

##################  LDAP/S CLIENT CONFIGURATION ################
##
## DIRENABLED_PLATFORM specifies whether the Avaya Vantage LDAP/S client is enabled or disabled.
##  Value  Operation
##    0    LDAP/S client is disabled (default)
##    1    LDAP/S client is enabled
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R2.0.1.0 and later
## SET DIRENABLED_PLATFORM 1
##

##
## DIRSRVR specifies the IP address or fully qualified domain name of LDAP/S server. The default
is "".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R2.0.1.0 and later
## SET DIRSRVR ldap.customer.com
##
## DIRSRVRPRT specifies the port number of the LDAP/S server.
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later, the default is 389.
##       Avaya Vantage Devices SIP R2.0.1.0 and later; the default is 636.
## SET DIRSRVRPRT 640
##
## DIRTOPDN specifies the LDAP search base. The default is "".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R2.0.1.0 and later
## SET DIRTOPDN "OU=Global Users,dc=global,dc=company,dc=com"
##
## DIRSECURE specifies whether to use TLS or TCP for LDAP/S.
##  For server authentication, the trusted certificate file shall be placed on an HTTP or HTTPS
server and include certificate file name in the TRUSTCERTS parameter value.
##  Client authentication uses identity certificate installed using SCEP or PKCS12.
##  Value  Operation
##    0    LDAP over TCP
##    1    LDAPS over TLS (default)
##  This parameter is supported by:
```

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```
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R2.0.1.0 and later
## SET DIRSECURE 0
##

##
## DIRSEARCH_FIELDS specifies the LDAP attributes that will be searched. The default is
"cn,sn,telephoneNumber".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
## SET DIRSEARCH_FIELDS "givenName,mail,middle
initials,telephoneNumber,sn,mobile,o,department,Rank,office,DoD SIP URI".
##
## DIRSHOW_FIELDS specifies the LDAP attributes to show in the detail view. This parameter is
also used for mapping specified LDAP attributes to user-friendly display fields.
##  The format of each DIRSHOW_FIELDS parameter shall be as follows:
##    SET DIRSHOW_FIELDS "[LDAP Attributes]=[Field Names],[LDAP Attribute 1]=[Field Name1]"
##  The default is "cn,sn,telephoneNumber,mail".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
## SET DIRSHOW_FIELDS "dn=Distinguished Name,rank,gn=First Name,office=Office,middle
initials=Middle Initial,Display Name=Full Name,sn=Last Name,job title=Job,cn=Common
Name,o=Office,c=Country,department=Department,street=Street,mail=Mail Box,l,telephoneNumber=Phone
number,st,mobile=Mobile,postalCode=Postal code,facsimileTelephoneNumber=Fax,DoD SIP URI=Number"
##
## DIRNAME_FIELDS specifies the LDAP attributes used for name in the search results and the order
they will be displayed. The default is "cn".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
## SET DIRNAME_FIELDS "cn,sn".
##
## DIRNUMBER_FIELDS specifies the LDAP attributes that contain a callable number. The primary
number is the first attribute listed in this config.
## The default is "telephoneNumber".
##  This parameter is supported by:
##       J139, J159, J169, J179 SIP R4.0.3.0 and later
## SET DIRNUMBER_FIELDS "telephoneNumber,mobile,DoD SIP URI"
##

#########  GUEST LOGIN (AND VISITING USER) SETTINGS   #########
##
## GUESTLOGINSTAT specifies whether the Guest Login feature is available to users.
##  Value  Operation
##    0    Guest Login feature is not available to users (default)
##    1    Guest Login feature is available to users
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.0 and later releases
##       96x0 H.323 R2.0 and later releases
## SET GUESTLOGINSTAT 0
##
## GUESTDURATION specifies the duration (in hours) before a Guest Login or a
##  Visiting User login will be automatically logged off if the telephone is idle.
##  Valid values are integers from 1 to 12, with a default value of 2.
```

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```
##  Note: Visiting user feature in this context related to H.323 endpoints using VUMCIPADD.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.0 and later releases
##       96x0 H.323 R2.0 and later releases
## SET GUESTDURATION 2
##
## GUESTWARNING specifies the number of minutes before time specified by GUESTDURATION that
##  a warning of the automatic logoff is initially presented to the Guest or Visiting User.
##  Valid values are integers from 1 to 15, with a default value of 5.
##  Note: Visiting user feature in this context related to H.323 endpoints using VUMCIPADD.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
##       96x1 H.323 R6.0 and later releases
##       96x0 H.323 R2.0 and later releases
## SET GUESTWARNING 5
##


#################  SERVER SETTINGS (SIP)  ################
## Note: Third party SIP call controllers (3PCC) support is only provided by J129 SIP R1.1.0.0,
J100 SIP R2.0.0.0 and later, Avaya Vantage Connect R2.0.1.0 and later.
##
## SIPDOMAIN specifies the domain name to be used during SIP registration.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; not applicable when Avaya Vantage Open
application is used.
##       Avaya Vantage Connect Application SIP R1.0.0.0 and later; The configuration file from
the Avaya Vantage Device
##       include the highest precedence value from the following sources (High to low): UI, AADS,
this file and PPM.
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET SIPDOMAIN example.com

## SIP_CONTROLLER_LIST specifies a list of IPv4 SIP controller designators,
##  separated by commas without any intervening spaces.
##  The list is used on IPv4-only and dual mode phones (if SIP_CONTROLLER_LIST_2 is not
provided).
##  Each controller designator has the following format:
##  host[:port][;transport=xxx]
##  host is an IP address in dotted-decimal (DNS name format is not supported unless stated
otherwise below).
##  [:port] is an optional port number.
##  [;transport=xxx] is an optional transport type where xxx can be tls, tcp or udp.
##  If a port number is not specified a default value of 5060 for TCP and UDP or 5061 for TLS is
used.
##  If a transport type is not specified, a default value of tls is used.
```

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```
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0); J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; DNS name format is supported for 3PCC environment only.
##         For 3PCC environment, only one SIP controller is supported.
##       J100 SIP R4.0.0.0 and later; used on dual mode phones if SIP_CONTROLLER_LIST_2 is not
provided.
##          When 3PCC_SERVER_MODE = 1 (a BroadSoft server), SIP_CONTROLLER_LIST should contain
one sip controller entry and host should be an FQDN (DNS name format).
##          The FQDN would resolve to primary and alternate servers to support redundant
configuration.
##          When 3PCC_SERVER_MODE = 0 (a generic SIP server), SIP_CONTROLLER_LIST may contain one
or two sip controller entries (to support redundant configuration).
##          "host" of sip controller entry could be an FQDN(DNS name format) or an IP address. If
FQDN is provided, it will resolve to one primary server.
##          IPv6 is not supported for 3PCC environment. In 3PCC environment, there is no support
for resolving an FQDN to an IPv6 address in the SIP_CONTROLLER_LIST or SIP_CONTROLLER_LIST_2.
##          IPv6 is supported for Aura environment and in Aura there is no support for FQDN yet
(only IP addresses can be configured).
##       J169/J179 SIP R1.5.0
##       Avaya Equinox 3.1.2 and later; DNS name format is supported.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; DNS name format is supported; UDP is not
supported; not applicable when Avaya Vantage Open application is used.
##       Avaya Vantage Connect Application SIP R1.0.0.0 and later; DNS name format is supported;
UDP is not supported. The
##         configuration file from the Avaya Vantage Device combines the configuration of this
parameter from all sources (in the following order):
##         UI, LLDP, DHCP, this file, PPM and AADS.
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.4.1 and later
##       H1xx SIP R1.0 and later; udp is not supported.
## SET SIP_CONTROLLER_LIST proxy1:5555;transport=tls,proxy2:5556;transport=tls
##
## SIP_CONTROLLER_LIST_2
## Valid Values
##    String The comma separated list of SIP proxy/registrar servers
##    0 to 255 characters: zero or more IP addresses in dotted decimal or colon-hex format,
##    separated by commas without any intervening spaces.
##    Default: "" (null)
## Description
##     This parameter replaces SIP_CONTROLLER_LIST for dual mode phones. It is used on IPv6-only
phones to provide the list of SIPv6 servers.
##     SIPv4 servers are ignored in IPv6-only mode. It is used to select the registration
address.
##     The list has the following format: host[:port][;transport=xxx]
##     where:
##         - host: is an IP addresses in dotted-decimal format or hex format
##         - port: is the optional port number. If a port number is not specified the default
##                 value (5060 for TCP, 5061 for TLS) will be used
##         - transport: is the optional transport type (where xxx is tls or tcp)
##                      If a transport type is not specified the default value TLS will be used
##     A dual mode controller has addresses of both families within curly brackets.
##     A settings file example is:
##           SIP_CONTROLLER_LIST_2
"{[2007:7::5054:ff:fe35:c6e]:5060;transport=tcp,47.11.15.142:5060;transport=tcp},
##
{[2007:7::5054:ff:fe80:d4b0]:5060;transport=tcp,47.11.15.174:5060;transport=tcp}"
```

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```
##     Dual mode phones use SIGNALING_ADDR_MODE to select SM IP addresses from
SIP_CONTROLLER_LIST_2.
##     If SIGNALING_ADDR_MODE is 4, register to the first IPv4 address in SIP_CONTROLLER_LIST_2.
##     IPv4 only phones use SIP_CONTROLLER_LIST. Dual mode phones use SIP_CONTROLLER_LIST if
SIP_CONTROLLER_LIST_2 is not provided.
##     SIP_CONTROLLER_LIST_2 should only be used if IPv6 addresses (FQDN is not supported) may be
used for SIP signaling.
##     SIP_CONTROLLER_LIST_2 should not be used if FQDN (DNS name format) is used for sip
controllers.
##
## This parameter is supported by:
##      J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## Example:
## Dual mode SIP controllers:
## SET SIP_CONTROLLER_LIST_2
"{[2007:7::5054:ff:fe35:c6e]:5060;transport=tcp,47.11.15.142:5060;transport=tcp},
## {[2007:7::5054:ff:fe80:d4b0]:5060;transport=tcp, 47.11.15.174 :5060;transport=tcp}"
## IPv6-only mode SIPv6 controllers:
##  SET SIP_CONTROLLER_LIST_2
"[2007:7::5054:ff:fe35:c6e]:5060;transport=tcp,[2007:7::5054:ff:fe80:d4b0]:5060;transport=tcp"
##
## SIP Transport UDP
## Determines whether SIP Transport = UDP can be manually configured on the phone.
##     0 for No (default)
##     1 for Yes
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET ENABLE_UDP_TRANSPORT 1
##
## SIPREGPROXYPOLICY specifies whether the telephone will attempt to maintain
##  one or multiple simultaneous registrations.
##      Value       Operation
##    alternate     Only a single registration will be attempted and maintained.
##   simultaneous   Simultaneous registrations will be attempted and maintained with all
available controllers.
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 or R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later,
J159 SIP R4.0.3.0 and later, the default is simultaneous.
##              The parameter shall be configured to "alternate" in IP Office and 3PCC
environments only.
##       Not supported in 96x1 SIP R6.2 and later;    the default value is simultaneous.
##       96x1 SIP R6.0.x;           the default value is alternate.
##       96x0 SIP R2.4.1 and later; the default value is alternate.
## SET SIPREGPROXYPOLICY simultaneous
##
## SIMULTANEOUS_REGISTRATIONS specifies the number of Session Managers
##  with which the telephone will simultaneously register.
##  Valid values are 1, 2 or 3; the default value is 3.
##  This parameter is supported by:
##       Avaya Equinox 3.4 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later
##       96x1 SIP R6.0 and later
```

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```
##       96x0 SIP R2.6 and later
##       H1xx SIP R1.0 and later; For IP office environment this parameter shall be set to 1.
## SET SIMULTANEOUS_REGISTRATIONS 3
##
## CONNECTION_REUSE specifies whether the telephone will use two UDP/TCP/TLS connection (for both
outbound
## and inbound) or one UDP/TCP/TLS connection.
##  Value  Operation
##    0 - disabled, the phone will open oubound connection to the SIP Proxy and listening socket
for inbound connection
##        from SIP proxy in parallel. This is the only and default behavior for pre-6.4 releases.
##    1 - enabled, the phone will not open a listening socket and will maintain and re-use the
sockets it creates with
##        the outbound proxies (default)
##  For IP office environment this parameter shall be set to 1 (default).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later - only value 1 is supported.
##       96x1 SIP R6.4 and later up to R7.1.0.0 (excluded) - values 0 and 1 are supported,
R7.1.0.0 and later only value 1 is supported.
##       H1xx SIP R1.0 and later
## SET CONNECTION_REUSE 0
##
## ENABLE_PPM_SOURCED_SIPPROXYSRVR parameter enables PPM as a source of SIP proxy server
information.
##  Value  Operation
##    0    Proxy server information received from PPM will not be used.
##    1    Proxy server information received from PPM will be used (default).
##  This parameter is not supported in IP Office environment as PPM is not supported.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.4.1 and later
##       H1xx SIP R1.0 and later
## SET ENABLE_PPM_SOURCED_SIPPROXYSRVR 1
##


## CONFIG_SERVER_SECURE_MODE specifies whether HTTP or HTTPS is used to access the configuration
server.
##  Value  Operation
##    0    use HTTP  (default for 96x0 R2.0 through R2.5)
##    1    use HTTPS (default for other releases and products)
##    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.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.0 and later
## SET CONFIG_SERVER_SECURE_MODE 1
##
## VOLUME_UPDATE_DELAY specifies the minimum interval, in seconds, between backups of the volume
levels to PPM service
```

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```
##  when the phone registered to Avaya Aura Session Manager. If no change to volume levels, there
will be no backup to PPM service.
##  Valid values are 2 through 900; the default value is 2.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.0.1 and later
## SET VOLUME_UPDATE_DELAY 20

## ENABLE_AVAYA_ENVIRONMENT specifies whether the telephone is configured
##  for use in an Avaya (SES) or a third-party proxy environment.
##  Value  Operation
##    0    3rd party proxy with "SIPPING 19" features
##    1    Avaya SES with AST features and PPM (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; for IP office and 3PCC environments this parameter shall be
set to 0.
##       J169/J179 SIP R1.5.0
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later; for IP office environment this parameter shall be set to 0.
##       96x0 SIP R1.0 through R2.4
## SET ENABLE_AVAYA_ENVIRONMENT 1
##
## SMGR_AUTO_FAVORITE specifies whether all the Avaya Aura System Manager (SMGR) configured
Features and supported Autodials, regardless of them being marked as Favorite or not, are put
onto the J159, J169 or J179 Phone screen.
##  This will only be applied to the phone if the phone is connected to Aura 8.0.0.0 or earlier.
##  Value  Operation
##    0    do not auto favorite (default)
##    1    auto favorite
##  This parameter is supported by:
##       J100  SIP  R4.0.0.1 and later (J169/J179 only), J159 SIP R4.0.3.0 and later
## SET SMGR_AUTO_FAVORITE 1
##
#########  NON-AVAYA ENVIRONMENT SETTINGS (SIP ONLY)  ########
##

## DIALPLAN specifies the dial plan used in the telephone.
##  It 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 ("").
##  See the telephone Administrator's Guide for format and setting alternatives.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.0 and later
##       H1xx SIP R1.0 and later
## SET DIALPLAN [23]xxxx|91xxxxxxxxx|9[2-9]xxxxxxxxx
##
## PHNNUMOFSA specifies the number of Session Appearances the telephone
##  should support while operating in a non-Avaya environment.
##  Valid values are 1 through 10; the default value is 3.
##  This parameter is supported by:
```

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```
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.0 and later
##       H1xx SIP R1.0 and later
## SET PHNNUMOFSA 3
##
##################  TIME SETTINGS (SIP ONLY) #################
##
## SNTPSRVR 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; the default value is null ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##             the default is changed to
"0.avaya.pool.ntp.org,1.avaya.pool.ntp.org,2.avaya.pool.ntp.org,3.avaya.pool.ntp.org" in R2.0.0.0
and later.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; the default is changed to
"0.avaya.pool.ntp.org,1.avaya.pool.ntp.org,2.avaya.pool.ntp.org,3.avaya.pool.ntp.org" in R2.0.0.0
and later. FQDN is supported in R2.0.0.0 and later.
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET SNTPSRVR 192.168.0.5
##
## SNTP_SYNC_INTERVAL specifies the time interval in minutes at which the phone will attempt to
synchronize its time with configured NTP servers.
## Valid values: 60-2880 (minutes), Default: 1440 minutes (1 day).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET SNTP_SYNC_INTERVAL 100
##
## GMTOFFSET 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). The default value is 0:00.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx SIP R1.0 only (TIMEZONE shall be used in R1.0.0.1 and later)
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
## SET GMTOFFSET 0:00
##
## DSTOFFSET 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.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx SIP R1.0 only (TIMEZONE shall be used in R1.0.0.1 and later)
##       96x1 SIP R6.0 and later
```

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```
##       96x0 SIP R1.0 and later
## SET DSTOFFSET 1
##
## DSTSTART specifies when to apply the offset for daylight savings time.
##  The default value for all telephones is 2SunMar2L
##   (the second Sunday in March at 2AM local time).
##  See the Administrator's Guide for format and setting alternatives.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx SIP R1.0 only (TIMEZONE shall be used in R1.0.0.1 and later)
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
## SET DSTSTART 2SunMar2L
##
## DSTSTOP specifies when to stop applying the offset for daylight savings time.
##  The default value for all telephones is 1SunNov2L
##   (the first Sunday in November at 2AM local time).
##  See the Administrator's Guide for format and setting alternatives.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       H1xx SIP R1.0 only (TIMEZONE shall be used in R1.0.0.1 and later)
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
## SET DSTSTOP 1SunNov2L
##

##################  TIMER SETTINGS (SIP ONLY)  ################
##
## WAIT_FOR_REGISTRATION_TIMER specifies the number of seconds that the telephone will wait
##  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; the default value is 32.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.5 and later
##  Note: For Avaya Distributed Office configurations prior to R2.0, this parameter must be set
to 60.
## SET WAIT_FOR_REGISTRATION_TIMER 60
##
## REGISTERWAIT specifies the number of seconds between re-registrations with the current server.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; valid values are 30 to 86400; the default value is 900.
##       Avaya Vantage Connect Application SIP R1.1.0.0 and later
##      J169/J179 SIP R1.5.0; valid values are 30 to 86400; the default value is 900.
##       96x1 SIP R6.0 and later; valid values are 30 to 86400; the default value is 900.
##       H1xx SIP R1.0 and later; valid values are 30 to 86400; the default value is 900.
##       96x0 SIP R2.4.1 and later; valid values are 30 to 86400; the default value is 900.
##       96x0 SIP R1.0 through R2.2; valid values are 10 to 1000000000; the default value is
3600.
```

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```
## SET REGISTERWAIT 1000
##

## WAIT_FOR_UNREGISTRATION_TIMER specifies the number of seconds that the telephone will wait
##  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; the default value is 32.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.5 and later
## SET WAIT_FOR_UNREGISTRATION_TIMER 45
##
## WAIT_FOR_INVITE_RESPONSE_TIMEOUT specifies the maximum number of seconds that the
##  telephone will wait for another response after receiving a SIP 100 Trying response.
##  Valid values are 30 through 180; the default value is 60.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later.
##       H1xx SIP R1.0 and later.
## SET WAIT_FOR_INVITE_RESPONSE_TIMEOUT 90
##
## OUTBOUND_SUBSCRIPTION_REQUEST_DURATION specifies the duration in seconds requested by the
##  telephone in SUBSCRIBE messages, which may be decreased in the response from the server.
##  Valid values are 60 through 31536000 (one year); the default value is 86400 (one day).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later.
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.0 and later.
## SET OUTBOUND_SUBSCRIPTION_REQUEST_DURATION 604800
##
## NO_DIGITS_TIMEOUT specifies the number of seconds that the telephone will wait
##  for a digit to be dialed after going off-hook before generating a warning tone.
##  Valid values are 1 through 60; the default value is 20.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.0 and later
## SET NO_DIGITS_TIMEOUT 15
##
## INTER_DIGIT_TIMEOUT specifies the number of seconds that the telephone will wait
##  after a digit is dialed before sending a SIP INVITE.
##  Valid values are 1 through 10; the default value is 5.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
```

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```
##       96x0 SIP R2.0 and later
## SET INTER_DIGIT_TIMEOUT 6
##
## FAILED_SESSION_REMOVAL_TIMER specifies the number of seconds the telephone will
##  display a session line appearance and generate re-order tone after an invalid
##  extension has been dialed if the user does not press the End Call softkey.
##  Valid values are 5 through 999; the default value is 30.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET FAILED_SESSION_REMOVAL_TIMER 15
##
## TCP_KEEP_ALIVE_STATUS specifies whether or not the telephone sends TCP keep alive messages.
##  Value  Operation
##    0    Keep-alive messages are not sent
##    1    Keep-alive messages are sent (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET TCP_KEEP_ALIVE_STATUS 0
##
## TCP_KEEP_ALIVE_TIME specifies the number of seconds that the telephone will wait
##  before sending out a TCP keep-alive (TCP ACK) message.
##  Valid values are 10 through 3600; the default value is 60.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET TCP_KEEP_ALIVE_TIME 45
##
## TCP_KEEP_ALIVE_INTERVAL specifies the number of seconds that the telephone will wait
##  before re-transmitting a TCP keep-alive (TCP ACK) message.
##  Valid values are 5 through 60; the default value is 10.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET TCP_KEEP_ALIVE_INTERVAL 15
##
## CONTROLLER_SEARCH_INTERVAL specifies the number of seconds the telephone will wait
##  to complete the maintenance check for monitored controllers.
##  Valid values are 4 through 3600.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later (default value is 16)
```

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```
##       96x1 SIP R6.0 and later (default value is 16)
##       H1xx SIP R1.0 and later (default value is 16)
##       96x0 SIP R2.6.5 and later (default value is 16)
##       96x0 SIP R2.4.1 - R2.6.4 (default value is 4)
## SET CONTROLLER_SEARCH_INTERVAL 20
##
## ASTCONFIRMATION specifies the number of seconds that the telephone will wait 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 and 3PCC environments as there is no
subscription to avaya-cm-feature-status.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later; the default value is 32.
##       96x1 SIP R6.0 and later; the default value is 32.
##       H1xx SIP R1.0 and later; the default value is 32.
##       96x0 SIP R2.6 and later; the default value is 60.
## SET ASTCONFIRMATION 90
##
## FAST_RESPONSE_TIMEOUT specifies the number of seconds that the telephone will wait
##  before terminating an INVITE transaction if no response is received.
##  However, a value of 0 means that this timer is disabled.
##  Valid values are 0 through 32; the default value is 4.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later - it is provided by SMGR for phones connected to Avaya Aura however the
settings file
##                                configuration is still applicable for non-Avaya Aura systems.
##       96x1 SIP 6.0 and later. In 96x1 SIP R6.2 it is provided by SMGR for phones connected to
Avaya Aura however the settings file
##                                configuration is still applicable for non-Avaya Aura systems.
##       96x0 SIP R2.4.1 and later
## SET FAST_RESPONSE_TIMEOUT 5
##
## RDS_INITIAL_RETRY_TIME specifies the number of seconds that the telephone will wait
##  the first time before trying to contact the PPM server again after a failed attempt.
##  Each subsequent retry will be delayed by double the previous delay.
##  Valid values are 2 through 60, the default value is 2.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.4.1 and later
## SET RDS_INITIAL_RETRY_TIME 4
##
## RDS_MAX_RETRY_TIME specifies the maximum delay interval in seconds after which
##  the telephone will abandon its attempt to contact the PPM server.
##  Valid values are 2 through 3600, the default value is 600.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.4.1 and later
```

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```
## SET RDS_MAX_RETRY_TIME 600
##
## RDS_INITIAL_RETRY_ATTEMPTS specifies the number of retries after which
##  the telephone will abandon its attempt to contact the PPM server.
##  Valid values are 1 through 30, the default value is 15.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.4.1 and later
## SET RDS_INITIAL_RETRY_ATTEMPTS 20
##
## SIP Timer T1 is an estimate of the Round Trip Time (RTT) and is defined in milliseconds.
##  Valid values are 500 through 10000 milliseconds; the default value is 500.
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET SIP_TIMER_T1 2000
##
## SIP Timer T2 is maximum retransmit interval for non-INVITE requests and INVITE responses and
is defined in milliseconds.
##  Valid values are 2000 through 40000 milliseconds; the default value is 4000.
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET SIP_TIMER_T2 5000
##
## SIP Timer T4 is maximum duration a message will remain in the network and is defined in
milliseconds.
##  Valid values are 2500 through 60000 milliseconds; the default value is 5000.
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET SIP_TIMER_T4 6000
##
## FORBIDDEN_SESSION_REMOVAL_TIMER specifies the duration of an off-hook
##  session before call is automatically ended in case no more call appearances
##  is available on the called/remote party.
##  Value: 5 - 20 seconds; Default 10 seconds
##  This parameter is supported by:
##      J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      96x1 SIP R7.1.0.0 and later
## SET FORBIDDEN_SESSION_REMOVAL_TIMER 5
##


#############  CONFERENCING SETTINGS (SIP ONLY)  #############
##
## CONFERENCE_FACTORY_URI specifies the URI for Avaya Aura Conferencing or Network Conferencing
in 3PCC environments.
##  Valid values contain zero or one URI,
##   where a URI consists of a dial string followed by "@" followed by a domain,
##   which must match the routing pattern configured in System Manager for Adhoc Conferencing.
##  Depending on the dial plan, the dial string may 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 ("").
##  This parameter is supported by:
```

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```
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later (for IPO environment only).
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later (for IPO environment only).
##       96x1 SIP R6.2.1 and later
##       H1xx SIP R1.0 and later
## SET CONFERENCE_FACTORY_URI 93375000@avaya.com

################  PRESENCE SETTINGS (SIP ONLY)  ###############
##
## ENABLE_PRESENCE specifies whether presence will be supported.
##  Value  Operation
##    0    Disabled
##    1    Enabled
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later (default is 1); For IP office and 3PCC environments this
parameter shall be set to 0 as presence is not supported.
##       J169/J179 SIP R1.5.0 (default is 1)
##       96x1 SIP R6.2 and later     (default is 1)
##       96x0 SIP R2.6.8 and later (default is 1)
##       96x0 SIP R2.6.6 and R2.6.7  (default is 0)
##       H1xx SIP R1.0 and later (default is 1); For IP office environment this parameter shall
be set to 0 as presence is not supported.
## SET ENABLE_PRESENCE 1

## PRESENCE_ACL_CONFIRM 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
##    2    Prompt - the phone directly prompting the user to Allow or Deny the watcher's request.
##  This parameter is not supported in IP Office environment as presence is not supported.
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later (values 0-1)
##       96x1 SIP R6.3 and later (values 0-1)
##       H1xx SIP R1.0 and later (values 0-2)
## SET PRESENCE_ACL_CONFIRM 1
##
## ALLOW_DND_SAC_LINK_CHANGE determines if the user will be allowed to change the DND and SAC
button link.
## If the change is allowed, the menu to set the DND and SAC link is displayed.
##  Value  Operation
##  0 - do not allow a user to change default behavior (Default)
##  1 - allow a user change default behavior; parameter will be included in the "A" menu
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.4 and later.
## SET ALLOW_DND_SAC_LINK_CHANGE 1
##
## DND_SAC_LINK specifies whether to activate the SendAllCall when user enables DoNotDisturb.
##  Value  Operation
##    0    Do not activate the SendAllCall when user enables DoNotDisturb (default)
##    1    Activate the SendAllCall when user enables DoNotDisturb
```

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```
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later; The value of this parameter is used if the
##              ALLOW_DND_SAC_LINK_CHANGE is set to 0.
##       96x1 SIP R6.4 and later; The value of this parameter is used if the
ALLOW_DND_SAC_LINK_CHANGE is set to 0.
##       Avaya Equinox 3.1.2 and later
## SET DND_SAC_LINK 1


## AWAY_TIMER_VALUE controls the amount of time in minutes where there was no interaction
##  with the device after which the device assumes that the user is away from the device.
##  The range is 1-1500 minutes. The default value is 30 minutes.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.4 and later.
##      H1xx SIP R1.0 and later; this parameter is not supported in IP Office environment as
presence is not supported.
## SET AWAY_TIMER_VALUE 10
##
## AWAY_TIMER controls whether the device report an 'away' state.
##  When this parameter is set to 1, the device will automatically report an 'away' state.
##  Value  Operation
##    0    Disabled
##    1    Enabled (Default)
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.4 and later.
##      H1xx SIP R1.0 and later; this parameter is not supported in IP Office environment as
presence is not supported.
## SET AWAY_TIMER 0

###########  MLPP SETTINGS (SIP ONLY)  ##########
##
## ENABLE_MLPP specifies whether MLPP feature is enabled or not.
## Value   Operation
##  0       Disable MLPP feature (default)
##  1       Enable MLPP feature
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET ENABLE_MLPP   1
##
## MLPP_NET_DOMAIN specifies MLPP Network Domain
## Value      Operation
## ""         No MLPP Network Domain is configured (default)
## "dsn"      DSN Network
## "uc"       UC Network
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
```

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```
## SET  MLPP_NET_DOMAIN  "dsn"
##
## MLPP_MAX_PREC_LEVEL specifies maximum allowed precedence level for the user
## Value   Operation
##   1         Maximum allowed precedence level is Routine (default)
##   2         Maximum allowed precedence level is Priority
##   3         Maximum allowed precedence level is Immediate
##   4         Maximum allowed precedence level is Flash
##   5         Maximum allowed precedence level is Flash Override
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET MLPP_MAX_PREC_LEVEL 2
##
## ENABLE_PRECEDENCE_SOFTKEY indicates whether precedence soft key should be enabled on idle line
appearances on Phone Screen.
## Value   Operation
##   0         Disable precedence soft key
##   1         Enable precedence soft key (default)
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET ENABLE_PRECEDENCE_SOFTKEY 0
##
## DSCPAUD_FO specifies the DSCP value for Flash Override precedence/priority level voice call
(0-63). Default value is 41.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET DSCPAUD_FO 42
##
## DSCPAUD_FL specifies the DSCP value for Flash precedence/priority level voice call (0-63).
Default value is 43.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET DSCPAUD_FL 44
##
## DSCPAUD_IM specifies the DSCP value for Immediate precedence/priority level voice call (0-63).
Default value is 45.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET DSCPAUD_IM 43
##
## DSCPAUD_PR specifies the DSCP value for Priority precedence/priority level voice call (0-63).
Default value is 47.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
```

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```
## SET DSCPAUD_PR 48
##
## DSCPMGMT specifies the DSCP value for OA&M management packet (0-63). The default value is 16.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET DSCPMGMT 15
##
###############  EXCHANGE SETTINGS (SIP ONLY)  ###############
##
## EXCHANGE_SERVER_LIST 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; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
##       H1xx SIP R1.0 and later
## SET EXCHANGE_SERVER_LIST exch1.myco.com,exch2.myco.com,exch3.myco.com
##

## PROVIDE_EXCHANGE_CONTACTS specifies whether menu item(s) for Exchange Contacts are displayed.
##  Value  Operation
##    0    Not displayed
##    1    Displayed (default)
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0;  J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.2 and later
##       96x0 SIP R2.0 through R2.4 only
## SET PROVIDE_EXCHANGE_CONTACTS 0
##
## PROVIDE_EXCHANGE_CALENDAR specifies whether menu item(s) for Exchange Calendar are displayed.
##  Value  Operation
##    0    Not displayed
##    1    Displayed (default)
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
## SET PROVIDE_EXCHANGE_CALENDAR 0
##

##
## EXCHANGE_USER_DOMAIN specifies the domain for the URL
##  used to obtain Exchange contacts and calendar data. The EXCHANGE_USER_DOMAIN is used as part
of the
##  user authentication.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
```

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```
##       J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0 and later; Users can
change this value in the "Options & Settings...". Refer to EXCHANGE_AUTH_USERNAME_FORMAT for how
EXCHANGE_USER_DOMAIN is used.
##       J169/J179 SIP R1.5.0; Users can change this value in the "Options & Settings...". Refer
to EXCHANGE_AUTH_USERNAME_FORMAT for how EXCHANGE_USER_DOMAIN is used.
##       96x1 SIP R6.0 and later; Users can change this value in the "Options & Settings...".
Refer to EXCHANGE_AUTH_USERNAME_FORMAT for how EXCHANGE_USER_DOMAIN is used.
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.5 and later
## SET EXCHANGE_USER_DOMAIN exchange.myco.com
##
## EXCHANGE_AUTH_USERNAME_FORMAT specifies the format of the username for user authentication.
##  Value  Operation
##    0    Office 2003/Office2016 username format - "EXCHANGE_USER_DOMAIN\Exchange Username" or
"Exchange Username" if EXCHANGE_USER_DOMAIN is "".
##         This is the default value.
##    1    Office 365 username format - "Exchange Username@EXCHANGE_USER_DOMAIN" or "Exchange
Username" if EXCHANGE_USER_DOMAIN is "".
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later(J169/J179 only), J159 SIP R4.0.3.0 and
later
##       96x1 SIP R7.1.0.0 and later.
## SET EXCHANGE_AUTH_USERNAME_FORMAT 1
##
## EXCHANGE_EMAIL_DOMAIN specifies the Exchange email domain.
##  Exchange Username with EXCHANGE_EMAIL_DOMAIN defines the email address: Exchange
Username@EXCHANGE_EMAIL_DOMAIN.
##  This parameter cannot be changed by end users in the "Options & Settings..." menu.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.3 and later
## SET EXCHANGE_EMAIL_DOMAIN avaya.com
##
## ENABLE_EXCHANGE_REMINDER specifies whether or not Exchange reminders will be displayed.
##  Value  Operation
##    0    Not displayed (default)
##    1    Displayed
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
## SET ENABLE_EXCHANGE_REMINDER 1
##
## EXCHANGE_REMINDER_TIME specifies the number of minutes before an appointment
##  at which a reminder will be displayed.
##  Valid values are 0 through 60; the default value is 5.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
## SET EXCHANGE_REMINDER_TIME 7
##
```

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```
## EXCHANGE_SNOOZE_TIME specifies the number of minutes after a reminder has been
##  temporarily dismissed at which the reminder will be redisplayed.
##  Valid values are 0 through 60; the default value is 5.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
## SET EXCHANGE_SNOOZE_TIME 4
##
## EXCHANGE_REMINDER_TONE 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 (default)
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
## SET EXCHANGE_REMINDER_TONE 0
##
## EXCHANGE_NOTIFY_SUBSCRIPTION_PERIOD specifies the number of seconds between re-syncs
##  with the Exchange server.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later; valid values are 60 through 3600; the default value is 180.
##       96x1 SIP R6.2 and later; valid values are 60 through 3600; the default value is 180.
##       96x1 SIP R6.0.x;         valid values are  0 through 3600; the default value is 180.
##       96x0 SIP R2.5 and later; valid values are  0 through 3600; the default value is 180.
## SET EXCHANGE_NOTIFY_SUBSCRIPTION_PERIOD 200
##

###################  OTHER SIP-ONLY SETTINGS  #################
##
## SPEAKERSTAT 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 (default)
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later; the parameter is not supported by J129.
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
## SET SPEAKERSTAT 1
##
## MUTE_ON_REMOTE_OFF_HOOK controls the speakerphone muting for a remote-initiated
## (a shared control or OOD-REFER)  speakerphone off-hook.
##
## Valid values are 0 and 1
##       0 - the speakerphone is Unmuted
##       1 - the speakerphone is Muted
##
##  The default value is  1 (Muted) for 96x1 SIP R6.3
```

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```
##  The default value is  0 (Unmuted) for 96x1 SIP R6.3.1 and later, J129 SIP R1.0.0.0 and later
and H1xx SIP R1.0 and later
##
##    This parameter is supported by:
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      96x1 SIP R6.3 and later
##      H1xx SIP R1.0 and later; R1.0.2 and later - this parameter is also supported in IPO
environment where it is used
##      to control auto answer calls whether they start muted (1) or not (0).
##
##  The value of the parameter MUTE_ON_REMOTE_OFF_HOOK will be applied to the phone only when the
phone is
##  deployed with a CM 6.2.2 and earlier releases.
##
##  If the phone is deployed with CM 6.3 or later, the MUTE_ON_REMOTE_OFF_HOOK variable is
ignored and instead
##  the feature is delivered via PPM by enabling the Turn on mute for remote off-hook attempt
parameter in the station form
##  via the Session Manager (System Manager) or Communication Manager (SAT) administrative
interfaces.
##
## SET MUTE_ON_REMOTE_OFF_HOOK 0

## SDPCAPNEG specifies whether or not SDP capability negotiation is enabled.
##  Value  Operation
##    0    SDP capability negotiation is disabled
##    1    SDP capability negotiation is enabled (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.6 and later
## SET SDPCAPNEG 0
##
## ENFORCE_SIPS_URI specifies whether a SIPS URI must be used for SRTP.
##  Value  Operation
##    0    Not enforced
##    1    Enforced (default)
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later;
##          For 3PCC environments using tls signaling, ENFORCE_SIPS_URI must be set to 0.
##       Avaya Equinox 3.4 and later
##       J169/J179 SIP R1.5.0
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.6 and later
## SET ENFORCE_SIPS_URI 1
##
## 100REL_SUPPORT specifies whether the 100rel option tag is included in the SIP INVITE header
field.
##  Value  Operation
##    0    The tag will not be included.
##    1    The tag will be included (default).
```

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```
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0  SIP  R2.6 and later
## SET 100REL_SUPPORT 1

## PLAY_TONE_UNTIL_RTP specifies whether locally-generated ringback tone will stop
##  as soon as SDP is received for an early media session, or whether it will continue
##  until RTP is actually received from the far-end party.
##  Value  Operation
##    0    Stop ringback tone as soon as SDP is received
##    1    Continue ringback tone until RTP is received (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.2 and later
##       H1xx  SIP  R1.0 and later
## SET PLAY_TONE_UNTIL_RTP 0

## PHONEKEY specifies list of pre-config keys. This parameter is used for mapping feature,
application, line and autodial keys available in the Phone screen.
##    The format of each PHONEKEY parameter shall be as follows:
##    SET PHONEKEY
"Key=[n1];Type=[Feature|Application|Line|Autodial];Name=[name];attr1=[value];attr2=[value];Label=
[label (optional)][;Forced (optional)]
##      [n1] corresponds to the number of the phone key to be configured. The allowed values are
positive integers from 1 to 96.
##      [Feature|Application|Line|Autodial] corresponds to the functionality to be assigned to a
key. The allowed values are: feature, application , line or autodial. Please note that
##         in the Avaya Aura® environment, it is recommended to configure the line keys on Avaya
Aura® System Manager.
##      [name], depending on the functionality entered in Type, can be either of the following:
##         - the name of the feature that will be accessed by pressing the customized phone key,
e.g. callpickup,groupcallpickup, etc. blf is supported in OpenSIP environment.
##             PHONEKEY can be configured with blf and particular phone extension as attr1 allows
BLF configuration in OpenSIP when BLF_LIST_URI is not specified.
##         - the name of the application, e.g., lock, logout, etc.
##         - the type of the phone line that will be accessed. The allowed values are: primary,
bca (Bridged Call Appearance), sca (Shared Call Appearance, supported in OpenSIP environment
only.).
##         - autodialing of a defined phone extension - e.g. autodial. PHONEKEY is the only way
to configure autodial in Open SIP environment.
##      [attr1] and [attr2] are optional or mandatory attributes per the [Type] and [Name]
values. For example, for bca attr1 is the bridged line number and attr2 is the extension.
##      [label] is the key label that will be displayed on the Phone screen.
##      [Forced] determines whether the user can move or delete a key. A few scenarios:
##         - If Forced is set and location is empty, the key is applied.
##         - If Forced is set for an occupied key location, it overrides the existing key or
moves it to a different location.
##         - If Forced is not set and the location is empty, the new definition for this location
is applied.
##         - If Forced is not set and the location is already occupied, the new definition is
dropped.
##         - If Forced is not set and the location is empty but this key type is already
configured for a different location, the new definition is dropped.
```

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```
##    Multiple PHONEKEY lines can be specified, one for each key. All the PHONEKEY values and
keywords are case-insensitive except values set in Label.
##    The parameter is supported in OpenSIP and Aura environments.
##    For full list of allowed Feature, Application, Line and Autodial values please refer to the
phone's administration and installation guide.
##    The default is "".
##  This parameter is supported by:
##       J100 SIP R4.0.1.0 and later (J169/J179 only), J159 SIP R4.0.3.0 and later
## SET PHONEKEY "Key=1;Type=application;Name=lock"
## SET PHONEKEY "Key=1;Type=autodial;Name=autodial;attr1=6837"
## SET PHONEKEY "Key=1;Type=line;Name=bca;attr1=1;attr2=6837"
## SET PHONEKEY "Key=1;Type=feature;Name=sac;attr1=6837"
## SET PHONEKEY "Key=2;Type=feature;Name=team;attr1=6836"
## SET PHONEKEY "key=6;Type=Feature;Name=BLF;Attr1=4004004;Label=Kichu4004"
##


## TEAM_BUTTON_RING_TYPE specifies the alerting pattern to use for team buttons.
##  Valid values are 1 through 8, the default value is 1.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.2 and later
## SET TEAM_BUTTON_RING_TYPE 3
##


## LOCALLY_ENFORCE_PRIVACY_HEADER specifies whether the telephone will display
##  "Restricted" (in the current language) instead of CallerId information when
##  a Privacy header is received in a SIP INVITE message for an incoming call.
##  Value  Operation
##    0    Disabled (default): CallerID information will be displayed
##    1    Enabled: "Restricted" will be displayed
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.2 and later
##       H1xx SIP R1.0 and later
## SET LOCALLY_ENFORCE_PRIVACY_HEADER 1
##
## ENABLE_SIP_USER_ID controls the display of the user ID input field on the Login Screen
##  Value  Operation
##    0    SIP User ID field is not available to user during Login (default)
##    1    SIP User ID field is available to user during Login
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET ENABLE_SIP_USER_ID 1
##
## ENABLE_STRICT_USER_VALIDATION specifies whether AOR received in 'Request-URI' of incoming call
should be validated or not with 'contact' header published by phone in REGISTRATION.
##  Value  Operation
##   0     validation is not done (Default)
##   1     validation is done
##  Note: This parameter is supported by J129 SIP R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later and J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET ENABLE_STRICT_USER_VALIDATION 1
##
```

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```
## BRANDING_VOLUME 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 (default)
##    4     3db below nominal
##    3     6db below nominal
##    2     9db below nominal
##    1    12db below nominal
##    0     No Volume
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
(Values 0-8)
##       J169/J179 SIP R1.5.0 (Values 1-8)
##       Avaya Vantage Devices SIP R1.0.0.0 and later (Values 1-8)
##       J129 SIP R1.0.0.0 (or R1.1.0.0) (Values 1-8)
##       96x1 SIP  R6.2 and later (Values 1-8)
##       H1xx SIP R1.0 and later (Values 1-8)
## SET BRANDING_VOLUME 2
##
## ENABLE_OOD_MSG_TLS_ONLY specifies whether 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 (default)
##  Note: A value of 0 is only intended for testing purposes.
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.2 and later
##       H1xx  SIP  R1.0 and later
## SET ENABLE_OOD_MSG_TLS_ONLY 1
##


## TEAM_BUTTON_REDIRECT_INDICATION controls if the redirection indication should be shown on
## a Team Button (on a monitoring station) in case it is not a redirect destination of the
monitored station.
##  Value  Operation
##  0 - disabled; the redirect indication will be shown only on a monitoring station which is
redirection destination (default).
##  1 - enabled; the redirection icon is displayed on all monitoring stations
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.4 and later
## SET TEAM_BUTTON_REDIRECT_INDICATION 1
##
## ENABLE_BLIND_TRANSFER indicates whether enable blind transfer or not
## Value   Operation
##   0     Disable blind transfer
##   1     Enable blind transfer (default)
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
```

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```
##       96x1 SIP R7.1.0.0 and later
## SET ENABLE_BLIND_TRANSFER 0
##
## USE_CONTACT_IN_REFERTO defines which transfer target address should be used in Refer-
To (header of REFER SIP request on attended transfer).
## Value   Operation
##   0     use TO URI of the transfer target in Refer-To header of REFER SIP request
##   1     use CONTACT URI of the transfer target in Refer-To header of REFER SIP request
(default)
##  This parameter is supported by:
##       J100 SIP R4.0.1.0 and later, J159 SIP R4.0.3.0 and later
## SET USE_CONTACT_IN_REFERTO 0
##

############  ACCESSIBILITY SETTINGS (SIP ONLY)  ##############
##
## PROVIDE_KEY_REPEAT_DELAY specifies how long a navigation button must be held down
##  before it begins to auto-repeat, and whether an option will be provided by which
##  the user can change this value.
##  Value  Operation
##    0    Default    (500ms) with user option (default)
##    1    Short      (250ms) with user option
##    2    Long      (1000ms) with user option
##    3    Very Long (2000ms) with user option
##    4    No Repeat          with user option
##    5    Default    (500ms) without user option
##    6    Short      (250ms) without user option
##    7    Long      (1000ms) without user option
##    8    Very Long (2000ms) without user option
##    9    No Repeat          without user option
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.2 and later
## SET PROVIDE_KEY_REPEAT_DELAY 2

######################  HEADSET PROFILES  ####################
##
## HEADSET_PROFILE_DEFAULT specifies the number of the default headset audio profile.
##  Valid values are 1 through 20; the default value is 1.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x1 SIP R6.3 and later.
##       H1xx SIP R1.0 and later
## SET HEADSET_PROFILE_DEFAULT 1
##
## HEADSET_PROFILE_NAMES specifies an ordered list of names to be displayed for headset audio
profile selection.
##  See support.avaya.com for the list of headset profiles.
##  The list can contain 0 to 255 UTF-8 characters; the default value is null ("").
##  Names are separated by commas without any intervening spaces.
##  Two commas in succession indicate a null name,
##   which means that the default name should be displayed for the corresponding profile.
```

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```
##  Names may contain spaces, but if any do, the entire list must be quoted.
##  There is no way to prevent a profile from being displayed.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x1 SIP R6.3 and later.
##       H1xx SIP R1.0 and later
## SET HEADSET_PROFILE_NAMES "Avaya L100 Headset,Vendor A,Vendor B,Vendor C,,,,"
##
###################### LONG TERM ACOUSTIC EXPOSURE PROTECTION SETTINGS  ####################
##
## ACOUSTIC_EXPOSURE_PROTECT_MODE_DEFAULT specifies the Long-term acoustic exposure protection
mode default setting.
##  Value  Operation
##    1       Off (default)
##    2       Dynamic
##    3       4 hours
##    4       8 hours
##  This parameter is supported by:
##       J169/J179 H.323 R6.8.2 and later
##       96x1 H.323 R6.8.2 and later
##       J100 SIP R4.0.1.0 and later (J139, J169, J179 only), J159 SIP R4.0.3.0 and later.
## SET ACOUSTIC_EXPOSURE_PROTECT_MODE_DEFAULT 2
##
######################  HANDSET PROFILES  ####################
##
## HANDSET_PROFILE_DEFAULT specifies the number of the default handset audio profile.
##  Valid values are 1 through 20; the default value is 1.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
##       Avaya Vantage Devices SIP R1.0.0.0 and later
## SET HANDSET_PROFILE_DEFAULT 1
##
## HANDSET_PROFILE_NAMES specifies an ordered list of names to be displayed for handset audio
profile selection.
##  The list can contain 0 to 255 UTF-8 characters; the default value is null ("").
##  Names are separated by commas without any intervening spaces.
##  Two commas in succession indicate a null name,
##   which means that the default name should be displayed for the corresponding profile.
##  Names may contain spaces, but if any do, the entire list must be quoted.
##  There is no way to prevent a profile from being displayed.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
##       Avaya Vantage Devices SIP R1.0.0.0 and later
## SET HANDSET_PROFILE_NAMES "Acme Earwigs,,Spinco Ear Horns"
##
################  EMERGENCY TELEPHONE NUMBER  ################
##
## PHNEMERGNUM specifies an emergency telephone number to be dialed if the associated button is
selected.
##  Valid values may contain up to 30 dialable characters (0-9, *, #); the default value is null
("").
```

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```
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later for OpenSIP environment only.
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      Avaya Vantage Devices SIP R1.1.0.1 and later for IP Office (R1.1.0.1+) and OpenSIP
(R2.0.1.0+) environments only (The emergency information is retrieved in Aura environment from
PPM).
##       96x1 H.323 R6.0 and later; the parameter is supported when the phone is registered to
Avaya Communication Manager only.
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.5 and later; the parameter is supported when the phone is registered to
Avaya Communication Manager only.
##       96x0  SIP  R2.0 and later
##       4630 H.323 R1.0 and later
## SET PHNEMERGNUM 9911
##
## PHNMOREEMERGNUMS specifies list of comma separated emergency numbers
##  Valid values may contain up to 30 dialable characters (0-9, *, #); the default value is null
("").
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later for OpenSIP environment only.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.1.0.1 and later for IP Office (R1.1.0.1+) and OpenSIP
(R2.0.1.0+) environments.
##       H1xx  SIP  R1.0.2 and later
## SET PHNMOREEMERGNUMS "911,109,115"
##
############  EMERGENCY NUMBER SOFTKEY (SIP ONLY)  ############
##
## ENABLE_SHOW_EMERG_SK specifies whether an emergency softkey,
##  with or without a confirmation screen, will be displayed when the phone is registered.
##  All emergency numbers will always be supported.
##  Value  Operation
##    0    An emergency softkey will not be displayed.
##    1    An emergency softkey will be displayed, without a confirmation screen.
##    2    An emergency softkey will be displayed, with a confirmation screen (default).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.2 and later
## SET ENABLE_SHOW_EMERG_SK 1
##
## ENABLE_SHOW_EMERG_SK_UNREG specifies whether an emergency softkey,
##  with or without a confirmation screen, will be displayed when the phone is not registered.
##  All emergency numbers will always be supported.
##  Value  Operation
##    0    An emergency softkey will not be displayed.
##    1    An emergency softkey will be displayed, without a confirmation screen.
##    2    An emergency softkey will be displayed, with a confirmation screen (default).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later;
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
```

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```
##       96x1  SIP  R6.2 and later
## SET ENABLE_SHOW_EMERG_SK_UNREG 1
##


####################  CALL LOG SETTINGS  ####################
##

##
## LOG_DIALED_DIGITS specifies if the call log will contain digits dialed by a user or
##  information about a remote party in case where the user dialed 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).
##
##   This parameter is supported by:
##            J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later,
J159 SIP R4.0.3.0 and later
##            96x1 SIP R6.5 and later
##
##  SET LOG_DIALED_DIGITS 0

##
############################################################
##
##                     WI-FI SETTINGS
##
############################################################
##
##########   NETWORK MODE OF OPERATION   ##########
##
## WIFISTAT specifies whether the user is given an option to enable Wi-Fi.
##  Value  Operation
##    0    Wi-Fi is disabled and the user is not given an option to enable it; the phone will
only use the Ethernet interface.
##    1    The user is given an option to enable Wi-Fi; the phone will connect to Ethernet
(Default), unless the UI is used to manually switch to Wi-Fi.
##    2    Wi-Fi is the preferred interface, but manual override to a different SSID or to
Ethernet is allowed; the phone will connect to WLAN_ESSID (i.e., the pre-configured Wi-Fi
##          network) unless the phone UI is used to manually switch to another SSID or to
Ethernet. Associated pre-configured Wi-Fi network security parameters must also be specified.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later; values 0-2 are supported.
##         Only J129 and J179 support a pluggable Wi-Fi/BT module (J139/J169 does not support Wi-
Fi/BT). If the phone does not support Wi-Fi/BT, WIFISTAT will be internally set
##         to 0, regardless of the value received from 46xxsettings.txt.
##         The Administrator should always test a new settings file configuration on a single
phone before committing it to several phones, as a configuration error (such as
##         specifying an incorrect WLAN_ESSID or Wi-Fi security settings) will cause phones to
become disconnected from the network, necessitating manual correction on each
##         phone's local User Interface, as the phone will not be reachable via any other means.
##         If the phone is using the pre-configured network (i.e., Ethernet or a specific Wi-Fi
WLAN_ESSID), and then the phone UI is used to manually switch to a different
```

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```
##         network, the phone will enter Manual Network Configuration Mode, which will cause the
phone to continue to connect to the manually-configured network on subsequent
##         reboots, regardless of the pre-configured network specified by WIFISTAT and any
associated parameters.
##         There are 2 ways to return the phone to Automatic Network Configuration mode (i.e., to
comply again with WIFISTAT, and if WIFISTAT=2, also WLAN_ESSID and
##         associated pre-configured Wi-Fi network security parameters):
##           - Use the phone's UI to explicitly toggle "Network config" from "Manual" to "Auto".
##           - Change WIFISTAT to 0 and reboot the phone, which will force the phone to use
Ethernet, after which, WIFISTAT can be changed to the desired value and the phone
##             rebooted again.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; values 0-1 are supported.
##       H1xx  SIP  R1.0 and later; values 0-1 are supported.
## SET WIFISTAT 0
##


##
## WLAN_MAX_AUTH_FAIL_RETRIES specifies how many times the phone will retry a secure connection
upon receiving (possibly successive) auth failures.
## Value range is 0-4. The default is 3.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET WLAN_MAX_AUTH_FAIL_RETRIES 2
##



  ##########   NETWORK CONFIGURATION USER PRIVILEGE   ##########
##
## ENABLE_NETWORK_CONFIG_BY_USER specifies whether network configuration can be modified by the
end user, either via the Settings menu, or when there is a network issue that
## could be remedied by the user.
##  Value  Operation
##    0    Disabled
##    1    Enabled (Default)
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET ENABLE_NETWORK_CONFIG_BY_USER 0
##
##########   WI-FI REGULATORY DOMAIN SETTINGS   ##########
##
## WLAN_COUNTRY specifies the 2-character ISO 3166 Alpha-2 Country code representing the Wi-Fi
regulatory domain.
##  The default value is "US".
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_COUNTRY CA
##
## WLAN_ENABLE_80211D specifies whether 802.11d is used or not.  When enabled, the Wi-Fi
regulatory domain will be used according to the 802.11d Country IE provided by the
## connected Wi-Fi Access Point.  When disabled, the Wi-Fi regulatory domain will be used
according to WLAN_COUNTRY.
##  Note: The use of 802.11d is banned in the United States, so this parameter must NOT be set to
1 in this regulatory domain.
##  Value  Operation
```

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```
##    0    Disabled (Default)
##    1    Enabled
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_ENABLE_80211D 1
##
##########   PRE-CONFIGURED WI-FI NETWORK   ##########
##
## WLAN_ESSID specifies the SSID string of the pre-configured Wi-Fi network.
## The value can contain 1 to 32 characters; the default value is null ("").
##  Valid characters are:
##    A-Z, a-z, 0-9, and the following:  *.-!$%&'()+,:;/\=@~#
##    The space character, ASCII 0x20, is NOT supported.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_ESSID mywlanSSID
##
## WLAN_SECURITY specifies the pre-configured Wi-Fi network Security Method.
##  Value     Operation
##   none     No security (Default)
##   wep      WEP security
##   wpa2psk  WPA/WPA2 PSK (pre-shared key) security
##   wpa2e    WPA2 Enterprise security (802.1x authentication)
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_SECURITY wpa2e
##
#####   Pre-configured Wi-Fi network WEP security settings   #####
##
## WEP_DEFAULT_KEY specifies the pre-configured Wi-Fi network index of the WEP default key.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wep.
##  The range of valid values is 1-4; the default value is 1.
##  Only Shared Key authentication is supported.  Open authentication is NOT supported.
##  Some Wi-Fi Routers can only be configured with 1 WEP key, in which case ONLY WEP_KEY1 should
be set.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WEP_DEFAULT_KEY 2
##
## WEP_KEY_LEN specifies the pre-configured Wi-Fi network WEP key length.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wep.
##  Value    Operation
##   64bit   WEP keys of 64 bits
##   128bit  WEP keys of 128 bits (default)
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WEP_KEY_LEN 64bit
##
## WEP_KEY1/2/3/4 specifies the pre-configured Wi-Fi network WEP Keys 1 to 4.
##  These parameters are only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wep.
```

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```
##  The value can contain 10 (for 64-bit WEP) or 26 (for 128-bit WEP) ASCII-Hex digits; the
default value is null ("").
##  Valid characters are:
##    0-9, A-F
##  These parameters are supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WEP_KEY1 0123456789ABCDEF0123456789
## SET WEP_KEY2 123456789ABCDEF01234567890
## SET WEP_KEY3 23456789ABCDEF012345678901
## SET WEP_KEY4 3456789ABCDEF0123456789012
##
#####   Pre-configured Wi-Fi network WPA/WPA2 PSK or 802.1X EAP security settings   #####
##
## WLAN_PASSWORD specifies the pre-configured Wi-Fi network password.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and:
##    - WLAN_SECURITY is wpa2psk
##      or
##    - WLAN_SECURITY is wpa2e and WLAN_WPA2E_EAP_METHOD is PEAP and WLAN_WPA2E_EAP_PHASE2 is
MSCHAPV2
##  If WLAN_SECURITY is wpa2psk, the value can contain 8 to 63 characters.
##  If WLAN_SECURITY is wpa2e, the value can contain 1 to 32 characters.
##  The default value is null ("").
##  Valid characters are:
##    A-Z, a-z, 0-9, and the following:  *.-!$%&'()+,:;/\=@~#
##    The space character, ASCII 0x20, is NOT supported.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_PASSWORD Avaya123
##
## WLAN_WPA2E_EAP_METHOD specifies the pre-configured Wi-Fi network 802.1x EAP method.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wpa2e.
##  Value  Operation
##   PEAP  Connect using PEAP (Default)
##   TLS   Connect using TLS
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_WPA2E_EAP_METHOD TLS
##
## WLAN_WPA2E_EAP_PHASE2 is the pre-configured Wi-Fi network 802.1x phase 2 Method.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wpa2e and
WLAN_WPA2E_EAP_METHOD is PEAP.
##  Value      Operation
##   none      No phase 2 authentication (Default, but not currently supported)
##   MSCHAPV2  As of J100 2.0, MUST be set to this value for forward compatibility
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_WPA2E_EAP_PHASE2 MSCHAPV2
##
## WLAN_WPA2E_IDENTITY specifies the pre-configured Wi-Fi network 802.1x identity.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wpa2e and:
##    - WLAN_WPA2E_EAP_METHOD is PEAP and WLAN_WPA2E_EAP_PHASE2 is MSCHAPV2
```

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```
##      or
##    - WLAN_WPA2E_EAP_METHOD is TLS
##  The value can contain 1 to 32 characters; the default value is null ("").
##  Valid characters are:
##    A-Z, a-z, 0-9, and the following:  *.-!$%&'()+,:;/\=@~#
##    The space character, ASCII 0x20, is NOT supported.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_WPA2E_IDENTITY User123
##
## WLAN_WPA2E_ANONYMOUS_IDENTITY specifies the pre-configured Wi-Fi network 802.1x anonymous
identity.
##  This parameter is only applicable when WIFISTAT enables Wi-Fi and WLAN_SECURITY is wpa2e and
##  WLAN_WPA2E_EAP_METHOD is PEAP and WLAN_WPA2E_EAP_PHASE2 is MSCHAPV2.
##  The value can contain 1 to 32 characters; the default value is null ("").
##  Valid characters are:
##    A-Z, a-z, 0-9, and the following:  *.-!$%&'()+,:;/\=@~#
##    The space character, ASCII 0x20, is NOT supported.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_WPA2E_ANONYMOUS_IDENTITY foo@example
##
##########  WLAN LAYER 2 QOS SETTINGS  ########
##
## WLAN_L2QAUD specifies the layer 2 priority value for audio frames generated by the telephone
when Wi-Fi interface is used.
##  Valid values are 0 through 7; the default value is 6.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_L2QAUD 1
##
## WLAN_L2QSIG specifies the layer 2 priority value for signaling frames generated by the
telephone when Wi-Fi interface is used.
##  Valid values are 0 through 7; the default value is 3.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_L2QSIG 1
##
##########  WLAN LAYER 3 QOS SETTINGS  ########
##
## WLAN_DSCPAUD specifies the layer 3 Differentiated Services (DiffServ) Code Point for audio
frames generated by the telephone when Wi-Fi interface is used.
##  Valid values are 0 through 63; the default value is 46.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_DSCPAUD 1
##
## WLAN_DSCPSIG specifies the layer 3 Differentiated Services (DiffServ) Code Point for signaling
frames generated by the telephone when Wi-Fi interface is used.
##  Valid values are 0 through 63; the default value is 34.
##  This parameter is supported by:
```

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```
##       J100 SIP R2.0.0.0 and later; only J129 and J179 support a pluggable Wi-Fi/BT module
(J139/J169 does not support Wi-Fi/BT), J159 SIP R4.0.3.0 and later
## SET WLAN_DSCPSIG 1
##

###################  CALL CENTER SETTINGS  #################
##
## HEADSYS specifies whether the telephone will go on-hook if the headset is active
##  when a Disconnect message is received.
##  Value  Operation
##    0    The telephone will go on-hook if a Disconnect message is received when the headset is
active
##    1    Disconnect messages are ignored when the headset is active
##  Note: a value of 2 has the same effect as a value of 0, and
##        a value of 3 has the same effect as a value of 1.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later (the default value is 0 unless the value
##            of CALLCTRSTAT is set to 1, in which case the default value is 1)
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later (the default value is 0)
##       96x1 H.323 R6.2.1 and later (the default value is 0 unless the value
##            of CALLCTRSTAT is set to 1, in which case the default value is 1)
##       96x1 H.323 R6.1 and R6.2 ignore this parameter, and will ignore Disconnect messages
##            if the user is logged in as a call center agent. If the user is not logged in
##            as a call center agent, the telephone will go on-hook if a Disconnect message
##            is received when the headset is active.
##       96x1 H.323 releases prior to R6.1 (the default value is 1)
##       96x1 SIP  R6.4 and later (the default value is 0)
##       96x1 SIP  R6.0 and later up to R6.4 (not included) (the default value is 1)
##       96x0 H.323 R1.2 and later (the default value is 1)
##       96x0 SIP  R1.0 and later (the default value is 1)
##       16xx H.323 R1.3 and later (the default value is 1)
## SET HEADSYS 0
##
##########  CALL CENTER SETTINGS (96x1/J100 SIP ONLY)  ############
##
## SKILLSCREENTIME specifies the duration, in seconds, that the Skills screen will be displayed.
##  Valid values are 0 through 60; the default value is 5.
##  A value of 0 means that the Skills screen will not be removed automatically when the agent
logs in.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.2 and later
## SET SKILLSCREENTIME 5
##
## UUIDISPLAYTIME specifies the duration, in seconds, that the UUI Information screen will be
displayed.
##  Valid values are 5 through 60; the default value is 10.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.2 and later
## SET UUIDISPLAYTIME 10
##
```

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```
## ENTRYNAME specifies whether the Calling Party Name or the VDN/Skill Name will be used in
History entries.
##  Value  Operation
##    0    Calling Party Name will be used (default)
##    1    VDN/Skill Name will be used
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.2 and later
## SET ENTRYNAME 1

## CC_INFO_TIMER specifies the duration, in hours, of the subscription to the SIP CC-Info event
package.
##  Valid values are 1 through 24; the default value is 8.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.2 and later
## SET CC_INFO_TIMER 8
##

##########  RECORDING TONE SETTINGS   #######
##

##
## RECORDINGTONE_INTERVAL specifies the number of seconds between Call Recording Tones.
##  Valid values are 1 through 60; the default value is 15.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.0.0 and later
##       96x1 H.323 R6.2 and later
##       B189 H.323 R1.0 and later
## SET RECORDINGTONE_INTERVAL 10
##
## RECORDINGTONE_VOLUME 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
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.0.0 and later
##       96x1 H.323 R6.2 and later
```

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```
##       B189 H.323 R1.0 and later
## SET RECORDINGTONE_VOLUME 8
##

################## TRUSTED CERTIFICATES AND GENERAL CERTIFICATES SETTINGS ##################
##
## TRUSTCERTS specifies a list of names of files that contain copies of CA certificates
##  (in PEM format) that will be downloaded, saved in non-volatile memory,
##  and used by the telephone to authenticate received identity certificates.
##  The list can contain up to 255 characters.
##  Values are separated by commas without intervening spaces.
##  The default value is null ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; support of up to 100 PEM and DER format
root and intermediate trusted certificates.
##             The list can contain up to 1024 characters. Avaya Vantage Open application does
not use the downloaded trusted certificates.
##              However, when Avaya Vantage Open application is installed, this parameter is used
to download trusted certificates for
##              to be used Avaya Vantage device (for example, 802.1x EAP-TLS) or by other
applications (for example, Android Browser, etc.).
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R2.0 and later
##       96x0  SIP  R1.0 and later
##
## SET TRUSTCERTS av_prca_pem_2033.txt,av_sipca_pem_2027.txt,av_csca_pem_2032.txt
## Note: The above is list of Avaya trusted certificates. You shall only use
## the ones that are required for your setup.
## Note: 96x1 H.323 R6.6 and later supports also intermediate certificates download for cases
## where servers do not provided the full certificate chain up to the root CA. There is no
support
## for certificate signature validation up to intermediate certificate. Certificate signature
validation
## is always supported up to the root CA.
## Note: Avaya Vantage Connect Application and Avaya Equinox uses the Android trusted certificate
repository and the downloaded certificates.
## using TRUSTCERTS.
##
## MAX_TRUSTCERTS specifies the maximum number of trusted
## certificates, which are defined by TRUSTCERTS
## parameter, can be downloaded to the phone.
## Note: each trusted certificate file may contain more than one certificate. MAX_TRUSTCERTS
enforces the number of certificates.
## Valid value:  1 to 10, default: 6
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP 7.1.0.0 and later
## SET MAX_TRUSTCERTS 8
##
```

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```
## ENABLE_PUBLIC_CA_CERTS specifies whether the embedded public root CA certificates are used for
services other than Device Enrollment Service (DES)
##  and re-directed file server using DES. DES always used the embedded public root CA
certificates (even if ENABLE_PUBLIC_CA_CERTS is 0).
##  For the re-directed file server using DES, there is use of embedded public root certificates
if DES service did not provide private CA. If DES provides private CA, then the
##  embedded public root CA certificates are ignored (however if DES is re-triggered from admin
menu and private CA is provided from DES then the embedded public root CA certificates will be
used according to ENABLE_PUBLIC_CA_CERTS).
##  For rest of the services, this parameter controls whether embedded public root CA
certificates are used (in addition, to downloaded trusted certificates) or not (only downloaded
trusted certificates are used).
##  If DES did not provide private CA, then the ENABLE_PUBLIC_CA_CERTS is set to "1" without
ability to change it. If DES provides private CA, then this parameter is configurable (in such
case, TRUSTCERTS shall include
##  DES service private CA, else the phone will not be able to re-connect to the re-directed file
server).
##  For cases where DES is not used, then the parameter is fully configurable and if
ENABLE_PUBLIC_CA_CERTS is "0" and no downloaded trusted certificates (TRUSTCERTS=="") then the
phone trusts for any HTTP/S file server
##  for configuration / image download and fails with rest of services (PPM/SIP, AADS, etc). If
either ENABLE_PUBLIC_CA_CERTS is "1" and/or TRUSTCERTS<> "" then the service must have identity
certificate that can be validated
##  using the embedded public root CA certificates (if ENABLE_PUBLIC_CA_CERTS is "1") or
downloaded trusted certificates (if TRUSTCERTS <>"") - there is no exception to configuration and
software files download from the HTTP/S file server
##  in such case.
##  Value  Operation
##    0    Embedded public CA certs are not trusted (Default).
##    1    Embedded public CA certs are always trusted (in addition to trusted certificates
downloaded according to TRUSTCERTS)
##  This parameter is supported by:
##      J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      Avaya Vantage Devices SIP R1.1.0.0 and later; the embedded public root CA certificates
are Android public root certificates which can be viewed in the settings application --> Security
--> Trusted credentials --> SYSTEM.
## SET ENABLE_PUBLIC_CA_CERTS 1
## Note: This parameter is used on Avaya Vantage devices to enable all Android root CA
certificates for non-Android applications such as AADS, configuration and firmware download using
HTTPS, PPM, 802.1x EAP-TLS, SCEP over HTTPS.
##       This parameter cannot be used to disable Android root CA certificates for Android
applications. CA_CERT_BLACKLIST shall be used to disable Android root CA certificates for both
Android and non-Android applications.
##
## TLSSRVRID specifies whether a certificate will be trusted only if the
##  identity of the device from which it is received matches the certificate,
##  per Section 3.1 of RFC 2818.
##  Value  Operation
##    0    Identity matching is not performed
##    1    Identity matching is performed (default)
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; Not used by Avaya Vantage Open
application.
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later; Supported by SIP/PPM and file downloads.
##       B189 H.323 R1.0 and later
```

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```
##       96x0 H.323 R2.0 and later
##       96x0  SIP  R2.0 and later
##      TLSSRVRID is not supported by 96x1 H.323 phones and instead
##       TLSSRVRVERIFYID is supported (see below)
## SET TLSSRVRID 0
##


##
## FQDN_IP_MAP specifies a comma separated list of name/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. Default is
"". String length is up to 255
## characters. No spaces are allowed inside the string.
## The purpose of this parameter is to support cases where the server certificate Subject Common
Name of Subject Alternative Names
## include FQDN (instead of IP address) and the SIP_CONTROLLER_LIST is defined using IP address.
The main use case is for Avaya Aura SM/PPM connectivity
## where the SIP controller list returned from Aura (PPM) to the endpoint is IP address only
while server certificate is defined with FQDN.
## Internet trusted CAs prefer signing of Internet public server certificates with FQDN only.
## This parameter is supported with any phone service running over TLS. Though, the main use case
if for Avaya Aura SM/PPM services.
## This parameter is not to be used as an alternative to a DNS lookup or reverse DNS lookup.
## The reverse case will not be supported. If the phone is accessing a server using an FQDN and
the server's certificate only contains an IP address,
## this will be considered a failure and the FQDN_IP_MAP will not be used.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0) (IPv6 is not yet supported)
##       J169/J179 SIP R1.5.0
##       96x1  SIP R7.1.0.0 and later
## SET FQDN_IP_MAP
"sm1.avaya.com=135.20.230.199,sm1.avaya.com=2000::204,sm2.avaya.com=135.20.230.201,ppm.ottawa.ava
ya.com=2000::207"
##
## SERVER_CERT_RECHECK_HOURS 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.
## SERVER_CERT_RECHECK_HOURS is applicable for H.323 over TLS signaling only in 96x1 H.323 R6.6.
## SERVER_CERT_RECHECK_HOURS is applicable for SIP and 802.1x EAP-TLS when used by J129 SIP
R1.0.0.0 and later.
## Valid values are: 0-32767. A value of 0 means that periodic checks will not be done.
## The default is 24.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET SERVER_CERT_RECHECK_HOURS 30
##
## CERT_WARNING_DAYS specifies how many days before the expiration of a certificate that a
warning
## should first appear on the phone screen. This includes trusted certificates, OCSP certificates
and identity certificate.
## Log and syslog message will be generated as well. The warning will reappear every 7 days.
```

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```
## Valid values are: 0-99 (60 is default), where 0 means no certificate expiration warning will
be generated.
## This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x1 H.323 R6.6 and later
## SET CERT_WARNING_DAYS 30
##
## DELETE_MY_CERT specifies whether the installed identity certificate (using SCEP or PKCS12 file
download) will be deleted.
##  Value  Operation
##    0    Installed Identity certificate remain valid (Default)
##    1    Installed Identity certificate is removed.
## This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
## SET DELETE_MY_CERT 1
##


## BLOCK_CERTIFICATE_WILDCARDS specifies whether the endpoint will accept server identity
certificates with wildcards.
##  Value  Operation
##    0    Accept wildcards in certificate (default)
##    1    Do not accept wildcards in certificates
## This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET BLOCK_CERTIFICATE_WILDCARDS 1

##################  TLS SETTINGS  ##################
##


##
## TLS_VERSION controls TLS version used for all TLS connections (except SLA monitor agent)
##  Value  Operation
##  0      TLS versions 1.0 and 1.2 are supported (default).
##  1      TLS version 1.2 only is permitted.
## This parameter is supported by:
##      Avaya Vantage Connect Application SIP R2.1.0.0 and later
##      Avaya Equinox 3.5.5 and later
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      J169/J179 H.323 R6.7 and later
##      Avaya Vantage Devices SIP R1.0.0.0 and later; Not used by Avaya Vantage Open application.
##      96x1  SIP  R7.0.1.0 and later releases
##      96x1 H.323 R6.6.2 and later releases
##      B189 H.323 R6.6.2 and later releases
## SET TLS_VERSION  1


################  HTTP PROXY SERVER SETTINGS  ##############
```

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```
##
## HTTPPROXY specifies the address of the HTTP proxy server used by SIP
##  telephones to access an SCEP server that is not on the enterprise network.
##  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 supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; HTTPPROXY is NOT supported for SCEP, but
for Android HTTP based applications.
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later; HTTPPROXY is NOT supported for SCEP, but for WEB Browser and
Exchange.
##       96x0 SIP R1.0 and later
##  Note that in H.323 telephones, SCEP uses WMLPROXY.
## SET HTTPPROXY proxy.mycompany.com
##
## HTTPEXCEPTIONDOMAINS specifies a list of one or more domains,
##  separated by commas without any intervening spaces,
##  for which HTTPPROXY will not be used.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; HTTPPROXY is NOT supported for SCEP, but
for Android HTTP based applications.
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later; HTTPEXCEPTIONDOMAINS is NOT supported for SCEP, but for WEB
Browser and Exchange.
##       96x0 SIP R1.0 and later
##  Note that in H.323 telephones, SCEP uses WMLEXCEPT.
## SET HTTPEXCEPTIONDOMAINS mycompany.com
##

######################  STUN SETTINGS  ######################
##
## The parameters below are applicable only when 3PCC_SERVER_MODE=0 or 3PCC_SERVER_MODE=2 and
ENABLE_3PCC_ENVIRONMENT=1
## BroadSoft mode (3PCC_SERVER_MODE=1) is not supported.
##
## STUN_SERVER_ADDRESS specifies STUN Server address, in IPv4, IPv6 or FQDN format. The default
is "".
##  The remaining NAT/STUN-related parameters in this section will be used by the phone ONLY if
##  STUN_SERVER_ADDRESS is set, and will be ignored otherwise.
##  If specified as an IPv4 address, the STUN server port will default to 3478.
## Valid characters are: ## 0-9, a-z, A-Z, .:
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later
## SET STUN_SERVER_ADDRESS stun.aura.avaya.com
##
## STUN_UDP_INITIAL_TIMEOUT_MSEC specifies initial timeout, in milliseconds, to wait for a
Response to a STUN Request sent over UDP.
## The timeout value is internally doubled after each retransmission.
## Valid values are 500 (1/2 sec) through 3000 (3 sec); the default value is 500.
##  This parameter is supported by:
```

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```
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later
## SET STUN_UDP_INITIAL_TIMEOUT_MSEC 1000
##
## STUN_UDP_MAX_TRANSMISSIONS specifies the number of times the phone will transmit a STUN
Request until a Response is received, after
## which the Request will be treated as failed.
## Valid values are 1 through 7; the default value is 7.
## This parameter is supported by:
##      J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later
## SET STUN_UDP_MAX_TRANSMISSIONS 3
##
## STUN_UDP_MAX_MEDIA_TRANSMISSIONS specifies the number of times the phone will transmit a STUN
Request to get NAT bindings for the phone's RTP/RTCP IP address and ports.
##  Retransmissions continue until a response is received, or until a total number of requests
have been sent.
##  Initial timeout, in milliseconds, to wait for a Response to a STUN Request sent over UDP
for  media is 500 msec. The timeout value is internally doubled after each [re]transmission.
## Valid values are 1 through 4; the default value is 3.
## This parameter is supported by:
##      J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later
## SET STUN_UDP_MAX_MEDIA_TRANSMISSIONS 3
##
## NAT_SIGNALING_KEEPALIVE_ENABLED Specifies whether or not the telephone sends keep-alives to
refresh NAT bindings for the phone's private signaling IP address and port.
##  Value Operation
##    0   Keep-alive messages are not sent
##    1   Keep-alive messages are sent (default)
## This parameter is supported by:
##      J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later
## SET NAT_SIGNALING_KEEPALIVE_ENABLED 0
##
## NAT_SIGNALING_KEEPALIVE_OVERRIDE_SEC specifies interval, in seconds, between keep-alives used
to refresh NAT bindings for the phone's private signaling IP address and port.
##  Valid values are 15 through 900 (15 minutes); the default value is 29.
##  Value     Operation
##  15-900    The phone will use this value as the keep-alive interval for every SIP registration
and dialog
## This parameter is supported by:
##      J100 SIP R4.0.0.2 and later, J159 SIP R4.0.3.0 and later
## SET NAT_SIGNALING_KEEPALIVE_OVERRIDE_SEC 20
##
######################  SCEP SETTINGS  ######################
##
##   Note for all endpoints with exception of J100 SIP R4.0.1.0/96x1 SIP 7.1.5.0 and later:
##    When FIPS_ENABLED is set to 1 (for endpoints which support FIPS mode), SCEP shall not be
used.
##   Note for J100 SIP R4.0.1.0/96x1 SIP 7.1.5.0 and later:
##    When FIPS_ENABLED is set to 1 (for endpoints which support FIPS mode), SCEP is supported.
An identity certificate
##    can be generated using SCEP when FIPS_ENABLED is set to 1. SCEPENCALG must be set to 1.
##
##   General note: If identity certificate was generated before FIPS_ENABLED is set to 1, the
phone will keep using it.
##   It is NOT recommended to use identity certificate generated using SCEP when FIPS_ENABLED is
0 and then
```

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```
##   configure the phone to work in FIPS mode (FIPS_ENABLED==1). It is recommended to CLEAR
(return to factory defaults)
##   before configuring the phone to FIPS mode (FIPS_ENABLED==1). Only J100 SIP R4.0.1.0/96x1 SIP
7.1.5.0 and later supports SCEP
##   when FIPS_ENABLED is set to 1. Other endpoints will need to download a PKCS12 file from a CA
configured to work in FIPS mode.
##   J100 SIP R4.0.1.0/96x1 SIP 7.1.5.0 and later also support download of a PKCS12 file.
##
## MYCERTURL specifies the URL of the SCEP server from which
##  the telephone should obtain an identity certificate,
##  if it does not already have one from that server.
##  Zero to 255 ASCII characters; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later  - the URL can be https or http.
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; the URL can be https or http. Avaya
Vantage Open application does not use identity certificate.
##       J129  SIP  R1.0.0.0 up to R1.1.0.0 (excluded) - the URL can be only http; R1.1.0.0 and
later - the URL can be https or http.
##       96x1 H.323 R6.0 and later
##       H1xx  SIP  R1.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 up to R7.1.0.0 (excluded) - the URL can be only http; R7.1.0.0 and later
- the URL can be https or http.
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
## SET MYCERTURL http://certsrvr.trustus.com/mscep/mscep.dll
## SET MYCERTURL https://10.10.10.10/certsrv/mscep/mscep.dll
##
## MYCERTCN specifies the Common Name (CN) used in the SUBJECT of an SCEP
##  certificate request. The value must be a string that contains either
##  "$SERIALNO" (which will be replaced by the telephone's serial number) or
##  "$MACADDR" (which will be replaced by the telephone's MAC address),
##  but it may contain other characters as well, including spaces.
##  Eight ("$MACADDR") to 255 characters; the default value is "$SERIALNO".
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; Avaya Vantage Open application does not
use identity certificate.
##       96x1 H.323 R6.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
##       Note that prior to R2.6.8, 96x0 SIP releases only support
##        "$MACADDR" or "$SERIALNO" as a value, not additional characters.
## SET MYCERTCN "Avaya telephone with MAC address $MACADDR"
##
## MYCERTDN specifies the part the SUBJECT of an SCEP certificate request
##  that is common for all telephones. It must begin with a "/" and may
##  include Organizational Unit, Organization, Location, State and Country.
##  Zero to 255 ASCII characters; the default value is null ("").
```

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```
##  Note: It is recommended that "/" be used as a separator between components.
##        Commas have been found to not work with some servers.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; Avaya Vantage Open application does not
use identity certificate.
##       96x1 H.323 R6.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
## SET MYCERTDN /C=US/ST=NJ/L=MyTown/O=MyCompany
##
## MYCERTCAID specifies an identifier for the CA certificate with which
##  the SCEP certificate request is to be be signed, if the server hosts
##  multiple Certificate Authorities.
##  Zero to 255 ASCII characters; the default value is "CAIdentifier".
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; Avaya Vantage Open application does not
use identity certificate.
##       96x1 H.323 R6.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
## SET MYCERTCAID EjbSubCA
##
## MYCERTKEYLEN specifies the bit length of the public and private keys
##  generated for the SCEP certificate request.
##  4 ASCII numeric digits, "1024" through "2048"; the default value is "1024".
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later; only "2048" is supported
##       J169/J179 H.323 R6.7 and later; only value "2048" is supported.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; only value "2048" is supported.
##       96x1 H.323 R6.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 and later; default value is "2048" in 96x1 SIP R6.5+. 96x1 SIP R7.1.0.0
and later - only "2048" is supported.
##       H1xx  SIP  R1.0 and later; default value is "2048" in H1xx SIP R1.0.1+.
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
## SET MYCERTKEYLEN 1024
##
## MYCERTRENEW specifies the percentage of the identity certificate's
##  Validity interval after which renewal procedures will be initiated.
##  Valid values are 1 through 99; the default value is 90.
##  This parameter is supported by:
```

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```
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       B189 H.323 R6.6 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R3.1 and later
##       96x0  SIP  R1.0 and later
## SET MYCERTRENEW 90
##


##
## SCEPPASSWORD specifies the password to be included (if not null)
##  in the challengePassword attribute of an SCEP certificate request.
##  Values may contain 0 to 32 ASCII characters (50 ASCII characters in 96x1/B189 H.323 6.6 and
later);
##  the default value is "$SERIALNO".
##  If the value contains "$SERIALNO", it will be replaced by the telephone's serial number.
##  If the value contains "$MACADDR", it will be replaced by the telephone's MAC address in hex.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; please note that if SCEP is configured and SCEPPASSWORD is
empty,
##                     the user will be prompted to enter the SCEP password.
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0
##       Avaya Vantage Devices SIP R1.0.0.0 and later; please note that if SCEP is configured and
SCEPPASSWORD is empty,
##                     the user will be prompted to enter the SCEP password. Avaya Vantage Open
application does not use identity certificate.
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.2 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R3.1 and later
##       B189 H.323 R6.6 and later
##  Note that to maintain the security of the password, this parameter should
##       not be set in a file that is accessible on an enterprise network,
##       it should only be set in a restricted staging configuration.
## SET SCEPPASSWORD "$SERIALNO"
##


## SCEPENCALG specifies SCEP Encryption Algorithm.
##  Value  Operation
##    0    DES (default)
##    1    AES-256
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.4.0 and later
## SET SCEPENCALG 1
##

######################  PKCS12 SETTINGS  ######################
##
## PKCS12URL specifies the URL to be used to download a PKCS #12 file
```

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```
## containing an identity certificate and its private key.
## 0 to 255 ASCII characters, zero or one URL. The value can be a string that contains either
## "$SERIALNO" (which will be replaced by the telephone's serial number) or "$MACADDR"
## (which will be replaced by the telephone's MAC address), but it may contain other characters
as well.
## If $MACADDR is added to the URL then the PKCS12 filename on the file server shall include MAC
address
## without colons (i.e., 6 pairs of ASCII hexadecimal characters AABBCCDDEEFF with hex characters
A-F
## encoded as upper-case characters). For example, if Ethernet MAC address of a specific phone
## is: 00-24-D7-E4-2E-98 and the PKCS12URL is: pkc12file_$MACADDR.cer, then the filename of the
## PKCS12 file for this phone on the file server shall be: pkc12file_0024D7E42E98.cer.
## PKCS12 file download is preferred over SCEP if PKCS12URL is defined.
## SIP endpoints note:
##   An empty parameter value means that PKCS#12 identity certificate download is disabled (if
there is
##   an already existing PKCS12 file on the phone then it will not be deleted if PKCS12URL is set
to "".
##   DELETE_MY_CERT shall be set to 1 or CLEAR procedure shall be used to delete existing PKCS12
file).
##   If the parameter is not empty, PKCS#12 file installation is preferred over SCEP.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0;        Same note as for 96x1 SIP R7.1.0.0 below.
##       96x1 SIP R7.1.0.0 and later; The URL can specify the file server using an IPv4/IPv6
address or an FQDN.
##       Avaya Vantage Devices SIP R1.0.0.0 (build 2304) and later; Avaya Vantage Open
application does not use identity certificate.
##       J129  SIP  R1.0.0.0 and later
##       96x1 H.323 R6.6 and later
## SET PKCS12URL pkc12file_$MACADDR.cer
## SET PKCS12URL http://example.com/exampledir/pkc12file_$MACADDR.cer
## SET PKCS12URL https://example.com/exampledir/pkc12file_$SERIALNO.cer
##
## PKCS12_PASSWD_RETRY specifies the number of retries for entering PKCS12 file password.
## Values: 0-100 and the default is 3. 0 means no retry.
## 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.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 (build 2304) and later; Avaya Vantage Open
application does not use identity certificate.
## SET PKCS12_PASSWD_RETRY 4
##

#####################  802.1X SETTINGS  ####################
##
## DOT1XSTAT specifies the 802.1X Supplicant operating mode.
##  Value  Operation
##    0    Supplicant disabled (default, unless indicated otherwise below)
##    1    Supplicant enabled, but responds only to received unicast EAPOL messages
```

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```
##    2    Supplicant enabled; responds to received unicast and multicast EAPOL messages
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R2.0 and later
##       96x0  SIP  R2.0 and later (default was 1 prior to R2.4.1)
## SET DOT1XSTAT 1
##
## DOT1X specifies the 802.1X pass-through operating mode.
##  Pass-through is the forwarding of EAPOL frames between the telephone's
##  Ethernet line interface and its secondary (PC) Ethernet interface
##  Value  Operation
##    0    EAPOL multicast pass-through enabled without proxy logoff (default)
##    1    EAPOL multicast pass-through enabled with proxy logoff
##    2    EAPOL multicast pass-through disabled
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##      Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch, All K155 devices have embedded Ethernet switch.
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R2.0 and later
##       96x0  SIP  R2.0 and later
##  Note: In 96x0 H.323 releases 1.0 through 1.5, DOT1X is supported, but it controls both
Supplicant and pass-through operation.
##        In these releases, operation is as follows:
##        Value  Operation
##          0    Unicast Supplicant and multicast pass-through enabled without proxy logoff
(default)
##          1    Unicast Supplicant and multicast pass-through enabled with proxy logoff
##          2    Unicast or multicast Supplicant operation enabled, without pass-through
## SET DOT1X 1
##
## DOT1XEAPS specifies the authentication method to be used by 802.1X.
##  Valid values are "MD5" (the default) and "TLS".
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1  H.323 R6.2.1 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R3.1.4 and later
##       96x0  SIP  R2.0 and later
##       B189 H.323 R6.6 and later
```

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```
## SET DOT1XEAPS MD5
##


##
################  FIPS SETTINGS  ############################
##
## FIPS_ENABLED specifies whether only FIPS-approved cryptographic algorithms will be supported.
##  Value  Operation
##    0    No restriction on using non FIPS-approved cryptographic algorithms (default)
##    1    Use only FIPS-approved cryptographic algorithms using embedded FIPS 140-2-validated
cryptographic module (Per NIST Certificate #1747,
##         for the exact operational environment used by the endpoint please refer to the Avaya
support team).
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 SIP 7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET FIPS_ENABLED 1
##
################  OCSP (Online Certificate Status Protocol) SETTINGS  ############################
##
## OCSP_ENABLED specifies whether OCSP will be used to check revocation status of certificates.
##  Value  Operation
##    0    OCSP is disabled (default)
##    1    OCSP is enabled. OCSP will be used to check revocation status for the certificates
##         presented by peers for any TLS connection (H.323 signaling over TLS, HTTPS,
##         802.1x with EAP-TLS, SLA Mon agent, IPSec VPN, SSO)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_ENABLED 1
##
## OCSP_ACCEPT_UNK 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 (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_ACCEPT_UNK 1
##
```

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```
## OCSP_NONCE specifies whether a nonce will be included in OCSP requests and expected in OCSP
responses.
##  Value  Operation
##    0    Nonce is NOT added to OCSP packets
##    1    Nonce is added to OCSP packets (Default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_NONCE 1
##
## OCSP_URI specifies the URI of an OCSP responder. The URI  can be an IP address or hostname.
## The default is "".  0 to 255 ASCII characters - zero or one URI.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_URI http://clients1.google.com/ocsp
##
## OCSP_URI_PREF specifies the preferred URI to use for OCSP requests if more than one is
available.
##  Value  Operation
##    1    Use the OCSP_URI first and then the OCSP field of the Authority Information Access
(AIA) extension
##         of the certificate being checked  (Default)
##    2    Use the OCSP field of the Authority Information Access (AIA) extension of the
##         certificate being checked first and then OCSP_URI
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_URI_PREF 0
##
## OCSP_TRUSTCERTS specifies list of OCSP trusted certificates which are used as
## OCSP signing authority for the certificate that its revocation status is being checked.
## This is needed in case the OCSP responder uses a different CA than the root CA of the
certificate that
## its revocation status is being checked.
## 0 to 255 ASCII characters: zero or more file names or URLs, separated by commas without any
intervening spaces
## The default is "".
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later
##       96x1 H.323 R6.6 and later
## SET OCSP_TRUSTCERTS ocsp.cer
##
## OCSP_HASH_ALGORITHM specifies the hashing algorithm for OCSP request.
```

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```
##  Value  Operation
##    0    SHA-1 (default)
##    1    SHA-256
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP 7.1.0.0 and later
## SET OCSP_HASH_ALGORITHM 1
##
## OCSP_USE_CACHE specifies if OCSP caching is used.
##  Value  Operation
##    0    Do not to use OCSP caching. Always check with OCSP responder.
##    1    Use OCSP cache caching (default).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP 7.1.0.0 and later
## SET OCSP_USE_CACHE 1
##
## OCSP_CACHE_EXPIRY specifies the cache expiry in minutes.
## Valid values: 60 to 10080 (60 min to 7 days) with default 2880 (2 days).
## Note that OCSP response cache expiry uses nextUpdate value in OCSP response message. Only if
nextUpdate is not present will the OCSP_CACHE_EXPIRY parameter value be used.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R7.1.0.0 and later
## SET OCSP_CACHE_EXPIRY 1
##
################  PUSH INTERFACE SETTINGS  #################
##
## TPSLIST (Trusted Push Server List) specifies a list of URI authority components
##  (optionally including scheme and path components) to be trusted.
##  A URI received in a Push Request will only be 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 (the default), Push will be disabled.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0.1 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.2, R2.5 and later
##       16xx H.323 R1.0 and later
## SET TPSLIST 135.20.21.20,push.avaya.com,http://135.20.21.33:80,http://apps.avaya.com/push
##
## SUBSCRIBELIST specifies a list of URIs to which the telephone will send a Subscribe
##  message (an HTTP GET for the URI with the telephone's MAC address, extension number,
##  IP address and model number appended as query values) after the telephone successfully
##  registers with a call server, or when a "subscribe" Push Request is received with
##  a type attribute of "all". The list can contain up to 255 characters.
##  Values are separated by commas without any intervening spaces.
##  If the value of SUBSCRIBELIST is null (the default), Subscribe messages will not be sent
```

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```
##  after registration or in response to a Push Request with a type attribute of "all".
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0.1 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.2, R2.5 and later
##       16xx H.323 R1.0 and later
## SET SUBSCRIBELIST http://135.20.21.21/subscribe,http://push.avaya.com/clients
##
## PUSHCAP allows the modes of individual Push types supported by the telephone to be controlled.
##  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. A digit of 0 means that all Push requests will be rejected for that push
type.
##  A digit of 1 means that only Push requests with a mode of "barge" will be accepted for that
push type.
##  A digit of 2 means that Push requests with a mode of "barge" or "normal" will be accepted for
that push type.
##  The Push types controlled by each digit are as follows:
##  11111
##  ||||+- 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.
##  J159 SIP R4.0.3.0 and later support 5-digit values (default is  00000) as J169/J179 SIP
R4.0.1.0
##  J100 SIP R4.0.1.0 and later support 5-digit values (default is  00000), as in R3.0.0.0 and
later, but also support by J169/J179 only display (WML browser) Push requests - Both "22222" and
"00222" values have the same meaning.
##  J100 SIP R3.0.0.0 and later support 5-digit values (default is  00000), Only receive audio
Push requests and top line Push requests are supported. Both "22222" and "00202" values have the
same meaning.
##  J169/J179 H.323 R6.7 and later support 3 and 4-digit values (default is  2222).
##  J169/J179 SIP R1.5.0 support 4-digit values (default is  0000)
##  96x1 H.323 R6.0 and later support 3 and 4-digit values (default is  2222).
##  96x1  SIP  R6.2 and later support       4-digit values (default is  0000).
##  96x1  SIP  R6.0.x         support       4-digit values (default is  0000)
##               but not display Push, so valid values are 0000 through 2202.
##  96x0 H.323 R2.0 and later support 3 and 4-digit values (default is  2222).
##  96x0  SIP  R2.2, R2.5 and later support 5-digit values (default is 00000).
## SET PUSHCAP 2222
##
## PUSHPORT specifies the TCP port number to be used by the HTTP server in the telephone for
Push.
##  Valid values are 80 through 65535; the default value is 80
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0.1 and later
##       96x0 H.323 R2.0 and later
##       96x0  SIP  R2.2, R2.5 and later
## SET PUSHPORT 80
```

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```
##
################  HTTP/S WEB SERVER  #################
##
## ENABLE_WEBSERVER specifies whether the HTTP/S WEB Server is enabled or disabled.
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later;
If the phone boots up in 3PCC environment and ENABLE_WEBSERVER is not explicitly set 0, it will
be internally set to 1.
##                                    This is to enable web server by default in 3PCC
environments.
## SET ENABLE_WEBSERVER 1
##
## WEBSERVER_ON_HTTP specifies whether HTTP access to the Web Interface is enabled or disabled.
## The WEB Server will be accessible using HTTP as long ENABLE_WEBSERVER and WEBSERVER_ON_HTTP
are set to 1.
## The WEB Server will be accessible using HTTPS as long ENABLE_WEBSERVER is set to 1 AND
(Identity certificate is installed in factory or
##  using WEB/SCEP/PKCS12 file download).
##  Value  Operation
##    0    Web Server will not be accessible via HTTP
##    1    Web Server will be accessible via HTTP (Default)
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET WEBSERVER_ON_HTTP 0
##
## WEB_HTTP_PORT specifies the HTTP port on which the Web Server running on the phone will be
accessed using HTTP.
##  Valid values are 80-65535. The default value is 80.
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET WEB_HTTP_PORT 81
##
## WEB_HTTPS_PORT specifies the HTTPS port on which the Web Server running on the phone will be
accessed using HTTPS.
##  Valid values are 443-65535. The default value is 443.
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET WEB_HTTPS_PORT 444
##
## FORCE_WEB_ADMIN_PASSWORD specifies the password to access the phone through Web as
Administrator.
## From settings file, FORCE_WEB_ADMIN_PASSWORD will be used instead of WEB_ADMIN_PASSWORD
(configured from the Web Interface).
## As long as FORCE_WEB_ADMIN_PASSWORD is configured in the Settings file, it will be used as the
Web admin password.
## It will overwrite any password user might have configured from the Web Interface.
## Valid values are: 8 to 31 alphanumeric characters including upper, lower and special
characters.
## Special characters allowed:~!@#$%^&*_-+=`|\(){}[]:;'<>,.?/. The default is "27238".
## Note: WEB_ADMIN_PASSWORD has no interaction with PROCPSWD or ADMIN_PASSWORD.
##  This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET FORCE_WEB_ADMIN_PASSWORD HelloWorld!01
```

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```
##

################  IDLE TIMER SETTINGS  ###################
##
## BAKLIGHTOFF 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; the default value is 120 (2 hours).
##  A value of 0 means that the display backlight will not be turned off automatically when the
phone is idle.
##  For ENERGY STAR® compliance on applicable phones, a value of 20 is recommended.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only); J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; default is changed to 10 minutes in
R2.0.0.0. The range for K155 is 1-60.
##                                R2.0.1+ - range is 0-999 with default 60 for all Avaya Vantage
devices including K155.
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.2 and later
##       B189 H.323 R1.0 and later
##       H1xx  SIP  R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET BAKLIGHTOFF 60

## SCREENSAVERON specifies the number of minutes of idle time after which the screen saver will
be displayed.
##  If an image file has been downloaded based on the SCREENSAVER (H.323), LOGOS and CURRENT_LOGO
##  (for 96x0 R1.0 SIP and later, 96x1 R6.0 SIP and later and J169/J179 SIP R1.5.0) or
SCREENSAVER_IMAGE
##  (for J100 SIP R2.0 and later) parameters, it will be used as the screen saver.
##  Otherwise, the built-in Avaya one-X(TM) screen saver will be 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.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       J169/J179 SIP R1.5.0
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later, but not supported by the 9610
##       96x0  SIP  R1.0 and later
## SET SCREENSAVERON 480

## SCREENSAVER_IMAGE specifies a list of screensaver images. The default value is "".
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
##                        Up to 5 background images are supported. Only jpeg/jpg files are
supported.
```

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```
##                          The maximum size of any jpeg file is 256 KB. The filenames are case
insensitive.
##                          J159/J169/J179 screen resolution is 320 pixels x 240 pixels. J179
color depth is 16 bits.
##                          J129 screen resolution is 128 pixels x 32 pixels.
##                        The files shall be stored in the same directory defined by HTTPDIR /
TLSDIR.
## SET SCREENSAVER_IMAGE "screensaver_example1.jpg,screensaver_example2.jpeg"
##
## SCREENSAVER_IMAGE_DISPLAY specifies the administrator choice of screensaver image.
## The filename shall be one of the filenames listed in SCREENSAVER_IMAGE.
## If SCREENSAVER_IMAGE_SELECTABLE is set to 1 then the end user may override this setting.
## The default value is "".
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
## SET SCREENSAVER_IMAGE_DISPLAY screensaver_example1.jpg
##
## SCREENSAVER_IMAGE_SELECTABLE specifies whether end users are allowed to choose screensaver
images
## (and overrides administrator choice as configured using SCREENSAVER_IMAGE_DISPLAY parameter).
##  Value      Operation
##    0        End user is not allowed to choose screensaver image and will not see the
screensaver image selection in the Settings -> Display menu.
##    1        End user is allowed to choose the screensaver image from the Settings -> Display
menu (Default)
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J159 SIP R4.0.3.0 and later
## SET SCREENSAVER_IMAGE_SELECTABLE 0
##


##
## BACKLIGHT_SELECTABLE specifies whether backlight timer will be determined per administrator
(BAKLIGHTOFF) or user configuration.
##  Value      Operation
##    0        "Backlight timer" value (BAKLIGHTOFF) will be obtained from 46xxsettings.
##    1        "Backlight timer" value (BAKLIGHTOFF) can be set by user using "User Menu-
>Settings->Display" submenu (Default)
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
## SET BACKLIGHT_SELECTABLE 0

#############  PHONE LOCK SETTINGS (SIP ONLY)  #############
##
## ENABLE_PHONE_LOCK specifies whether a softkey (on the idle Phone screen) and
##  a feature button will be displayed to allow the user to manually lock the phone.
##  Value  Operation
##    0    Disabled: Lock softkey and feature button will not be displayed (default)
##    1    Enabled:  Lock softkey and feature button will be displayed
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later; Please note that on J129 the Lock
option appears
##         in the main menu. There is no Lock softkey or feature button.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; Please note that ENABLE_PHONE_LOCK is used
as enable/disable
```

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```
##         of lock screen when Avaya Breeze Client SDK application defined in
ACTIVE_CSDK_BASED_PHONE_APP is installed.
##       96x1 SIP R6.0 and later
##       96x0 SIP R2.5 and later
##       H1xx SIP R1.0 and later, Please note that ENABLE_PHONE_LOCK is used on H1xx as
enable/disable
##                                of lock screen.
## SET ENABLE_PHONE_LOCK 1
##
## PHONE_LOCK_IDLETIME specifies the interval of idle time, in minutes, after which
##  the phone will automatically lock if the value of ENABLE_PHONE_LOCK is 1.
##  A value of 0 means that the phone will not lock automatically.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; valid values are 0 through 10080; the default value is 0. The
parameter is supported
##         no matter what is the ENABLE_PHONE_LOCK value is.
##       J169/J179 SIP R1.5.0; valid values are 0 through 10080; the default value is 0.
##       Avaya Vantage Devices SIP R1.0.0.0 and later; valid values are 1 through 10080; the
default value is 60. Please note PHONE_LOCK_IDLETIME
##         specifies the maximum interval of idle time, in minutes, allowed for user
configuration (unless exchange policy enforces lower number).
##         User can choose smaller value than this value in the settings application. By default,
user choice is 5 minutes.
##       96x1 SIP R6.2 and later; valid values are 0 through 10080; the default value is 0.
##       96x1 SIP R6.0.x;         valid values are 0 through   999; the default value is 0.
##       96x0 SIP R2.5 and later; valid values are 0 through   999; the default value is 0.
##       H1xx SIP R1.0 and later; valid values are 1 through 10080; the default value is 60.
Please note PHONE_LOCK_IDLETIME
##         specifies the maximum interval of idle time, in minutes, allowed for user
configuration (unless exchange policy enforces lower number).
##         User can choose smaller value than this value in the settings application. By default,
user choice is 5 minutes.
## SET PHONE_LOCK_IDLETIME 30
##


############  CODEC AND RTP SETTINGS (SIP ONLY)  ############
##
## ENABLE_G711A specifies whether the G.711 a-law codec is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled (default)
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later (in all environments - Aura, IP
Office and OpenSIP)
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET ENABLE_G711A 0
##
## ENABLE_G711U specifies whether the G.711 mu-law codec is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled (default)
```

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```
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later (in all environments - Aura, IP
Office and OpenSIP)
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET ENABLE_G711U 0
##
## ENABLE_G722 specifies whether the G.722 codec is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later (in all environments - Aura, IP
Office and OpenSIP)
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later; the default value is 1.
##       96x1 SIP R6.2 and later; the default value is 1.
##       96x1 SIP R6.0.x;         the default value is 0.
##       96x0 SIP R2.0 and later; the default value is 0.
##       H1xx SIP R1.0 and later; the default value is 1.
## SET ENABLE_G722 1
##
## ENABLE_G726 specifies whether the G.726 codec is enabled.
##  Value  Operation
##    0    Disabled (default for 96x0 R1.0)
##    1    Enabled (default for all other releases and models)
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later (in all environments - Aura, IP
Office and OpenSIP)
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later; For IP office environment this parameter shall be set to 0 as
G.726 is not supported by IP Office.
## SET ENABLE_G726 0
##
## G726_PAYLOAD_TYPE specifies the RTP payload type to be used for the G.726 codec.
##  Valid values are 96 through 127; the default value is 110.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##      H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET G726_PAYLOAD_TYPE 111
##
## ENABLE_G729 specifies whether the G.729A codec is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled without Annex B support (default)
##    2    Enabled with Annex B support
```

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```
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later (in all environments - Aura, IP
Office and OpenSIP)
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET ENABLE_G729 0
##
## ENABLE_OPUS specifies whether the OPUS codec is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled WIDEBAND_20K (default value).
##    2    Enabled NARROWBAND_16K
##    3    Enabled NARROWBAND_12K
##  This parameter is supported by:
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; supported in both Aura and IPO
environments. R2.0.1.0 and later in all environments - Aura, IP Office and OpenSIP.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later;
##           Up to R2.0.0.0 (excluded) for IP office and 3PCC environments this parameter shall
be set to 0 (As OPUS is supported in Avaya Aura environment only).
##           R2.0.0.0 and later supports OPUS in all environments.
##       Avaya Equinox 3.1.2 and later
## SET ENABLE_OPUS 0
##
## OPUS_PAYLOAD_TYPE specifies the RTP payload type to be used for the OPUS codec.
##  Valid values are 96 through 127; the default value is 116.
##  This parameter is used when media offer is sent to the far end
##  in an INVITE (or 200 OK when INVITE with no SDP is received).
##  This parameter is supported by:
##      Avaya Vantage Connect Application SIP R2.1.0.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later
## SET OPUS_PAYLOAD_TYPE 111
##
## SEND_DTMF_TYPE specifies whether 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 (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0 and later
##       96x0 SIP R1.0 and later
##       H1xx SIP R1.0 and later
## SET SEND_DTMF_TYPE 1
##
## DTMF_PAYLOAD_TYPE specifies the RTP payload type to be used for RFC 2833 signaling.
##  Valid values are 96 through 127; the default value is 120.
##  This parameter is supported by:
```

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```
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET DTMF_PAYLOAD_TYPE 121
##
## SYMMETRIC_RTP specifies whether or not the telephone should discard
##  received RTP/SRTP datagrams if their UDP Source Port number is not
##  the same as the UDP Destination Port number that the telephone is
##  including in RTP/SRTP datagrams intended for that endpoint.
##  Value  Operation
##    0    Ignore the UDP Source Port number in received RTP/SRTP datagrams.
##    1    Discard received RTP/SRTP datagrams if their UDP Source Port number
##          does not match the UDP Destination Port number that the telephone is
##          including in RTP/SRTP datagrams intended for that endpoint (default).
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1 SIP R7.1.1.0 and later (9608 and SIP9611 HW version 3 and higher)
##       96x1 SIP R6.0 and later (hardware version below 3).
##       H1xx SIP R1.0 and later
##       96x0 SIP R2.4 and later
## SET SYMMETRIC_RTP 0
##

##################  OTHER SIP-ONLY SETTINGS  #################
##
## PHNMUTEALERT_BLOCK specifies whether the Mute Alert feature will be Blocked or Unblocked.
##  Value  Operation
##    0    Unblocked
##    1    Blocked (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.0.1 and later.
## SET PHNMUTEALERT_BLOCK 1
##
## MATCHTYPE specifies how a calling party number is compared to the numbers
##  in the user's Contacts to obtain a name to display for the incoming call.
##  Value  Operation (for 96x1 SIP R6.2 to R7.0 (excluded), 96x0 R2.6.5 and later)
##    0    The Contact name is displayed if the rightmost 4 digits of the calling
##          party number match the rightmost 4 digits of a Contacts number (default)
##    1    The Contact name is displayed if the entire calling party number
##          exactly matches the all of the digits in a Contacts number
##  Value  Operation (for 96x1 SIP R7.0 and later, J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179
SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and
later)
##    0    The Contact name is displayed if the entire calling/called party number exactly
matches
##         the number stored in the contact (ELD rules are applied)  (default)
##    1    The Contact name is displayed if all the digits of the shorter number (contacts,
calling/called party number)
```

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```
##         match to the rightmost digits of the longer number (contacts, calling/called party
number).
##    2    The Contact name is displayed if at least 4 rightmost digits of the calling/called
##         party number match the rightmost 4 digits of a Contacts number
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R6.2 and later
##       96x0 SIP R2.6.5 and later.
## SET MATCHTYPE 0

##
################  ETHERNET INTERFACE SETTINGS  #################
##
## PHY1STAT specifies the speed and duplex settings for the Ethernet line interface.
##  Valid values are 1 through 6; the default value is 1.
##  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-negotiate
##  This parameter is supported by:
##       Avaya Vantage SIP R2.1.0.0 and later - values 1,5 are supported. The parameter is
permanently configured to "Auto-Negotiate" on Avaya Vantage SIP R1.1.0.0 up to R2.1.0.0.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later
(values 1-5 only), J139 SIP R3.0.0.0 and later (values 1-5 only), J159 SIP R4.0.3.0 and later
(values 1-5 only)
##       J169/J179 H.323 R6.7 and later
##       H1xx  SIP  R1.0 and later (values 1-5 only)
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.2 and later (values 1-5 only)
##       B189 H.323 R1.0 and later
##       96x1  SIP  R6.0.x
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET PHY1STAT 1
##
## PHY2STAT specifies the speed and duplex settings for the secondary (PC) Ethernet interface.
##  Valid values are 0 through 6; the default value is 1.
##  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-negotiate
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later
(values 0-5 only), J139 SIP R3.0.0.0 and later (values 1-5 only), J159 SIP R4.0.3.0 and later
(values 1-5 only)
##       J169/J179 H.323 R6.7 and later
```

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```
##       Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch (values 0-1 only), All K155 devices have embedded Ethernet switch (values 0-1 are
supported only).
##         Value 5 is also supported in R2.1.0.0 and later.
##       H1xx  SIP  R1.0 and later (values 0-5 only)
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.2 and later (values 0-5 only)
##       96x1  SIP  R6.0.x
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET PHY2STAT 1
##
## PHY2_AUTOMDIX_ENABLED specifies whether auto-MDIX is enabled on PHY2.
##  Valid values are 0 through 1; the default value is 1.
##  Value  Operation
##    0    auto-MDIX is disabled
##    1    auto-MDIX is enabled (default).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.1.0.0 and later for K165/K175 models with embedded Ethernet
switch; All K155 devices have embedded Ethernet switch.
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.3 and later
##       96x1 SIP R6.3 and later
## SET PHY2_AUTOMDIX_ENABLED 1
##


#############  LOCAL PROCEDURE ACCESS SETTINGS  ############
##
## PROCSTAT specifies whether local (craft) procedures can be used to configure the telephone.
##  Value  Operation
##    0    Local procedures can be used (default)
##    1    Local procedures cannot be used
##  Note: Be very careful before setting PROCSTAT to 1
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET PROCSTAT 1
##
## PROCPSWD specifies an access code for access to local (craft) procedures.
##  Valid values contain 0 through 7 ASCII numeric digits.
##  The default value is 27238 (CRAFT) unless indicated otherwise below.
##  A null value implies that an access code is not required for access.
##  Note: Setting this parameter via CM (for H.323) or PPM (for SIP) is more secure
```

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```
##        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 telephone or server software that do not support setting this
##        value from the server.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later (must contain at least 4 digits
##                              else default value 27238 is used)
##       J169/J179 H.323 R6.7 and later (must contain at least 4 digits,
##                                else default value 27238 is used)
##       Avaya Vantage Devices SIP R1.0.0.0 and later (must contain at least 4 digits)
##       96x1 H.323 R6.0 and later (must contain at least 4 digits for R6.2.4 and later,
##                                else default value 27238 is used)
##       96x1  SIP  R6.0 and later (must contain at least 4 digits for R6.3 and later
##                              else default value 27238 is used)
##       H1xx  SIP  R1.0 and later (must contain at least 4 digits else default value
##                              27238 is used)
##       B189 H.323 R1.0 and later (must contain at least 4 digits for R1.0 and later
##                              else default value 27238 is used)
##       96x0 H.323 R1.0 and later (default is null ("") prior to R1.2, must contain at
##                                least 4 digits for R3.2.1 and later else default value
##                              27238 is used)
##       96x0  SIP  R1.0 and later (must contain at least 4 digits for R2.6.10 and later
##                              else default value 27238 is used)
##       16xx H.323 R1.0 and later (default is null ("") prior to R1.3.3, and
##                                 must contain at least 4 digits for R1.3.3 and later)
## SET PROCPSWD 572958
##
## ADMIN_PASSWORD specifies a complex access code for access to local (craft) procedures.
##  Valid values contain 6 and 31 alphanumeric characters including upper, lower and special
characters.
##  The default value is 27238 which implies that PROCPSWD is used as access code for access to
local (craft) procedures.
##  If ADMIN_PASSWORD length is less than 6 or greater than 31, the parameter is treated as not
defined.
##  If ADMIN_PASSWORD is configured, then PROCPSWD is ignored.
##  The special characters supported are: ~!@#$%^&*_-+=`|\(){}[]:;'<>,.?/. " is not supported.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      Avaya Vantage Devices SIP R1.0.0.0 (build 2304) and later; the default value is "".
ADMIN_PASSWORD is not applied on Boot Recovery Menu (BRM).
##       96x1 SIP R7.1.0.0 and later
##      Avaya Vantage Connect Application SIP R1.0.0.0 and later; default value is "".
## Note: For Avaya Vantage Connect Application and Avaya Vantage Devices, if ADMIN_PASSWORD is
not configured and PROCPSWD is not configured, then the
## local (craft) procedures are not accessible.
## Note: The parameter is also used by "Avaya Vantage Connect Application" to allow administrator
## to unpin/pin the application when PIN_APP is defined to "Avaya Vantage Connect Application"
package name.
## SET ADMIN_PASSWORD ComPlexPSWD12?!
##
## ADMIN_LOGIN_ATTEMPT_ALLOWED specifies the number of failed attempts for entering the access
code (PROCPSWD or ADMIN_PASSWORD)
## before the local (craft) procedures will be locked for a period specified by
ADMIN_LOGIN_LOCKED_TIME.
```

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```
## Valid values are 1 to 20, default 10.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET ADMIN_LOGIN_ATTEMPT_ALLOWED 11
##
## ADMIN_LOGIN_LOCKED_TIME specifies the time in minutes that local (craft) procedures are locked
once the
## number of failed attempts for entering the access code (PROCPSWD or ADMIN_PASSWORD) is
reached.
## Valid values 5 min to 1440 min, default 10 min.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## SET ADMIN_LOGIN_LOCKED_TIME 2


####################  SNMP SETTINGS  ######################
##
## SNMPSTRING 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 ("") unless indicated otherwise below.
##  A null value results in SNMP being disabled.
##  Note: Setting this parameter via CM (for H.323) or PPM (for SIP) 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
##        older versions of telephone, CM or PPM software that do not support setting
##        this value from the server.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET SNMPSTRING mystring
##
## SNMPADD specifies a list of source IP addresses from which SNMP query messages
##  will be accepted and processed.
##  Addresses can be in dotted-decimal (IPv4), colon-hex (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 ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
```

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```
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET SNMPADD 192.168.0.22,192.168.0.23
##
#######  LINK LAYER DISCOVERY PROTOCOL (LLDP) SETTINGS  ######
##
## LLDP_ENABLED specifies whether LLDP is enabled.
##  Value  Operation
##    0    Disabled
##    1    Enabled
##    2    Enabled, but only begin transmitting if an LLDP frame is received (default)
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; the default is 1.
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later; the default is 1.
##       96x0  SIP  R2.0 and later
##  Note that the following do NOT support the LLDP_ENABLED parameter,
##  but they always operate consistent with a value of 2 above:
##       96x1 H.323 R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.2 and later
##       16xx H.323 R1.2 and later
## SET LLDP_ENABLED 1
##

##################  EVENT LOGGING SETTINGS  ##################
##

##
## LOGSRVR specifies one address for a syslog server
##  in dotted-decimal (IPv4), colon-hex (IPv6, if supported), or DNS name format.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       H1xx  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET LOGSRVR 192.168.0.15

## LOCAL_LOG_LEVEL 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 will be logged
##   (note that lower numeric severity values correspond to higher severity levels).
##  Value  Operation
```

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```
##    0    Emergency events are logged
##    1    Alert and Emergency events are logged
##    2    Critical, Alert and Emergency events are logged
##    3    Error, Critical, Alert and Emergency events are logged (default)
##    4    Warning, Error, Critical, Alert and Emergency events are logged
##    5    Notice, Warning, Error, Critical, Alert and Emergency events are logged
##    6    Informational, Notice, Warning, Error, Critical, Alert and Emergency events are logged
##    7    Debug, Informational, Notice, Warning, Error, Critical, Alert and Emergency events are
logged
##  Warning: A setting of 7 can impact the performance of the telephone due to the number of
events generated.
##  This parameter is supported by:
##       J129 SIP R1.0.0.0 (or 1.1.0.0), J169/J179 SIP R1.5.0 and J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; No SNMP support in R1.0.0.0. This
parameter affects local log files stored on the device. Values 3-7 are supported.
##        Default value is 4.
##       96x0  SIP  R1.0 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later; No SNMP support in R1.0. This parameter affects local log
files stored on the device.
## SET LOCAL_LOG_LEVEL 3
##
## LOG_CATEGORY specifies a list of categories of events to be logged via syslog and locally.
##  This parameter must be specified to log events below the Error level.
##  The list can contain up to 255 characters. The default is "".
##  Category names are separated by commas without any intervening spaces.
##  See Administrator's guide for additional detail.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later; the default is "ALL" which implies all
categories.
##       96x0  SIP  R1.0 and later
##       96x1  SIP  R6.0 and later
##       H1xx  SIP  R1.0 and later; the default is "ALL" which implies all categories. New
categories for H1xx compare to
##                                    96x1 SIP: "ANDROID" and "KERNEL".
## SET LOG_CATEGORY DHCP,NETMGR,AUDIO
##


###################  AUDIO DEBUG RECORDING  ##################
##
## ENABLE_RECORDING specifies whether audio debug recording is enabled for users.
##  Value  Operation
##    0    Audio debug recording is disabled (default)
##    1    Audio debug recording is enabled
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R1.1.0.0 and later (controls whether audio recording is
enabled or disabled as part of the audio report).
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.3 and later
##       H1xx  SIP  R1.0 and later
## SET ENABLE_RECORDING 1
##
```

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```
## WARNING_FILE 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; the default value is null ("").
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       96x1  SIP  R6.3 and later
## SET WARNING_FILE "Warning.wav"
##
###############  SECURE SHELL (SSH) SETTINGS  ################
##
## Note: The SSH server on the endpoints is used by Avaya Services only for debugging purposes
only.
## The SSH server supports only Avaya Services Logins ("craft" and "sroot").
## By enabling Avaya Services Logins you are granting Avaya access to your endpoints.
## This is necessary required to maximize the performance and value of your Avaya
## support entitlements, allowing Avaya to resolve product issues in a timely manner.
## In addition to enabling the Avaya Logins, the Avaya Product that the endpoints register with
must be registered
## using the Avaya Global Registration Tool (GRT, see https://grt.avaya.com) to be eligible for
Avaya remote connectivity.
## Please see the Avaya support site (support.avaya.com/registration) for additional information
for registering products
## and establishing remote access and alarming.
## By disabling Avaya Services Logins you are preventing Avaya access to
## your endpoints. This is not recommended, as it can impacts Avaya's ability to provide
## support for the product. Unless the customer is well versed in managing the product
## themselves, Avaya Services Logins should not be disabled.
## The access to the SSH server is protected by ASG (Legacy authentication algorithm)
## or EASG (new authentication algorithm). Enhanced Access Security Gateway (EASG) provides a
more secure authentication
## compared to ASG for SSH server access. Endpoints that support EASG are no longer support ASG.
## EASG is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later ("craft" only)
##       J169/J179 H.323 R6.7 and later ("craft" only)
##       Avaya Vantage Devices SIP R1.0.0.0 and later ("craft" and "sroot").
##       96x1  SIP  R7.1.0.0 and later ("craft" only)
##
## SSH_ALLOWED specifies whether SSH is supported.
##  Value  Operation
##    0    Disabled
##    1    Enabled
##    2    Configured using local craft procedure - the SSH server can be enabled or disabled
from local craft procedure.
##         When this mode is configured, then by default the SSH server is disabled.
##  The default of 96x1 H.323 R6.2 up to 6.4 (not included) is 0 (disabled). The default value
for 96x1 H.323 6.4 and later is 2.
##  The default of 96x1 SIP R6.2 and later is 0 (disabled).
##  The default of B189 is 0 (disabled).
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later (values 0-2), the default is 0.
##       J169/J179 H.323 R6.7 and later (values 0-2)
##       J169/J179 SIP R1.5.0 (values 0-1)
```

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```
##       Avaya Vantage Devices SIP R1.0.0.0 and later (values 0-1, default value is 0)
##       96x1 H.323 R6.2 and later (values 0-1), value 2 is added in R6.4 and later.
##       96x1  SIP  R6.2 and later (values 0-1)
##       B189 H.323 R1.0 and later (values 0-1)
##       H1xx  SIP  R1.0 and later (values 0-1, default==0)
## SET SSH_ALLOWED 1
##

## SSH_BANNER_FILE specifies the file name or URL for a custom SSH banner file.
##  If the value is null, a default English banner will be used for SSH.
##  The value can contain 0 to 255 characters; the default value is null ("").
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x1 H.323 R6.2 and later
##       96x1  SIP  R6.2 and later
##       B189 H.323 R1.0 and later
##       H1xx  SIP  R1.0 and later
## SET SSH_BANNER_FILE http://security.myco.com/files/SSH-Banner.txt
##
## SSH_IDLE_TIMEOUT specifies the number of minutes of inactivity
##  after which an SSH connection will be terminated
##  Valid values are 0 through 32767; the default value is 10.
##  A value of 0 means that the connection will not be terminated due to inactivity.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       96x1 H.323 R6.2 and later
##       96x1  SIP  R6.2 and later
##       B189 H.323 R1.0 and later
##       H1xx  SIP  R1.0 and later
## SET SSH_IDLE_TIMEOUT 30
##
## EASG_SITE_CERTS 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.
## 0 to 255 ASCII characters: zero or more file names or URLs, separated by commas
## without any intervening spaces
## The default is "".
##  This parameter is supported by:
##       J129  SIP R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Vantage Devices SIP R1.0.0.0 (build 2304) and later
##       96x1 SIP 7.1.0.0 and later
## SET EASG_SITE_CERTS "mySiteCert.p7b"
##
## EASG_SITE_AUTH_FACTOR specifies Site Authentication Factor code associated with
## the EASG site certificate being installed.
## Valid value: a 10 to 20 character alphanumeric string
```

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```
## Default value: ""
##  This parameter is supported by:
##       J129  SIP R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##      Avaya Vantage Devices SIP R1.0.0.0 (build 2304) and later
##       96x1 SIP 7.1.0.0 and later
## SET EASG_SITE_AUTH_FACTOR  "avaya12345abcd"
##
## CERT_WARNING_DAYS_EASG 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 generated as well.
## Valid values are: 90-730, default is 365 days.
## This parameter is supported by:
##       J129  SIP  R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       96x1 SIP 7.1.0.0 and later
## SET CERT_WARNING_DAYS_EASG 100
##

########  Enhanced Debugging Capabilities Support  #######
##
##  AUTHCTRLSTAT controls whether enhanced debugging capabilities can be activated from the SSH
server by
##  Avaya technicians only. The parameter shall only be set to 1 for the debugging period by
Avaya technicians and
##  shall be configured back to 0 when the debugging period is end.
##  Value  Operation
##  0      Enhanced debugging capabilities are disabled (default).
##  1      Enhanced debugging capabilities are enabled.
## This parameter is supported by:
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      J169/J179 H.323 R6.7 and later
##      Avaya Vantage Devices SIP R1.0.0.0 and later; While this parameter is supported, Avaya
Technician is NOT expected
##      to use it in the field and customers are encouraged to verify it is remain with its
default value.
##      96x1  SIP  R7.0.1.0 and later releases (hardware version 3 and up).
##      96x1 H.323 R6.6.2 and later releases (hardware version 3 and up).
## SET AUTHCTRLSTAT  1
##

#####  SERVICE LEVEL AGREEMENT (SLA) MONITOR SETTINGS  #####
##
## Please note that SLA Monitor agent requires specification of the root
## (and intermediate  if applicable) trusted certificates using TRUSTCERTS for verifying
## the SLA Monitor server certificate.
##
## SLMSTAT specifies whether or not the SLA Monitor agent is enabled.
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled
##  This parameter is supported by:
```

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```
##       Avaya Vantage SIP R2.1.0.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  H.323 R6.4 and later
##       96x1  SIP  R6.2 and later
##       96x0 H.323 R3.1.4 and later
## SET SLMSTAT 1
##
## SLMCAP specifies whether the SLA Monitor agent is enabled for packet capture (sniffing).
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled with payloads are removed from RTP packets
##    2    Enabled with payloads included in RTP packets
##    3    Controlled from craft menu - enable of RTP packets capture or disable packets capture
##         using local CRAFT procedures.
##  This parameter is supported by:
##       Avaya Vantage SIP R2.1.0.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later (values 0-3)
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  H.323 R6.4 and later (values 0-3)
##       96x1  SIP  R6.2 to R6.5 (values 0-2)
##       96x1  SIP  R7.0 and later (values 0-3)
##       96x0 H.323 R3.1.4 and later (values 0-2)
## SET SLMCAP 1
##
## SLMCTRL specifies whether the SLA Monitor agent is enabled for device control.
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled
##    2    Controlled from craft menu
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later (values 0-2)
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  H.323 R6.4 and later (values 0-2)
##       96x1  SIP  R6.2 to R6.5 (values 0-1)
##       96x1  SIP  R7.0 and later (values 0-2)
##       96x0 H.323 R3.1.4 and later (values 0-1)
## SET SLMCTRL 1
##
## SLMPERF specifies whether the SLA Monitor agent is enabled for device performance monitoring.
##  Value  Operation
##    0    Disabled (default)
##    1    Enabled
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
```

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```
##       96x1  H.323 R6.4 and later
##       96x1  SIP  R6.2 and later
##       96x0 H.323 R3.1.4 and later
## SET SLMPERF 1
##
## SLMPORT 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.
## Important note: If default port is not used, both the SLA Mon agent and server must
## be configured with the SAME port. SLMPORT impacts the phone's SLA Mon agent configuration.
## A corresponding configuration must also be made on the SLA Mon server agentcom-slamon.conf
file.
##  This parameter is supported by:
##       Avaya Vantage SIP R2.1.0.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  H.323 R6.4 and later
##       96x1  SIP  R6.2 and later
##       96x0 H.323 R3.1.4 and later
## SET SLMPORT 43210
##
## SLMSRVR specifies the IP address and the port number of the SLA Mon server in the
## aaa.bbb.ccc.ddd:n format.
## Set the IP address of the SLA Mon server in the aaa.bbb.ccc.ddd format
## to restrict the registration of agents only to that server. 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, 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 n, set the value of SLMSRVR in the
## aaa.bbb.ccc.ddd:n format, where aaa.bbb.ccc.ddd is the IP address
## of the SLA Mon server.
## Important 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.
##  This parameter is supported by:
##       Avaya Vantage SIP R2.1.0.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##       B189 H.323 R6.7.1 and later
##       J169/J179 H.323 R6.7 and later
##       96x1  H.323 R6.4 and later
##       96x1  SIP  R6.2 and later
##       96x0 H.323 R3.1.4 and later
## SET SLMSRVR 192.168.27.35:50011
##


##############  ENHANCED LOCAL DIALING RULES  ###############
##
## These settings affect certain dialing behaviors, such as
```

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```
## dialing numbers from the incoming Call Log or from web pages
## Please note that the enhanced local dialing rules are not applicable
## when using 96x0/96x1 H.323 phones in IP office environment.
## The parameters below are supported by 16xx H.323 phones.
##
##   Dialing Algorithm Status
##   Controls whether algorithm defined by parameters in
##   this section is used during certain dialing behaviors.
##     0 disables algorithm.
##     1 enables algorithm, but not for Contacts (default). For B189 H.323 only, value 1 has the
same meaning as value 0
##       since B189 does not support call log application and WML browser.
##     2 enables algorithm, including Contacts (96xx SIP R2.0 and later, J129  SIP  R1.0.0.0 (or
R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159
SIP R4.0.3.0 and later
##      96x1 SIP R6.0 and later, 96x1/B189 H.323 6.6 and later, H1xx SIP R1.0 and later,
J169/J179 H.323 R6.7 and later)
##     Note: Avaya Vantage Connect Application SIP R1.0.0.1 and later and Avaya Equinox 3.1.2 and
later support values 0 (disabled) and 1 where value means that enhanced local dialing rules are
applied on
##           all outgoing calls (whether originated from contacts, history or dialer*). All
Enhanced local dialing rules parameters mentioned in this section which are marked as supported
##           by Avaya Vantage Connect Application SIP R1.0.0.1 and later and Avaya Equinox 3.1.2
and later can be supported by any Avaya Breeze Client SDK based application.
##     Note for dialer*: In pre Avaya Vantage Connect Application SIP R2.0.1 the enhanced local
dialing rules are applied for edit dialing, but not off hook dialing.
##           In Avaya Vantage Connect Application SIP R2.0.1 and on the enhanced local dialing
rules are not applied for both off hook dialing end edit dialing.
##
## SET ENHDIALSTAT 1
##
##   Country Code
##   For United States the value is '1'
##   Valid values 1 to 999. The default value is 1.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
## SET PHNCC 1
##
##   Internal extension number length
##   If your extension is 12345, your dial plan length is 5.
##   On 96xx phones, the maximum extension length is 13.
##   This value must match the extension length set on your
##   call server.
##   Valid values are 3-13. The default value is 5.
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
```

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```
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
## SET PHNDPLENGTH 5
##
##   International access code
##   For the United States, the value is 011.
##   Valid values are 0 to 4 dialable characters (0-9,*,#). The default value is "011".
##   This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
## SET PHNIC 011
##
##   Long distance access code
##   Valid values are 0 through 9 and empty string. The default value is 1.
##   if no long distance access code is needed then SET PHNLD "".
##   This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
## SET PHNLD 1
##
##   National telephone number Length
##   For example, 800-555-1111 has a length of 10.
##   Valid values are 5-15. The default value is 10.
##   This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
```

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```
## SET PHNLDLENGTH 10
##
##   Outside line access code
##   The number you press to make an outside call.
##   Valid values are 0 to 2 dialable characters (0-9, *, #). The default value is 9.
##   This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       J169/J179 H.323 R6.7 and later
##       Avaya Equinox 3.1.2 and later; default is "".
##       Avaya Vantage Connect Application SIP R1.0.0.1 and later; default is "".
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R2.5 and later
## SET PHNOL 9
##
## ELD_SYSNUM
##  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 (Default)
##  This parameter is supported by:
##     J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
##     96x1 SIP R7.0.1.2 and later.
##
## SET ELD_SYSNUM 1
##

####################  AUDIO SETTINGS  ######################
##
## Automatic Gain Control (AGC).
##   These settings enable or disable AGC.
##
##   A value of 1 (default) enables AGC. A value of 0 disables AGC.
##   AGCHAND controls handset AGC.
##   AGCHEAD controls headset AGC
##   AGCSPKR controls speaker AGC.
##   Note: AGCHAND, AGCHEAD and AGCSPKR are supported by J169/J179 H.323 R6.7 and later, H1xx SIP
R1.0 and later and Avaya Vantage Devices SIP R1.0.0.0 and later.
##   Note: AGCHAND and AGCSPKR are supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP
R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later. AGCHEAD is
supported by J100 SIP R2.0.0.0 and later (J169/J179 only),
##    J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later.
##   Note: For 96x1 H.323 and J169/J179 H.323- User can also change the "Handset/Headset/Speaker
Auto Gain Control" fields
##   in HOME-> Options & Settings-> Advanced Options -> Automatic Gain Control... menu.
##   AGCHAND/AGCHEAD/AGCSPKR will be enforced only in case user did not change at all the
relevant "Handset/Headset/Speaker Auto Gain Control" field value.
##   Please note that user changes are stored in backup/restore file as "Handset AGC", "Headset
AGC" and "Speaker AGC" (if BRURI has a valid value) which means that if the
##   restored file include "Handset AGC", "Headset AGC" and/or "Speaker AGC" parameters then they
will take precedence over AGCHAND, AGCHEAD and AGCSPKR respectively.
```

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```
##   If BRURI is not valid, but user still change the content of "Handset/Headset/Speaker Auto
Gain Control" fields, then user value will take precedence over
##   AGCHAND, AGCHEAD and AGCSPKR respectively. The only way to clear user configuration in this
case is by doing:
##   a. "CLEAR" operation in CRAFT menu,
##   b. New user login.
## SET AGCHAND 0
## SET AGCHEAD 0
## SET AGCSPKR 0
##
## Audio Environment Index
##   Enables you to customize the telephone's audio
##   performance.  (0-299)  This parameter affects settings
##   for AGC dynamic range, handset and headset noise
##   reduction thresholds, and headset transmit gain. It is
##   highly recommended you consult Avaya before changing
##   this parameter.
##   This parameter is supported by:
##      J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0 and J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      J169/J179 H.323 R6.7 and later
##      96x1  H.323 R6.0 and later
##      96x1  SIP  R6.0 and later
##      96x0 H.323 "0" through "80" (R1.0, R1.1), "0" through "191" (R1.2 - R1.5), "0" through
"299" (R2.0+)
##      96x0 SIP R2.6 and later
##      16xx H.323 R1.3 and later
## SET AUDIOENV 0

#####################  CUSTOM RING TONES  ####################
##
## RINGTONESTYLE specifies the style of ring tones that are offered to the user
##  for Personalized Ringing when "Classic" (as opposed to "Rich") is selected.
##  Value  Operation
##    0    North American ring tones are offered (default)
##    1    European ring tones are offered
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.1.0.0 and later
##       9670 H.323 R2.0 and later
##       96x1 H.323 R6.0 and later
##       96x1 SIP R6.3 and later
##       H1xx SIP R1.0 and later
## SET RINGTONESTYLE 1
##
## RINGTONES 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 may contain spaces, but if any do, the entire list must be quoted.
##  Ring tone files must be single-channel WAV files
```

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```
##   coded in ITU-T G.711 u-law or A-law PCM with 8-bit samples at 8kHz.
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R1.1.0.0 and later; Avaya Vantage supports list of ringtones
files (without the tuple format) if the files are stored in the same directory as FILE_SERVER_URL
points to.
##                                                    If a different directory is needed, then
the tuple format shall be used and the format shall be <filename with suffix>=”path/filename”.
##                                                    For example: “name.wav/mp3/ogg=URI”.
##                                                    If the ringtones files are stored on the
same directory then list of filenames shall be in the following format:
“ring1.wav,ring2.wav,ring3.mp3,rin4.ogg”
##                                                    If there is use of mp3/ogg files which
includes the ID3 metadata container with non-empty title field then this title field will be
##                                                    presented by Android in the list of
ringtones.
##                                                    If the mp3/ogg file include ID3 metadata
container with empty title field then the filename shall be displayed.
##                                                    If wav file is used then the filename is
always presented.
##                                                    When using tuple format then the filename
MUST include the audio file suffix (.mp3/.wav.ogg).
##                                                    Changing file suffix only with the same
filename will not trigger new download (I.e. this is not considered as new file).
##                                                    The filename must be changed in order to
trigger new download of the ringtones files.
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1  SIP  R6.3 and later
##       H1xx SIP R1.0 and later
## SET RINGTONES "Steam Whistle=tones/swhistle.wav,Car
Horn=tones/chorn.wav,Siren=tones/siren.wav"
## Example for Avaya Vantage:
## SET RINGTONES "whistle.wav,chorn.wav,siren.wav"
## SET RINGTONES "you.wav=wavefiles/you_talkin.wav,ring4.mp3=mp3files/ring.mp3"
## Note: In order to set RINGTONES for Avaya Vantage and other phones then shall be use of the IF
conditional statement (e.g. IF $MODEL4 SEQ K175 GOTO SETTINGSK1XX)
## with $MODEL4 to separate between K155/K175/K165 to other phones.
##
## RINGTONES_UPDATE specifies whether the phone will query the file server to determine whether
##  there is an updated version of each custom ring tone file each time the phone starts up or
resets.
##  Value  Operation
##    0    Phone will only try to download ring tones with new display names (default)
##    1    Phone will check for updated version of each ring tone file at startup
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later (J169/J179 only), J159 SIP R4.0.3.0
and later
##       96x1 SIP R6.3 and later.
## SET RINGTONES_UPDATE 1
##
## PROVIDE_CF_RINGTONE specifies whether 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
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0
and later, J159 SIP R4.0.3.0 and later
```

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```
##           The user option is displayed in "Settings > Audio > Personalize ringing > Call Fwd
Ring".
##       96x1 SIP R6.3 and later.
## SET PROVIDE_CF_RINGTONE 1
##

##################  FILE SERVER SETTINGS  ##################
##
## HTTP Server Addresses
##   [If you set your HTTP Server Addresses via DHCP, do not
##   set them here as they will override your DHCP settings.
##   Server used to download configuration script files.
##   Zero or more HTTP server IP addresses in dotted-decimal,
##   colon-hex (96x1 H.323 R6.0 onwards), or DNS name format,
##   separated by commas without any intervening spaces.
##   (0 to 255 ASCII characters, including commas).
##   This parameter may also be changed via LLDP.
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       B189 H.323 R1.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## SET HTTPSRVR 192.168.0.5
##
## HTTP Server Directory Path
##   Specifies the path name to prepend to all file names
##   used in HTTP and HTTPS GET operations during startup.
##   (0 to 127 ASCII characters, no spaces.)
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP
R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later,
J169/J179 H.323 R6.7 and later.
## SET HTTPDIR myhttpdir
##
## HTTP port
##   Sets the TCP port used for HTTP file downloads from
##   non-Avaya servers. (0-65535) The default value is 80.
##   Applies only to 96xx phones, 96x1 phones, J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP
R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later,
J169/J179 H.323 R6.7 and later.
## SET HTTPPORT 80
##

##
## Server Authentication
##   Sets whether script files are downloaded from an
##   authenticated server over an HTTPS link.
##   0 for optional, 1 for mandatory
```

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```
##   Note: This parameter is also supported by J169/J179 H.323 R6.7 and later, J129  SIP
R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later,
##    H1xx SIP R1.0 and later and Avaya Vantage Devices SIP R1.0.0.0 and later.
## SET AUTH 0
##
## HTTPS Server Addresses
##   [If you set your HTTP/S Server Addresses via DHCP, do not
##   set them here as they will override your DHCP settings.
##   Server used to download configuration script files.
##   Zero or more HTTPS server IP addresses in dotted-decimal,
##   colon-hex, or DNS name format,
##   separated by commas without any intervening spaces.
##   (0 to 255 ASCII characters, including commas).
##   This parameter may also be changed via LLDP.
##   This parameter is supported by:
##       J129  SIP  R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.1.0.0 and later
##       96x1  SIP  R7.1.0.0 and later
## SET TLSSRVR 192.168.0.5
##
## HTTPS Server Directory Path
##   Specifies the path name to prepend to all file names
##   used in HTTPS GET operations during startup.
##   (0 to 127 ASCII characters, no spaces.)
##   This parameter is supported by:
##       J129  SIP  R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1  SIP  R6.0 and later
##       96x0  SIP  R1.0 and later
## SET TLSDIR myhttpdir
##
## HTTPS port
##   Sets the port used for HTTPS file downloads from
##   non-Avaya servers. (0-65535) The default value is 443.
##   This parameter is supported by:
##       J129  SIP  R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Devices SIP R1.0.0.0 and later
##       H1xx  SIP  R1.0 and later
##       96x1  SIP  R6.0 and later
##       96x0  SIP  R1.0 and later
## SET TLSPORT 411
##
##################  DEVICE ENROLLMENT SERVICE (DES)   ##################
##
## DES_STAT Specifies if DES discovery is to be attempted during the boot process if there is no
configuration file server provisioned on the phone.
##  Value  Operation
##    0    DES discovery is disabled and can only be restored with Reset to Defaults
##    1    DES discovery is disabled
```

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```
##    2    DES discovery is enabled (default); user prompt is displayed for end user to enforce
"DES" or not.
##    3    DES discovery is enforced without dependency on user to select "yes" on the prompt
appears after reboot.
##   This parameter is supported by:
##        J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, value 3 is supported by J100
SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
##        Avaya Vantage Devices SIP R1.1.0.0 and later; if FILE_SERVER_URL/HTTPSRVR/TLSSRVR are
received from DHCP/LLDP/UI/configuration file/AADS then DES will not be activated.
##          No support for user prompt which can enforce DES or not. Value 3 is not supported.
##          R2.1+ DES_STAT value 3 is supported. DES discovery is enforced even if
FILE_SERVER_URL, HTTPSRVR, TLSSRVR are configured from DHCP/LLDP.
## SET DES_STAT 1

###################  RTCP MONITORING  ####################
##
## The RTCP monitor
##   One RTCP monitor (VMM server) IP address in
##   dotted-decimal format or DNS name format (0 to 15
##   characters). Note that for H.323 telephones only this
##   parameter may be changed via signaling from Avaya
##   Communication Manager. For 96xx/J100 SIP models in Avaya Aura
##   environments, this parameter is set via the PPM server.
##   Note : This setting is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0,
J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later,
##    H1xx SIP R1.0 and later for non-Aura environment (For example: IP Office, etc).
## SET RTCPMON 192.168.0.10
##
## RTCPMONPORT sets the port used to send RTCP information
## to the IP address specified in the RTCPMON parameter.
## RTCPMONPORT is only supported on 96xx/J100 in non-Avaya environments.  For 96xx/J100 SIP
## models in Avaya environments, this parameter is set via the PPM server. The default value is
5005.
##   Note : This setting is supported by H1xx SIP R1.0 and later and J129  SIP  R1.0.0.0 (or
R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
##    J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later for non-Aura environment (For
example: IP Office, etc).
## SET RTCPMONPORT 5005
##
## RTCP Monitor Report Period
##   Specifies the interval for sending out RTCP monitoring
##   reports (5-30 seconds).  Default is 5 seconds. This
##   parameter applies only to 96xx/J100 SIP telephones.
##   Note : This setting is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0,
J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##   for non-Aura environment (For example: IP Office, etc).
## SET RTCPMONPERIOD 5
##
#####################  ICMP SETTINGS  ####################
##
## Destination Unreachable Message Control
##   Controls whether ICMP Destination Unreachable messages
##   are generated.
##     0 for No
##     1 for limited Port Unreachable messages
##     2 for Protocol and Port Unreachable messages
```

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```
##   Note 1: This settings is also applicable for J169/J179 H.323 R6.7 and later, J129  SIP
R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later,
##     J159 SIP R4.0.3.0 and later, H1xx SIP R1.0 and later and Avaya Vantage Devices SIP
R1.0.0.0 and later.
## SET ICMPDU 1
##
## Redirect Message control
##   Controls whether received ICMP Redirect messages will
##   be processed
##     0 for No
##     1 for Yes
##   Note 1: This settings is also applicable for J169/J179 H.323 R6.7 and later, J129  SIP
R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later,
##     J159 SIP R4.0.3.0 and later, H1xx SIP R1.0 and later and Avaya Vantage Devices SIP
R1.0.0.0 and later.
## SET ICMPRED 0
##
############  BACKUP/RESTORE SETTINGS (H.323 and SIP) ##########
##
## Backup and Restore URI
##   URI used for HTTP backup and retrieval of user data.
##   Specify HTTP server and directory path to backup file.
##   Do not specify backup file name.
##   BRURI is supported by 16xx H323 phones using http only (not https).
## Note: This parameter is supported by J169/J179 H.323 R6.7 and later, 96x1 H323 R6.2.3 and
later, J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later,
## J159 SIP R4.0.3.0 and later, H1xx SIP R1.0 and later and 96x1 SIP R7.1.0.0 and later releases
for sending a phone report to a HTTP/S file server (with the URI of the server defined by the
BRURI parameter).
## In order to send the report the phone must be registered and the administrator must access the
## phone's Admin menu and select the Send Report feature. This parameter is supported in both IPO
and Aura environment with file server that support HTTP PUT messages.
## SET BRURI http://192.168.0.28
## Note: 96x0 H.323 R3.2/96x1 H.323 R6.0 phones support in addition a format of
"http://username:password@..." or "https://username:password@..." for HTTP Basic authentication.
## The username and password are removed from the configured URI and used in the Authorization
Header. The HTTP request will be sent to the URI without the username and password fields.
## Note: Avaya Vantage Devices SIP R2.0.1.0 and later supports this parameter. When configured
the audio and debug report screens are pre-populated with the BRURI value to simplify report
##       generation by the end users.
## For example:
## SET BRURI http://Administrator:Catt*123@10.10.10.6/Backup/
## SET BRURI http://ipphone:Avaya1234@10.10.10.1
##


##
#################### AUDIBLE ALERTING ######################
##
##   Specifies the audible alerting setting for the telephone
##   and whether users may change this setting.
##
##   A value of 0 turns off audible alerting; user cannot
##     adjust ringer volume at all.
##   A value of 1 turns on audible alerting; user can adjust
```

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```
##     ringer volume but cannot turn off audible alerting.
##   A value of 2 turns off audible alerting; user can adjust
##     ringer volume and can turn off audible alerting.
##   A value of 3 turns on audible alerting; user can adjust
##     ringer volume and can turn off audible alerting.
##
##   The default value is 3.
##   SET AUDASYS 3
##
##   NOTE: This AUDASYS value is applicable for 16xx phones starting with R1.3.
##   Note: AUDASYS is not supported by J129 SIP R1.0.0.0/R1.1.0.0.  Supported by J100 SIP
R2.0.0.0 and later, J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later.
##   Note: This parameter is also supported by J169/J179 H.323 R6.7 and later.
##

############################################################
##
##            VOICE MAIL SETTINGS
##
############################################################
##
## Voice Mail Telephone Number
##   Specifies the telephone number to be dialed
##   automatically when the telephone user presses the
##   Messaging button.  The specified number is used to
##   connect to the user's Voice Mail system.
##   Note: PSTN_VM_NUM shall be used instead of MSGNUM in cases of IP Office environment, 3PCC
SIP environment or when there is failover from Aura environment to a non-Aura server.
##   Note: This parameter is ignored by 16xx phones when registered to CM version above 5.2.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later.
##       H1xx  SIP  R1.0 and later
##       96x1 H.323 R6.0 and later
##       96x1  SIP  R6.0 and later
##       96x0 H.323 R1.0 and later
##       96x0  SIP  R1.0 and later
##       16xx H.323 R1.0 and later
## Example:
## SET MSGNUM 1234

############################################################
## IPv6 related settings are applicable for 96x1 H.323 R6.0 and later, J169/J179 H.323 R6.7 and
later, 96x1 SIP R7.1.0.0 and later, J169/J179 SIP R1.5.0 and J100 SIP R2.0.0.0 and later, J159
SIP R4.0.3.0 and later.
## Avaya Vantage Devices SIP R1.0.0.0 and later support only IPV6STAT.

## DHCPSTAT
## Valid Values
##    1    run DHCPv4 only                (IPv4only-mode, if no own IPv6 address is programmed
statically), Default.
##    2    run DHCPv6 only                (IPv6only-mode, if no own IPv4 address is programmed
statically)
##    3    run both DHCPv4 & DHCPv6       (dual-stack mode)
## Description
```

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```
##    Specifies whether DHCPv4, DHCPv6, or both will be used in case IPV6STAT has enabled IPv6
support generally
## Example : Setting dual stack mode
##            SET DHCPSTAT 3
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##          For J100 SIP R4.0.0.0 and later - Specifies whether DHCPv6 is enabled or disabled.
DHCPSTAT is used in dual (IPV6STAT=1) and IPv6-only (IPV6STAT=2) modes.
##              NOTE: DHCPv4 is always enabled in IPv4 only and dual mode. DHCPv4 is disabled in
IPv6 only mode.
##             Value 1: disable DHCPv6 client, (For IPV6STAT=1 (IPv6 enabled) & "Use SLAAC=No"
(SLAAC disabled): IPv4only-mode, if no Phone(v6) IPv6 address is programmed statically
##                   For IPV6STAT=2 (IPv6 only) & "Use SLAAC=No" (SLAAC disabled): Phone(v6)
address has to be set manually).
##             Value 2,3: enable DHCPv6 client (dual-stack mode).
##              The default value is 3.
##       J169/J179 H.323 R6.7 and later
##       96x1  SIP  R7.1.0.0 and later; Value 1 as described above, Value 2/3 - run both DHCPv4 &
DHCPv6
##       96x1 H.323 R6.0 and R6.0
## SET DHCPSTAT 1
##
## PRIVACY_SLAAC_MODE
## Valid Values:
##   0 - Privacy extension disabled, one stable address is generated using modified EUI-64 format
interface identifier (Based on MAC address).
##        The phone address selection preference is based on default RFC6724 SASA rules.
##   1 - Privacy extension enabled, one stable address is generated using modified EUI-64 format
interface identifier (Based on MAC address)
##        and one temporary private address is generated. The phone address selection preference
is based on RFC6724 SASA rules.
##        PRIVACY_SLAAC_MODE changes default SASA rules (i.e. Rule 7) to prefer a manual, DHCPv6
or Stable SLAAC over SLAAC temporary private address. (default)
##   2 - Privacy extension enabled, one stable address is generated using modified EUI-64 format
interface identifier (Based on MAC address) and
##        one temporary private IPv6 address is generated. The phone address selection preference
is based on default RFC6724 SASA rules. Default
##        SASA Rule 7 is used to prefer SLAAC temporary private address over a manual, DHCPv6 or
Stable SLAAC addresses.
## Definition: Specifies the preference for Privacy Extensions(RFC3041)
## This parameter is supported by:
##   J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## Example:
## SET PRIVACY_SLAAC_MODE 2


## DUAL_IPPREF
## Valid Values
##    4    IPv4 preference (default)
##    6    IPv6 preference
## Description:
##    DUAL_IPPREF controls:
##       1. The selection of SSON either from DHCPv4 or DHCPv6 server, when phone is in dual
mode, and
##       2. Whether an IPv4 or IPv6 addresses returned by DNS would be tried first during dual-
mode operation.
```

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```
## DHCP clients use DUAL_IPPREF to decide which SSON configuration attributes to apply for
DHCPv4/DHCPv6 interworking in dual mode. Based on DUAL_IPPREF the phone selects SSON attributes
either from DHCPv4 or DHCPv6 server.
##     If DNS server name is provided, and if DNS resolver returns both IPv4 and IPv6 addresses,
the order in which they will be tried will be based on DUAL_IPPREF parameter.
## NOTE: SIP server FQDNs are resolved into addresses that are ordered based on
SIGNALING_ADDR_MODE, not DUAL_IPPREF
## Example: Setting preference to IPv4
## SET DUAL_IPPREF 6
##  This parameter is supported by:
##      J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
##
## IPV6STAT
## Valid Values
##    0    IPv6 will not be supported (IPv4 only mode).
##    1    Dual mode (IPv4 and IPv6) will be supported.
##    2    IPv6 only mode (only supported by J100 SIP 4.0.0.0 and greater)
## Description
##     Specifies whether IPv6 will be supported
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later (Default
is 0), J100 SIP R4.0.0.0.0 and later, J159 SIP R4.0.3.0 and later (default is 1).
##      Avaya Vantage Devices SIP R1.0.0.0 and later; (Default is 1); IPV6STAT shall be set to 0
as IPv6 is not supported by Avaya Vantage Device.
##       96x1 H.323 R6.0 and R6.0 (Default is 0).
##       96x1  SIP  R7.1.0.0 and later (Default is 0).
## SET IPV6STAT 1
##
## SIGNALING_ADDR_MODE
## Valid Values
##    4    IPv4 (default)
##    6    IPv6
## Description
##     This parameter is used by SIP signaling on a dual mode phone (phone with both IPv4 and
IPv6 addresses configured) to select the preferred SIP controller  IP addresses
##     from SIP_CONTROLLER_LIST_2. The phone registers to SIP controllers using IPv4 address if
SIGNALING_ADDR_MODE=4,
##     otherwise registration is over IPv6.
##     The single IPv4 mode phone ignores SIGNALING_ADDR_MODE and SIP_CONTROLLER_LIST_2 and
selects the SIP controller's IP addresses from SIP_CONTROLLER_LIST.
##     The single IPv6 mode phone ignores SIGNALING_ADDR_MODE and SIP_CONTROLLER_LIST and selects
the SIP controller's IPv6 addresses from SIP_CONTROLLER_LIST_2.
## This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## Example:
## SET SIGNALING_ADDR_MODE 4
##
## MEDIA_NEG_PREFERENCE
## Valid Values
##    0    Remote or offerer's precedence (default)
##    1    Local
##     NOTE: MEDIA_NEG_PREFERENCE is NOT used in Avaya environment. Default is remote preference.
```

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```
##           It is used by a dual mode answerer in non-Avaya environment to allow a local
preference
##           It is used in non-Avaya environment to allow a local preference.
##     NOTE: Not applicable on single mode phones.
## Description
##     MEDIA_NEG_PREFERENCE option is used by the answerer only to change the default address
##     family preference.
##     In dual IPv4/IPv6 mode, during SIP ANAT negotiation,
##     MEDIA_NEG_PREFERENCE is used to prioritize media lines in SDP.
##     By default offerer's preference is used.
##
##     MEDIA_NEG_PREFERENCE of zero means when there is a choice between IPv4 and IPv6 address,
##     the answerer honors the offerer's preference.
## This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## Example
## SET MEDIA_NEG_PREFERENCE 0
##
## MEDIA_ADDR_MODE
## Valid Values
##    4    IPv4 (default)
##    6    IPv6
##    46   Prefer IPv4 over IPv6
##    64   Prefer IPv6 over IPv4
## Description
##     MEDIA_ADDR_MODE specifies the preference of SDP media group lines [per RFC 4091, 4092 and
5888] and the SDP answer / offer format.
##     By default v4 media line is preferred.
##     MEDIA_ADDR_MODE is only used by dual stack phones which are configured with both IPv4 and
IPv6 addresses.
##     IPv4 only or IPv6 only phones ignores MEDIA_ADDR_MODE.
##     Environment            SDP Offer                                              SDP Answer
(Note2)
##     Avaya                  4 - Non ANAT IPv4 only is advertised (Note 3)          4 - IPv4 is
chosen (see Note1,3)
##                              6 - Non ANAT IPv6 only is advertised (Note 3)          6 - IPv6
is chosen (see Note1,3)
##                              46 - ANAT offer where IPv4 is preferred over IPv6      46 -
Follow the remote preference.
##                              64- ANAT offer where IPv6 is preferred over IPv4       64 -
Follow the remote preference.
##     Non-Avaya              Same as for Avaya environment                          4 - IPv4 is
chosen (see Note1,3)
##                                                                                    6 - IPv6
is chosen (see Note1,3)
##                                                                                    46 -
Prefer IPv4 (if available in SDP offer) only if MEDIA_NEG_PREFERENCE
##                                                                                   is set to
local, otherwise grants the remote preference.
##                                                                                    64 -
Prefer IPv6 (if available in SDP offer) only if MEDIA_NEG_PREFERENCE
##                                                                                   is set to
local, otherwise grants the remote preference.
##     Note1: MEDIA_ADDR_MODE=4 and 6 answerers select the MEDIA_ADDR_MODE address family in ANAT
offer.
```

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```
##            For non-ANAT offers or ANAT offers with selected "m" line (e.g. re-INVITE),
answerers reject the call
##            with 488, if MEDIA_ADDR_MODE does not match any of offered audio lines (with non-
zero port).
##     NOTE2: Answerers are always ANAT capable.
##     NOTE3: MEDIA_ADDR_MODE 4 or 6 enforces a dual stack phones which are configured with both
IPv4 and IPv6 address to behave as IPv4 only or IPv6 only phone.
##            MEDIA_NEG_PREFERENCE is ignored when MEDIA_ADDR_MODE is 4 or 6.
##  This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1  SIP  R7.1.0.0 and later
## Example : Setting to use IPv6 only
## SET MEDIA_ADDR_MODE 6
## Example : Setting to preference of IPv6 over IPv4
## SET MEDIA_ADDR_MODE 64
##
## IPV6DADXMITS specifies whether Duplicate Address Detection is performed
##    on tentative addresses, as specified in RFC 4862.
##    Non zero value specifies the maximum number of transmitted Neighbor Solicitation messages
##    to determine whether an IPv6 address is already in use.
##  Value  Operation
##    0    DAD is disabled
##    1-5  maximum number of transmitted NS messages
##         Default value is 1
## This parameter is supported by:
##       J169/J179 SIP R1.5.0; J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
##       96x1 SIP R7.1.0.0 and later
## Example:
## SET IPV6DADXMITS 1

############################################################
##
##                      SIP SETTINGS
##    Settings applicable only to 96xx/J100 telephone models
##    in non-Avaya environments
##
############################################################
##
## CALLFWDSTAT sets the call forwarding mode.
##    0   No call forwarding is permitted
##    1   Permits unconditional call forwarding
##    2   Permits call forward on busy
##    3   Permits call forward on busy and unconditional call forwarding
##    4   Permits call forward/no answer
##    5   Permits call forward/no answer and unconditional call forwarding
##    6   Permits call forward/no answer and call forward on busy
##    7   Permits call forward/no answer, call forward on busy and unconditional call forwarding
##  The default is 0.
##  Note: This parameter is supported by 96x1 SIP, J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP
R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later when failed
##        over from Aura SM to a non-Aura survivable server (excluding BSM)
##  Note: This parameter is supported by J129 SIP R1.0.0.0 and later when configured for IP
Office SIP or 3PCC SIP
```

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```
## SET CALLFWDSTAT 3
##
## 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.
##  Note: This parameter is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and
later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later for IP Office Environment.
##   The range is 0 to 20 with Default is 1.
## SET CALLFWDDELAY 5
##
## CALLFWDADDR sets the address to which calls are forwarded for the call forwarding feature.
##  The default is null ("").
##  Note the user can change or replace this administered value if CALLFWDSTAT is not 0.
##  Note: This parameter is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and
later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later for IP Office Environment.
## SET CALLFWDADDR cover@avaya.com
##


##
## ENABLE_AUTO_ANSWER_SUPPORT specifies whether the Auto Answer feature is available to users.
##  Value  Operation
##    0    Auto Answer feature is not available to users default)
##    1    Auto Answer feature is available to users
##  Note: This parameter is supported by J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET ENABLE_AUTO_ANSWER_SUPPORT 1
##
## Auto Answer Mute controls the speakerphone muting when call is auto answered by phone.
##  Value  Operation
##    0    Speakerphone is Unmuted when Auto Answered
##    1    Speakerphone is Muted when Auto Answered (default)
##  Note: This parameter is supported by J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET AUTO_ANSWER_MUTE_ENABLE 0
##
## ENABLE_DND specifies whether Do Not Disturb feature is available to users.
##  Value  Operation
##    0    Do Not Disturb feature is not available to users
##    1    Do Not Disturb feature is available to users (default)
##  This parameter is supported by:
##       J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later and J159
SIP R4.0.3.0 and later only for 3PCC environment when 3PCC_SERVER_MODE = 0 (generic) or 1
(BroadSoft)
## SET ENABLE_DND 0
##
## DND_PRIORITY_OVER_CFU_CFB defines the priority between features Do Not Disturb and Call
Forward Unconditional/Busy when both are activated by user.
##  Value  Operation
##    0    Call Forward Unconditional/Busy feature has priority over Do Not Disturb feature
(default)
##    1    Do Not Disturb feature has priority over Call Forward Unconditional/Busy feature
##  Note: This parameter is supported by J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET ENABLE_DND_PRIORITY_OVER_CFU_CFB 1
##
```

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```
## HOLD_REMINDER_TIMER specifies the number of seconds after which the phone will alert (visual
and audible) user when any call is kept on hold.
##  Valid values are 0 through 999 seconds; the default value is 0.
##  Value 0 means phone will not alert user when any call is kept on hold.
##  Note: This parameter is supported by J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later only for 3PCC environment.
## SET HOLD_REMINDER_TIMER 60
##

############################################################
##
##               96xx, J100, H1xx SIP SETTINGS
##    Settings applicable only to 96xx, J100 and H1xx Video collaboration Station
##                      running the SIP protocol
##
############################################################

## RTCP_XR specifies whether VoIP Metrics Report Block as defined in RTP Control Protocol
Extended Reports (RTCP XR)
## (RFC 3611) is sent as part of RTCP packets to remote peer or to RTCP monitoring server.
##     0 for No (Default)
##     1 for Yes
##  This parameter is supported by:
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       96x1 SIP R7.0.0 and later
## SET RTCP_XR 1
##


## MEDIAENCRYPTION specifies which media encryption (SRTP) options will be supported.
##  Up to 2 or 3 options may be specified in a comma-separated list.
##  2 options are supported by:
##     1. Prior releases to 96x1 SIP 7.0.0
##     2. H1xx SIP R1.0 and later
##     3. 96x0 SIP R1.0 to R2.6.14.1
##  3 options are supported by 96x1 SIP R7.0.0 and later, J129  SIP  R1.0.0.0 (or R1.1.0.0),
J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0
and later and H1xx SIP R1.0.1 and later.
##  For 96x0 SIP R2.6.14.5 and later, up to 3 options may be specified, but only the first two
supported options are used.
##  Options should match those specified in CM IP-codec-set form.
##     1 = aescm128-hmac80
##     2 = aescm128-hmac32
##     3 = aescm128-hmac80-unauth
##     4 = aescm128-hmac32-unauth
##     5 = aescm128-hmac80-unenc
##     6 = aescm128-hmac32-unenc
##     7 = aescm128-hmac80-unenc-unauth
##     8 = aescm128-hmac32-unenc-unauth
##     9 = none (default)
##    10 = aescm256-hmac80
##    11 = aescm256-hmac32
##  Options 10 and 11 are supported by 96x1 SIP R7.0.0 and later, H1xx SIP R1.0.1 and later and
J129 SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP
R3.0.0.0 and later,
##   J159 SIP R4.0.3.0 and later.
```

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```
##  Note: The list of media encryption (SRTP) options is ordered from high (left) to the low
(right) options. The phone will publish this list in the SDP-OFFER
##  or choose from SDP-OFFER list according to the list order defined in MEDIAENCRYPTION. Please
note that  Avaya Communication Manager has the capability
##  to change the list order in the SDP-OFFER (for audio only) when the SDP-OFFER pass through
CM.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later; supported values: 1,2,9,10 and 11. The default value is
1,2,9.
##       Avaya Vantage Connect Application SIP R1.0.0.0 and later; supported values: 1,2,9,10 and
11. The default value is 1,2,9.
##       96x1 SIP R6.0 and later
##       H1xx SIP R1.0 and later
##       96x0 SIP R1.0 and later
## SET MEDIAENCRYPTION 1,9
## SET MEDIAENCRYPTION 10,1,9
##
## ENCRYPT_SRTCP specifies whether RTCP packets are encrypted or not. SRTCP is only used if SRTP
is enabled using
## MEDIAENCRYTION (values other than 9 (none) are configured).
## This parameter controls RTCP encryption for RTCP packets exchanged between peers.
## RTCP packets sent to Voice Monitoring Tools are always sent unencrypted.
##  Value  Operation
##   0     SRTCP is disabled (default).
##   1     SRTCP is enabled.
##  This parameter is supported by:
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Equinox 3.1.2 and later
##       96x1 SIP R7.1.0.0 and later
##       Avaya Vantage Connect Application SIP R1.0.0.0 and later
## SET ENCRYPT_SRTCP 1


####################  IP OFFICE SETTINGS  ####################
##
## ENABLE_IPOFFICE specifies whether the deployment environment is IP Office
##  Value  Operation
##   0     Not IP Office environment (except failover mode to IP Office in Avaya Aura
environment) (Default)
##   1     IP Office environment; Native support of IP Office with a limited feature set.
##   2     IP Office environment; Additional features driven by the IP Office SIP proxy.
##  This parameter is supported by:
##       Avaya Equinox 3.4 and later
##       J100 SIP R2.0.0.0 and later (J129 supports values 0-1 and J169/J179 support values 0 and
2). J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later supports values 0 and 2.
##           When ENABLE_IPOFFICE is set to 2, some of the 46xxsettings.txt file parameters have
no effect.
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later (values 0-1)
##       Avaya Vantage Devices SIP R1.1.0.1 and later (values 0-1)
##       J169/J179 SIP R1.5.0 (values 0-1)
##       J129 SIP R1.0.0.0 (or R1.1.0.0) (values 0-1)
##       H1xx SIP R1.0.2 and later (values 0-1)
## SET ENABLE_IPOFFICE 1
```

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```
##
## MEDIA_PRESERVATION specifies whether a call will be preserved when there is no SIP
connectivity to IP Office.
##  This parameter is only applicable when ENABLE_IPOFFICE is set to 2.
##  Value  Operation
##    0    Phone will not preserve a call. As soon as the phone detects SIP connectivity failure
to IP Office, phone will drop a call and make re-registration attempt.
##    1    Phone will try to preserve a call for a duration specified by PRESERVED_CALL_DURATION
settings parameter (Default).
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET MEDIA_PRESERVATION 0
##
## PRESERVED_CALL_DURATION specifies how long the call will be preserved if ENABLE_IPOFFICE is
set to 2 and if MEDIA_PRESERVATION is set to 1.
##  In such case, the call will be preserved for a duration of PRESERVED_CALL_DURATION minutes.
Valid values are 10-120. Default value is 120 minutes.
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later; J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET PRESERVED_CALL_DURATION 10
##
## SUBSCRIBE_LIST_NON_AVAYA specifies comma separated list of event packages to subscribe to
after registration.
##  Possible values are: "reg", "dialog", "mwi", "ccs", "message-summary" which is identical to
"mwi", "avaya-ccs-profile" which is identical to "ccs"
##  The values are case insensitive.
##  For IPO the recommended value shall be "reg,message-summary,avaya-ccs-profile".
##  For 3PCC environment the value "message-summary" may be required.
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##       Avaya Vantage Connect Application SIP R1.1.0.1 and later
##       H1xx SIP R1.0.2 and later
## SET SUBSCRIBE_LIST_NON_AVAYA "reg,message-summary,avaya-ccs-profile"
##
## USER_STORE_URI for User Data
##   URI used for HTTP/S backup and retrieval of user data.
##   Specify HTTP/S server and directory path to backup file.
##   Do not specify backup file name.
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R1.1.0.1 and later up to R2.1.0.0 (excluded)
##       J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET USER_STORE_URI https://192.168.0.28
## Note: This parameter is supported by Avaya Vantage Devices SIP R1.1.0.1 and later up to
R2.1.0.0 (excluded) to define user store URI for personal/enterprise contacts. Used in IP Office
environment only.
##       The default ports are 80 for HTTP and 443 for HTTPS. When "Use Preferred Port" is
enabled in IP Office 11.0 and later then the ports are changed to 8411 for HTTP and 411 for
HTTPS.
##       IPO contacts will be supported in R2.0.1.0 and on, by the Avaya Breeze Client SDK
application and not by the Avaya Vantage device.
## SET USER_STORE_URI https://192.168.0.28:411
## SET USER_STORE_URI http://192.168.0.28:8411
## Note: This parameter is supported by J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later for 3PCC Environment and IP Office R10.1 and
later.
##
```

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```
## ENABLE_OOD_RESET_NOTIFY specifies whether the phone supports out of dialog (OOD) SIP NOTIFY
message with
##  Event:resync or Event:check-sync only. The events are used to remotely restart the phone
(once all calls end).
##  Value  Operation
##   0     OOD is not supported (Default)
##   1     OOD is supported
##  This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J129, J169, J179), J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
## SET ENABLE_OOD_RESET_NOTIFY  1
##


####################  3PCC ENVIRONMENT SETTINGS  ####################
##
## ENABLE_3PCC_ENVIRONMENT specifies whether the deployment environment is third party SIP Server
##  Value  Operation
##   0     Not 3PCC environment
##   1     3PCC environment (Default)
##  Note: This parameter should be set to '0' for Aura environment and IP Office
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later; the default is 0.
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later; the default is 0.
##       J129  SIP  R1.1.0.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
## SET ENABLE_3PCC_ENVIRONMENT 0
##
## 3PCC_SERVER_MODE specifies if the phone expects a generic 3PCC server or a BroadSoft server
(applicable when ENABLE_3PCC_ENVIRONMENT is set to 1).
##  Value  Operation
##    0    Generic (Default)
##    1    BroadSoft
##    2    3CX
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later; value 2 is supported in R4.0.2.0 and later, J159 SIP
R4.0.3.0 and later
##       Avaya Vantage Devices SIP R2.0.1.0 and later, value 1 only is supported.
##       Avaya Vantage Connect Application SIP R2.0.1.0 and later, value 1 only is supported.
## SET 3PCC_SERVER_MODE 1
##
######################## BROADSOFT CALL FEATURES SETTINGS ##################
##
## The parameters below are applicable when 3PCC_SERVER_MODE=1 and ENABLE_3PCC_ENVIRONMENT=1.
##
## BLF_LIST_URI specifies the BroadSoft Busy Lamp Field (BLF) Resource URI. It defines the unique
name for the list of users to be monitored.
##   BLF_LIST_URI must be configured to enable BLF feature unless Xtended Services Interface
(XSI) is enabled. When XSI_URL parameter is defined, the phone retrieves BLF configuration from
XSP server and
##   ignores BLF_LIST_URI value. The default value is "".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET BLF_LIST_URI sip:mylist1@as.iop1.broadworks.net
## SET BLF_LIST_URI mylist1@as.iop1.broadworks.net
## SET BLF_LIST_URI sip:mylist1
```

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```
## SET BLF_LIST_URI mylist1
##
## CALL_PICKUP_FAC specifies the Directed Call Pickup feature access code that phone will use to
pick up the call for a Busy Lamp Field (BLF) monitored station.
##   In a BroadSoft environment if Xtended Services Interface (XSI) is enabled and XSI_URL
parameter is defined, the phone retrieves and ignores the CALL_PICKUP_FAC value defined in this
##   settings file.
##   The default is "*97".
##   To disable Call Pickup define a NULL/Empty value surrounded by quotes. SET CALL_PICKUP_FAC
""
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET CALL_PICKUP_FAC "*98"
##
## CALL_PICKUP_BARGEIN_FAC specifies the Directed Call Pickup with Barge-in feature access code
that phone will use to barge into a call between a remote user
##   and Busy Lamp Field (BLF) monitored station. In a BroadSoft environment if Xtended Services
Interface (XSI) is enabled and XSI_URL parameter is defined,
##   the phone retrieves and ignores CALL_PICKUP_BARGEIN_FAC value defined in this settings file.
##   The default value is "*33".
##   To disable Directed Call Pickup define a NULL/Empty value surrounded by quotes. SET
CALL_PICKUP_BARGIN_FAC ""
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET CALL_PICKUP_BARGEIN_FAC *32
##
## CALL_UNPARK_FAC specifies the Call Unpark feature access code that phone will use to retrieve
a call that has been parked to a Busy Lamp Field (BLF) monitored station.
##   In a BroadSoft environment if Xtended Services Interface (XSI) is enabled and XSI_URL
parameter is defined, the phone retrieves and ignores CALL_UNPARK_FAC value.
##   To disable Call Unpark define a NULL/Empty value surrounded by quotes. SET CALL_UNPARK_FAC
""
##  The default is "*88".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET CALL_UNPARK_FAC *89
##
## CALL_PARK_FAC specifies FAC that is used to make a park operation via calling of
"CALL_PARK_FAC + Destination" number. This parameter is required when the shortcut action for
"Active call shortcut keys" feature is set to Park(4).
##  Refer to SHORTCUT_ACTION_BLF, SHORTCUT_ACTION_CONTACT and SHORTCUT_ACTION_AUTODIAL,
SHORTCUT_ACTION_BLF_PARK
##  Valid values are 1 to 4 dialable characters (0-9, * and #); the default value is "".
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET CALL_PARK_FAC "#95"
##
## ALLOW_BLF_LIST_CHANGE specifies the user permissions for adding/removing BroadSoft Busy Lamp
Field (BLF) monitored users from the phone.
##  Value  Operation
##    0    User is not allowed to add/delete BLF monitored users
##    1    User is allowed only to delete BLF monitored users
##    2    User is allowed only to add BLF monitored users
##    3    User is allowed to add and delete BLF monitored users (default)
```

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```
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET ALLOW_BLF_LIST_CHANGE 2
##
## SHORTCUT_ACTION_BLF specifies the action to perform if the user has an active call and a free
call appearance and the user presses a BLF button on the phone screen,
##  the setting only applies to 3PCC server environments.
##  Value  Operation
##    0    Hold active call and dial destination (default)
##    1    Transfer active call to destination (behavior based on ENABLE_BLIND_TRANSFER)
##    2    Blind transfer active call to destination
##    3    Conference Pending state to destination
##    4    Park – hold active call and dial Park FAC+Destination (currently option "4" is only
supported in a BroadSoft environment)
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SHORTCUT_ACTION_BLF 2
##
## SHORTCUT_ACTION_CONTACT specifies the action to perform if the user has an active call and a
free call appearance and the user presses a Contact button on the phone screen.
##  Value  Operation
##    0    Hold active call and dial destination (default)
##    1    Transfer active call to destination (behavior based on ENABLE_BLIND_TRANSFER)
##    2    Blind transfer active call to destination
##    3    Conference Pending state to destination
##    4    Park – hold active call and dial Park FAC+Destination (currently option "4" is only
supported in a BroadSoft environment)
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SHORTCUT_ACTION_CONTACT 2
##
## SHORTCUT_ACTION_AUTODIAL specifies the action to perform if the user has an active call and a
free call appearance and the user presses an Autodial button on the phone screen.
##  Value  Operation
##    0    Hold active call and dial destination (default)
##    1    Transfer active call to destination (behavior based on ENABLE_BLIND_TRANSFER)
##    2    Blind transfer active call to destination
##    3    Conference Pending state to destination
##    4    Park – hold active call and dial Park FAC+Destination (currently option "4" is only
supported in a BroadSoft environment)
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SHORTCUT_ACTION_AUTODIAL 2
##
## SHORTCUT_ACTION_BLF_PARK specifies the action which should be performed when BLF Park line is
activated during an active call.
##  Value  Operation
##    0    Blind transfer to BLF Park destination (default)
##    1    Dial Park FAC + BLF Park destination
##  This parameter is supported by:
##       J100 SIP R4.0.2.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
```

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```
##  Note: For the "Dial Park FAC + BLF Park destination" option please verify that the "Call Park
FAC" is defined correctly.
## SET SHORTCUT_ACTION_BLF_PARK 1
##
###################### BROADSOFT XTENDED SERVICES INTERFACE (XSI) SETTINGS #################
##
## The parameters below are applicable when 3PCC_SERVER_MODE=1 and ENABLE_3PCC_ENVIRONMENT=1.
##
## XSI_URL specifies BroadWorks Xtended Service Platform (XSP) server FQDN/IP address, HTTP or
HTTPS mode and port.  If port is not defined, 80 is used for HTTP and 443 for HTTPS by default.
##  The default is "".
##  This is the main parameter to make features work in BroadSoft environment.
##  If value of this parameter is non-empty, phone will initiate Xtended Services Interface (XSI)
connections establishment to retrieve feature list.
##  Note: If XSI_URL is defined the following local call features will not be available. They
must be enabled for the user on the BroadSoft server:
##   - Do Not Disturb (ENABLE_DND, ENABLE_DND_PRIORITY_OVER_CFU_CFB)
##   - Call Forward CFA, CFB, CFNA
##   - Auto Answer (ENABLE_AUTO_ANSWER_SUPPORT, AUTO_ANSWER_MUTE_ENABLE)
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET XSI_URL http://xsp1.iop2.broadworks.net
## SET XSI_URL https://xsp1.iop2.broadworks.net:443
## SET XSI_URL http://192.168.111.111:8080
## SET XSI_URL https://192.168.111.111
##
## XSI_CHANNEL_DURATION defines the time duration in minutes for Xtended Services Interface (XSI)
event channel, i.e. phone will ask BroadWorks Xtended Service Platform (XSP) server
##  to maintain the established Comet HTTP connection for the specified period of time. After 50%
of this time phone will reestablish Comet HTTP connection.
##  The values range is 60-1440. The default is 60.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET XSI_CHANNEL_DURATION 100
##
## XSI_HEARTBEAT defines the interval in seconds to send heartbeat messages over Comet HTTP
connection to BroadWorks Xtended Service Platform (XSP) server.
##  Ideally XSI_HEARTBEAT should be configured to BroadSoft’s eventTimeout/2. The eventTimeout
value is configurable on
##  BroadSoft, by default it is 30 seconds.
##  The values range is 1-999. The default value is 15.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET XSI_HEARTBEAT 20
##
## FORCE_XSI_USER_ID specifies the BroadSoft's User Id which phone should use for Xtended
Services Interface (XSI) authentication (SIP or Web methods).
##  Note: It shall be User Id without @ and domain. The default is "".
##  It is not recommended to configure BroadSoft's User Id in the 46xxsettings.txt file
##  if HTTPS protocol is not used and if there is no HTTP authentication or mutual certificate
based authentication else the 46xxsettings.txt
##  file will be retrieved by anyone.
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
```

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```
## SET FORCE_XSI_USER_ID userhandle
##
## FORCE_XSI_WEB_PASSWORD specifies the BroadSoft's Web portal password which phone should use
for Xtended Services Interface (XSI) authentication (Web method).
##  If empty, it means that SIP authentication method should be used. The default is "".
##  It is not recommended to configure BroadSoft's Web portal password in the 46xxsettings.txt
file
##  if HTTPS protocol is not used and if there is no HTTP authentication or mutual certificate
based authentication else the 46xxsettings.txt
##  file will be retrieved by anyone.
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET FORCE_XSI_WEB_PASSWORD userpassword
##
###################### BROADSOFT SHARED CALL APPEARANCE SETTINGS ##################
##
## The parameters below are applicable when 3PCC_SERVER_MODE=1 and ENABLE_3PCC_ENVIRONMENT=1.
##
## PRIMARY_LINE_TYPE specifies if the phones primary line is a private or shared line. The
primary line is the one associated with the user's login credentials.
##  Value  Operation
##    0    Primary line is a private line (default)
##    1    Primary line is a shared line
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET PRIMARY_LINE_TYPE 1
##
## PRIMARY_LINE_BARGE_IN_ENABLED specifies whether a primary line which is shared is configured
in the BroadSoft server to either enable or disable the ability
##  of a user to barge into a call at a different location on the shared line.
##  This setting is ignored if PRIMARY_LINE_TYPE = 0
##  Value  Operation
##    0    Barge in is disabled for the shared line
##    1    Barge in is enabled for the shared line (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET PRIMARY_LINE_BARGE_IN_ENABLED 0
##
## SCA1_ENABLED, SCA2_ENABLED, SCA3_ENABLED specifies if first, second or third Shared Call
Appearance (SCA) is/are enabled.
##  For example, SCA1_ENABLED defines if the first shared line is enabled.
##  Value  Operation
##    0    The shared line is disabled (default)
##    1    The shared line is enabled
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_ENABLED 1
## SET SCA2_ENABLED 1
## SET SCA3_ENABLED 1
##
## SCA1_MAX_CALL_APPEARANCES, SCA2_MAX_CALL_APPEARANCES, SCA3_MAX_CALL_APPEARANCES specify the
maximum number of simultaneous calls on the first, second and third shared line
```

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```
##  respectively. Value range 1-5. Default value is 3.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_MAX_CALL_APPEARANCES 4
## SET SCA2_MAX_CALL_APPEARANCES 5
## SET SCA3_MAX_CALL_APPEARANCES 2
##
## SCA1_SIPUSERID, SCA2_SIPUSERID, SCA3_SIPUSERID specify the AOR(Address of Record) for first,
second and third shared line respectively.
##  It should only specify the handle (e.g. 123456_2) since the domain is specified
independently.
##  Note that shared lines are only supported for the same SIP domain as the primary line.
##  The default value is "".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_SIPUSERID 6459137
## SET SCA2_SIPUSERID 6459138
## SET SCA3_SIPUSERID 6459139
##
## SCA1_USERNAME, SCA2_USERNAME, SCA3_USERNAME specify the username to be used for authentication
when challenged for credentials on SIP requests associated with
##  the first, second and third shared line respectively.
##  This value is optional and if not specified the SCA1_SIPUSERID, SCA2_SIPUSERID,
SCA3_SIPUSERID would be used for authentication.
## The default value is "".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_USERNAME 6459137
## SET SCA2_USERNAME 6459138
## SET SCA3_USERNAME 6459139
##
## SCA1_PASSWORD, SCA2_PASSWORD, SCA3_PASSWORD specify the password to be used for authentication
when challenged for credentials on SIP requests associated
##  with the first, second and third shared line respectively. The default is "".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_PASSWORD sca1pass
## SET SCA2_PASSWORD sca1pass
## SET SCA3_PASSWORD sca1pass
##
## SCA1_EXTENSION, SCA2_EXTENSION, SCA3_EXTENSION specify the display name for the first, second
and third shared line respectively.
##  If not specified the SCA1_SIPUSERID, SCA2_SIPUSERID, SCA3_SIPUSERID would be used. The
default is "".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_EXTENSION UsernameA
## SET SCA2_EXTENSION UsernameB
## SET SCA3_EXTENSION UsernameC
##
## SCA1_BARGE_IN_ENABLED, SCA2_BARGE_IN_ENABLED, SCA3_BARGE_IN_ENABLED specify for each shared
line in the BroadSoft server to either enable or disable the
```

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```
##  ability of a user to barge into a call at a different location on the first, second, third
shared line respectively.
##  Value  Operation
##    0    Barge in is disabled for the shared line
##    1    Barge in is enabled for the shared line (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA1_BARGE_IN_ENABLED 0
## SET SCA2_BARGE_IN_ENABLED 0
## SET SCA3_BARGE_IN_ENABLED 0
##
## SCA_LINE_SEIZE_DURATION specifies the length of time in seconds to be used for a line-seize
subscription on any (first, second, third) shared line or on a primary shared line.
##  Value range is 5-40. The default is 15.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET SCA_LINE_SEIZE_DURATION 0
##
## PROVIDE_SHARED_LINE_CONFIG specifies if the user has the ability to change Shared Line
configuration using the Settings menu on the phone.
##  Value  Operation
##    0    Shared lines is not displayed in settings menu
##    1    Shared lines is displayed in settings menu but all information is read-only
##    2    Shared lines is displayed in settings menu and is fully configurable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (J129 and J139 do not support
this parameter)
## SET PROVIDE_SHARED_LINE_CONFIG 0
##
###################### BROADWORKS DIRECTORY SETTINGS ##################
##
## The parameters below are applicable when 3PCC_SERVER_MODE=1 and ENABLE_3PCC_ENVIRONMENT=1.
##
## BW_ENABLE_DIR specifies BroadWorks Directory feature availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR 0
##
## BW_ENABLE_DIR_ENTERPRISE specifies BroadWorks Enterprise directory availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_ENTERPRISE 0
##
## BW_ENABLE_DIR_ENTERPRISE_COMMON specifies BroadWorks Enterprise Common directory availability
state.
##  Value  Operation
##    0    disable
```

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```
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_ENTERPRISE_COMMON 0
##
## BW_ENABLE_DIR_GROUP specifies BroadWorks Group directory availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_GROUP 0
##
## BW_ENABLE_DIR_GROUP_COMMON specifies BroadWorks Group Common directory availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_GROUP_COMMON 0
##
## BW_ENABLE_DIR_PERSONAL specifies BroadWorks Personal directory availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_PERSONAL 0
##
## BW_ENABLE_DIR_CUSTOM specifies BroadWorks Custom directory availability state.
##  Value  Operation
##    0    disable
##    1    enable (default)
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_ENABLE_DIR_CUSTOM 0
##
## BW_DIR_ENTERPRISE_DESCRIPTION specifies the display name for BroadWorks Enterprise directory.
##  The default is "Enterprise".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_ENTERPRISE_DESCRIPTION "LargeEnterprise"
##
## BW_DIR_ENTERPRISE_COMMON_DESCRIPTION specifies the display name for BroadWorks Enterprise
Common directory.
##  The default is "Enterprise Common".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_ENTERPRISE_COMMON_DESCRIPTION "DirectoryName"
##
## BW_DIR_GROUP_DESCRIPTION specifies the display name for BroadWorks Group directory.
##  The default is "Group".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
```

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```
## SET BW_DIR_GROUP_DESCRIPTION "Sales"
##
## BW_DIR_GROUP_COMMON_DESCRIPTION specifies the display name for BroadWorks Group Common
directory.
##  The default is "Group Common".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_GROUP_COMMON_DESCRIPTION "SalesTeam"
##
## BW_DIR_PERSONAL_DESCRIPTION specifies the display name for BroadWorks Personal directory.
##  The default is "Personal".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_PERSONAL_DESCRIPTION "PersonalList"
##
## BW_DIR_CUSTOM_DESCRIPTION specifies the display name for BroadWorks Custom directory.
##  The default is "Custom".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_CUSTOM_DESCRIPTION "CustomDescription"
##
## BW_DIR_ENTERPRISE_EXTENSION specifies the display name for BroadWorks Enterprise directory
extension.
##  The default is "BWEntr".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_ENTERPRISE_EXTENSION "BWEntrA"
##
## BW_DIR_ENTERPRISE_COMMON_EXTENSION specifies the display name for BroadWorks Enterprise Common
directory extension.
##  The default is "BW EnCom".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_ENTERPRISE_COMMON_EXTENSION "BW EnCom1"
##
## BW_DIR_GROUP_EXTENSION specifies the display name for BroadWorks Group directory extension.
##  The default is "BW Group".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_GROUP_EXTENSION "BW GroupA"
##
## BW_DIR_GROUP_COMMON_EXTENSION specifies the display name for BroadWorks Group Common directory
extension.
##  The default is "BW GrCom".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_GROUP_COMMON_EXTENSION "BW GrComA"
##
## BW_DIR_PERSONAL_EXTENSION specifies the display name for BroadWorks Personal directory
extension.
##  The default is "BW Pers".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_PERSONAL_EXTENSION "BW PersA"
##
```

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```
## BW_DIR_CUSTOM_EXTENSION specifies the display name for BroadWorks Custom directory extension.
##  The default is "BW Cust".
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later, J159 SIP R4.0.3.0 and later (Not supported by J129)
## SET BW_DIR_CUSTOM_EXTENSION "BW CustA"
##
####################  OPENSIP NON BROADSOFT SETTINGS  ####################
##
## DOWNLOADABLE_DIRECTORY specifies the name of the downloadable directory file (XML) that has
directory contacts. This file is expected to be present at the same location where
46xxsettings.txt is available on the file server.
##    This feature is applicable to OpenSIP environment excluding BroadSoft.
##    The default is "".
##  This parameter is supported by:
##       J100 SIP R4.0.1.0 and later (J139/J169/J179 only), J159 SIP R4.0.3.0 and later
## SET DOWNLOADABLE_DIRECTORY directory.xml
##

####################  DISPLAY SETTINGS  ####################
##


 ## BACKGROUND_IMAGE specifies a list of background images. The default value is "".
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J100 SIP R4.0.3.0 and later (J139 and
J159)
##                        Up to 5 background images are supported. Only jpeg/jpg files are
supported.
##                          The maximum size of any jpeg file is 256 KB. The filenames are case
insensitive.
##                          J139/J159/J169/J179 screen resolution is 320 pixels x 240 pixels. J179
color depth is 16 bits.
##                          J129 screen resolution is 128 pixels x 32 pixels.
##                        The files shall be stored in the same directory defined by HTTPDIR /
TLSDIR.
## SET BACKGROUND_IMAGE "background_example1.jpg,background_example2.jpeg"
##
## BACKGROUND_IMAGE_DISPLAY specifies the administrator choice of background image.
## The filename shall be one of the filenames listed in BACKGROUND_IMAGE.
## If BACKGROUND_IMAGE_SELECTABLE is set to 1 then the end user may override this setting.
## The default value is "".
## To select one of the six built-in default images, specify a value from 0 to 5 for this setting
where 0 corresponds to "Default image 1" and 5 corresponds to "Default image 6".
## This parameter is supported by:
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J100 SIP R4.0.3.0 and later (J139 and
J159)
## SET BACKGROUND_IMAGE_DISPLAY background_example1.jpg
##
## BACKGROUND_IMAGE_SELECTABLE specifies whether end users are allowed to choose background
images
## (and overrides administrator choice as configured using BACKGROUND_IMAGE_DISPLAY parameter).
##  Value      Operation
##    0        End user is not allowed to choose background image and will not see the background
image selection in the Settings -> Display menu.
##    1        End user is allowed to choose the background image from the Settings -> Display
menu (Default)
## This parameter is supported by:
```

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```
##       J100 SIP R2.0.0.0 and later (J169 and J179 only), J100 SIP R4.0.3.0 and later (J139 and
J159)
## SET BACKGROUND_IMAGE_SELECTABLE 0

## Options Menu Display
##   Determines whether Options & Settings menu is displayed
##   on phone.
##     0 for No
##     1 for Yes
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
## SET PROVIDE_OPTIONS_SCREEN 1
##
## Network Info Menu Display
##   Determines whether Network Information menu is displayed
##   on phone.
##     0 for No
##     1 for Yes
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
## SET PROVIDE_NETWORKINFO_SCREEN 1
##
## Logout Enabled
##   Determines whether user can log out from phone.
##     0 for No
##     1 for Yes
##  This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later; Determines whether  log out  option is available or not in
Avaya Menu options.
## SET PROVIDE_LOGOUT 1
##
## DISPLAY_SSL_VERSION - display version of OpenSSH/OpenSSL
##  Value  Operation
##  0      No display of OpenSSH/OpenSSL version (default)
##  1       Display of OpenSSH/OpenSSL version
##  This parameter is supported by:
##      J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later,
J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      96x1 SIP R7.1.0.0 and later
## SET DISPLAY_SSL_VERSION 1

####################  CALL LOG SETTINGS  ####################
##
## Call Log Enabled
##   Determines whether call logging and associated menus
##   are available on the phone.
##     0 for No
##     1 for Yes
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later and Avaya Vantage Connect
Application SIP R1.0.0.1 and later.
## SET ENABLE_CALL_LOG 1
##
```

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```
## Redial Enabled
##   Determines whether redial softkey is available.
##     0 for No
##     1 for Yes
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later and
## Avaya Vantage Connect Application SIP R1.0.0.0 and later.
## SET ENABLE_REDIAL 1
##
## Redial List Enabled
##   Determines whether phone redials last number or
##   displays list of recently dialed numbers.
##     0 for last number redial
##     1 user can select between last number redial and
##       redial list (default). By default, user choice is "last number redial".
## Note: This parameter is also supported by J100 SIP R2.0.0.0 and later (J169/J179 only) and
J159 SIP R4.0.3.0 and later. ENALBE_REDIAL_LIST is enforced to 1 when ENABLE_CALL_LOG is set to
1.
## SET ENABLE_REDIAL_LIST 1
##

####################  CONTACTS SETTINGS  ###################
##
## Contacts Enabled
##   Determines whether the contacts application and
##   associated menus are available on the phone.
##     0 for No
##     1 for Yes
## Note: This parameter is also supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later and Avaya Vantage Connect
Application SIP R1.0.0.1 and later.
## Note: This parameter is also supported by H1xx R1.0.1 SIP and later, but it controls only
## the "Contacts" virtual button LED whether it is dimmed and pressing on it has no effect
(ENABLE_CONTACTS==0) or
## whether "Contacts" virtual button LED is ON and pressing on it has effect (ENABLE_CONTACTS==1,
default).
## SET ENABLE_CONTACTS 1

## CONTACT_NAME_FORMAT specifies how contact names are displayed.
##  Value  Operation
##   0     "Last Name, First Name" (Default)
##   1     "First Name Last Name"
##  This parameter is supported by:
##       J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later, J159 SIP R4.0.3.0 and later
## SET CONTACT_NAME_FORMAT 0

####################  LANGUAGE SETTINGS  ####################
##
## System-Wide 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 shall use
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```
##   its built-in English text strings. 0 to 32 ASCII
##   characters.  Filename must end in .xml
##
## NOTE:
##   For 96xx SIP Release 1.0 phones only, all language
##   filenames begin with Mls_Spark_. For example,
##   Mls_Spark_English.xml
##
##   For 96xx SIP Release 2.0 and later,
##   all language filenames begin with Mlf_
##
## SET SYSTEM_LANGUAGE Mlf_English.xml
##
##  The language files of 96x0 SIP 2.6.13 and later in the 96x0 SIP firmware distributions are
different than 96x1
##  and therefore their filenames were changed to Mlf_S96x0_<Language name>.xml.
##  Mlf_<language name>.xml filename convention is used by:
##  1. 96x1 SIP Release 6.0 and later and
##  2. 96xx SIP Release 2.0 up to 2.6.13 (excluded).
##  In mutual environment of 96x0 SIP and 96x1 SIP phones there shall be use of IF conditional
statement
##  base on MODEL/GROUP to assign different language files for each phone family.
## Note: SYSTEM_LANGUAGE is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0,
J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later as 96x1
SIP phones.
## SET SYSTEM_LANGUAGE Mlf_English.xml
## SET SYSTEM_LANGUAGE Mlf_S96x0_English.xml
##
## Installed Languages
##   Specifies the language files to be installed/downloaded
##   to the phone. Filenames may be full URL, relative
##   pathname, or filename. (0 to 1096 ASCII characters,
##   including commas). Filenames must end in .xml.
##
## NOTE:
##   For 96xx SIP Release 1.0 phones only, all language
##   filenames begin with Mls_Spark_  For example,
##   Mls_Spark_English.xml
##
##   For 96xx SIP Release 2.0 and later,
##   all language filenames begin with Mlf_
##
##  The language files of 96x0 SIP 2.6.13 and later in the 96x0 SIP firmware distributions are
different than 96x1
##  and therefore their filenames were changed to Mlf_S96x0_<Language name>.xml.
##  Mlf_<language name>.xml filename convention is used by:
##  1. 96x1 SIP Release 6.0 and later and
##  2. 96xx SIP Release 2.0 up to 2.6.13 (excluded).
##  In mutual environment of 96x0 SIP and 96x1 SIP phones there shall be use of IF conditional
statement
##  base on MODEL/GROUP to assign different language files for each phone family.
## Note: LANGUAGES is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100
SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later as 96x1 SIP
phones.
##
```

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```
## SET LANGUAGES Mlf_German.xml,Mlf_ParisianFrench.xml,Mlf_LatinAmericanSpanish.xml
## SET LANGUAGES
Mlf_S96x0_German.xml,Mlf_S96x0_ParisianFrench.xml,Mlf_S96x0_LatinAmericanSpanish.xml
##

####################  COUNTRY AND DATE SETTINGS  ####################
##
## Call Progress Tone Country
##   Country used for network call progress tones.
##   For Argentina use keyword      "Argentina"
##   For Australia use keyword      "Australia"
##   For Brazil use keyword         "Brazil"
##   For Canada use keyword         "USA"
##   For France use keyword         "France"
##   For Germany use keyword        "Germany"
##   For Italy use keyword          "Italy"
##   For Ireland use keyword        "Ireland"
##   For Mexico use keyword         "Mexico"
##   For Spain use keyword          "Spain"
##   For United Kingdom use keyword "UK"
##   For United States use keyword  "USA"
##
## NOTE 1: For a complete list of supported countries, see your telephone's Administrators Guide.
## Note 2: Country names with spaces shall be enclosed in double quotes, as in:
##           SET COUNTRY "Saudi Arabia"
## NOTE 3: This setting is also applicable for J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP
R1.5.0, J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
For J100 SIP R2.0.0.0 (J129, J159 and J179 only) please use
##         WLAN_COUNTRY for Wi-Fi Regulatory Domain configuration.
## Note 4: This parameter is supported by H1xx SIP R1.0 and later. For H1xx this parameter is
used for country configuration for the following:
##           a. Call Progress Tones, b. cordless handset, c. Wi-Fi and d. default anti-flickering
("50" or "60" Hz).
##         This parameter MUST be configured for cordless handset operation (only certain
countries
##         are supported with cordless handset. Refer to Administrator guide for the full list).
##         The default of this parameter is "Undefined" which means:
##           a. Call progress Tones for "USA", b. Cordless handset is disabled, c. Wi-Fi is
configured as WorldWide and d. "60 Hz" anti-flickering is used.
## Note 5: This parameter is supported by Avaya Vantage Devices SIP R1.0.0.0 and later. For Avaya
Vantage Devices this parameter is used for
##         country configuration for Wi-Fi.  The default of this parameter is "USA".
##
## SET COUNTRY USA

## Time Format
##   Specifies the format of the time displayed in the phone.
##     0 for am/pm format (Default)
##     1 for 24h format
##   The TIMEFORMAT configured in the settings file is used in the following cases:
##    1. The phone is not registered to Avaya Aura Session Manager
##    2. If the phone is registered to Avaya Aura Session Manager AND
##       A. there is no information stored for the timeformat for this specific user (first time
login of this user)
##          AND
```

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```
##       B. there is no support of "Profile Settings" in the "Endpoint Template" (which is
supported by SMGR 6.3.8 and up).
## This parameter is supported by:
##     J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later,
J159 SIP R4.0.3.0 and later - The TIMEFORMAT can also be set from the phone's local menu
##      independent of the environment (3PCC, Aura, IP Office). The TIMEFORMAT will be used in
the above cases until user manually changes the value.
##     96x1 SIP R6.0 and later
##     H1xx SIP R1.0 and later
##     Avaya Vantage Devices SIP R1.0.0.0 and later (up to R1.0.0.0 build 2304).
## SET TIMEFORMAT 1

## Daylight Savings Time Mode
##   Specifies daylight savings time setting for phone.
##     0 for no daylight saving time
##     1 for daylight savings activated (time set to DSTOFFSET)
##     2 for automatic daylight savings adjustment (as
##       specified by DSTSTART and DSTSTOP)
## Note: This parameter is supported by H1xx SIP R1.0 only (TIMEZONE shall be used in R1.0.0.1
and later).
## Note: This parameter is also supported by J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
## SET DAYLIGHT_SAVING_SETTING_MODE 2

#####################  PORT SETTINGS (SIP ONLY)  #####################
##
## UDP Minimum Port Value
##   Specifies the lower limit of the UDP port range
##   to be used by RTP/RTCP or SRTP/SRTCP connections.
##   (1024 -65503).
##   Note : This setting is also applicable for J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP
R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later,
##   Avaya Vantage Connect Application SIP R1.0.0.0 and later and Avaya Equinox 3.1.2 and later.
##   Note: For H1xx SIP R1.0 and later the first half of the range is used for audio
##         and the second half for video.
## SET RTP_PORT_LOW 5004
##
## UDP Port Range
##   Specifies the range or number of UDP ports
##   available for RTP/RTCP or SRTP/SRTCP connections.
##   This value is added to RTP_PORT_LOW to determine
##   the upper limit of the UDP port range (32-64511).
##   Note : This setting is also applicable for J129 SIP R1.0.0.0 (or R1.1.0.0), J100 SIP
R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later,
##   Avaya Vantage Connect Application SIP R1.0.0.0 and later and Avaya Equinox 3.1.2 and later.
##   Note: For H1xx SIP R1.0 and later the first half of the range is used for audio
##         and the second half for video.
## SET RTP_PORT_RANGE 40
##
## Signaling Port Minimum Value
##   Specifies the minimum port value for SIP
##   signaling.
##   (1024 -65503).
##  This parameter is supported by:
```

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```
##      J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later -
the default and minimum values are 5062.
##      J169/J179 SIP R1.5.0 - the default and minimum values are 5062
##      J129 SIP R1.0.0.0 (or R1.1.0.0) - the default is 1024
##      96xx SIP R2.0 and later
##      96x1 SIP R6.0 and later; Pre R7.1.1.0 the default is 1024. R7.1.1.0.0+ the default and
minimum values are 5062.
##      H1xx SIP R1.0 and later
## SET SIG_PORT_LOW 1024
##
## Signaling 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).
##  This parameter is supported by:
##      J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later -
the maximum value is 60473
##      J169/J179 SIP R1.5.0 - the maximum value is 60473
##      J129 SIP R1.0.0.0 (or R1.1.0.0) - the maximum value is 64511
##      96xx SIP R2.0 and later
##      96x1 SIP R6.0 and later; Pre R7.1.0.0 the maximum value is 64511. R7.1.1.0.0+ the maximum
value is 60473.
##      H1xx SIP R1.0 and later
## SET SIG_PORT_RANGE 64511
##
############################################################
##
##      96xx/96x1/H1xx/J129/J139/J159/J169/J179 SIP TELEPHONE SETTINGS
##
############################################################
##
## INGRESS_DTMF_VOL_LEVEL specifies the power level of tone, expressed in dBm0.
##  The possible values are in the range of -20dBm to -7dBm.
##  The default value is -12dBm.
##  This parameter is supported by:
##      J129 SIP R1.0.0.0 (or R1.1.0.0), J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J139
SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later
##      96xx SIP R2.0 and later
##      96x1 SIP R6.0 and later
##      H1xx SIP R1.0 and later
## SET INGRESS_DTMF_VOL_LEVEL -12
##

#########################################
##
## Conference transfer on primary appearance
##  When CONF_TRANS_ON_PRIMARY_APPR is set to 1,
##  conference and transfer setup will first attempt
##  to use an idle primary call appearance even if
##  initiated from a bridged call appearance.
##  If an idle primary call appearance is not available,
##  then an idle bridged call appearance will be used.
##  Conference and transfer setup initiated from a bridged call
##  appearance when no idle primary call appearance is available
```

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```
##  will next attempt to use an idle bridged call appearance of
##  the same extension and if not available, an idle bridged call
##  appearance of a different extension.
##  Note:  When CONF_TRANS_ON_PRIMARY_APPR is set to 1, AUTO_SELECT_ANY_IDLE_APPR is ignored.
##
##  When CONF_TRANS_ON_PRIMARY_APPR is set to 0,
##  conference and transfer setup initiated from a primary call
##  appearance will first attempt to use an idle primary call appearance.
##  If an idle primary call appearance is not available, it will use an idle
##  bridged call appearance regardless of the setting of AUTO_SELECT_ANY_IDLE_APPR.
##  Conference and transfer setup initiated from a bridged call appearance will attempt
##  to use an idle bridged call appearance of the same extension.
##  If an idle bridged call appearance of the same extension is not available
##  and AUTO_SELECT_ANY_IDLE_APPR is set to 1, then conference and transfer
##  setup will use any idle call appearance (primary or bridged).
##  It will first attempt to find an idle primary call appearance and if not
##  available will then attempt to find an idle bridged call appearance of a different extension.
##  However, if AUTO_SELECT_ANY_IDLE_APPR is set to 0, transfer and conference setup
##  initiated on a bridged call appearance will be denied if an idle bridged call appearance
##  of the same extension is not available.
##
##  The Default value of CONF_TRANS_ON_PRIMARY_APPR is 0.
## Note:  These parameters are supported on SIP release R2.4.1 and later release of 96xx SIP
telephones.
## Note: CONF_TRANS_ON_PRIMARY_APPR is supported by J100 SIP R2.0.0.0 and later (J169/J179 only),
J139 SIP R3.0.0.0 and later and J159 SIP R4.0.3.0 and later.
##
## Auto Select any idle appearance
##  When AUTO_SELECT_ANY_IDLE_APPR is active then any idle appearance is selected.
##  When AUTO_SELECT_ANY_IDLE_APPR is set to 0 and CONF_TRANS_ON_PRIMARY_APPR is 0,
##  then if no associated call appearance is selected,
##  the conference or transfer operation will be denied.
##  When AUTO_SELECT_ANY_IDLE_APPR is set to 1 and CONF_TRANS_ON_PRIMARY_APPR is 0,
##  then if no associated call appearance is selected, the conference or transfer
##  operation will be tried on any available call appearance (primary or bridged).
##  This parameter is supported by:
##       96x0 SIP R2.4.1 and later releases
##       J100 SIP R2.0.0.0 and later (J169/J179 only), J139 SIP R3.0.0.0 and later, J159 SIP
R4.0.3.0 and later
## SET AUTO_SELECT_ANY_IDLE_APPR 0


## WAIT_FOR_CALL_OPERATION_RESPONSE specifies the time in seconds before providing a response for
user initiated call operation.
##  This parameter is applicable to all server environments (Aura, IP Office and 3PCC).
##  When user goes off-hook, then phone sends an invite. If there is no response from the SIP
proxy for the number of seconds defined in WAIT_FOR_CALL_OPERATION_RESPONSE,
##  it will result in a user notification that the operation is in progress but is delayed.
##  Value range is 0-4. The default is 3. If the value is 0, then the notification will never be
shown.
##  This parameter is supported by:
##       J100 SIP R4.0.0.0 and later (value 0 is supported in R4.0.1.0.12 and later), J159 SIP
R4.0.3.0 and later
##  SET WAIT_FOR_CALL_OPERATION_RESPONSE 2
##
## Dynamic Feature Set Discovery
```

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```
##  If the DISCOVER_AVAYA_ENVIRONMENT parameter value is 1, the phone discovers (determines)
##  if that controller supports the AST feature set or not. The phone will send a SUBSCRIBE
##  request to the active controller for the Feature Status Event Package (avaya-cm-feature-
status).
##  If the request succeeds, then the phone proceeds with PPM Synchronization.
##  If the request is rejected, is proxied back to the phone or does not receive a response,
##  the phone will assume that AST features are not available.
##  If the parameter value is 0, the phone operates in a mode where AST features are not
available.
##  Note: This parameter is supported on R2.4.1 and later release of 96xx SIP telephones, H1xx
SIP R1.0 and later
##  and J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later and J159 SIP R4.0.3.0 and later.
##  For IP office and 3PCC environments this parameter shall be set to 0.
## SET DISCOVER_AVAYA_ENVIRONMENT 1

## PSTN Access Prefix
##  ENABLE_REMOVE_PSTN_ACCESS_PREFIX parameter allows telephone to
##  perform digit manipulation during failure scenarios. This parameter
##  allows removal of PSTN access prefix from the outgoing number.
##  0 - PSTN access prefix is retained in the outgoing number
##  1 - PSTN access prefix is stripped from the outgoing number.
##  Note: This parameter is supported on R2.4.1 and later release of 96xx SIP telephones, H1xx
SIP R1.0 and later
##  and J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and later, J139 SIP R3.0.0.0 and
later and J159 SIP R4.0.3.0 and later when the phone is failed over.
## This parameter is not supported in IP Office and 3PCC environments as there is no support for
failover.
## SET ENABLE_REMOVE_PSTN_ACCESS_PREFIX 0
##
## Local Dial Area Code
##  LOCAL_DIAL_AREA_CODE indicates whether user must dial area code for calls within same
##  area code regions. when LOCAL_DIAL_AREA_CODE is enabled (1), the area code parameter (PHNLAC)
##  should also be configured (ie. not the empty string).
##  0 - User don't need to dial area code.
##  1 - User need to dial area code.
##  Note: This parameter is supported on R2.4.1 and later release of 96xx SIP telephones
##        when the phone is failed over.
##  Note: This parameter is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and
later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
## SET LOCAL_DIAL_AREA_CODE 0
##
## Phone's Local Area Code
##  When PHNLAC is set,it indicates the telephone'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 telephone.
##  Note: This parameter is supported on R2.4.1 and later release of 96xx SIP telephones
##        when the phone is failed over.
##  Note: This parameter is supported by J129  SIP  R1.0.0.0 (or R1.1.0.0), J100 SIP R2.0.0.0 and
later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later.
## SET PHNLAC ""
##
#####################  SIP USER CREDENTIALS SETTINGS  #####################
##
## SIP User Credentials settings
##   Configure Username, Password and User ID to be used
```

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```
##   for SIP Registration. Usernames are often identical
##   to User ID.
##   FORCE_SIP_USERNAME replaces user field entered by user during Login
##   FORCE_SIP_PASSWORD replaces password entered by user during Login
##   FORCE_SIP_EXTENSION replaces User ID entered by user during Login
##   If these are set, the user will not be prompted to Login on power cycle.
##  Note: This parameter is supported by:
##       Avaya Vantage Devices SIP R2.0.1.0 and later; FORCE_SIP_PASSWORD and FORCE_SIP_EXTENSION
are supported in Aura, IP Office and OpenSIP environments.
##             FORCE_SIP_USERNAME is not supported by Avaya Vantage Devices.
##       J129 SIP  R1.1.0.0, J169/J179 SIP R1.5.0, J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0
and later
##       96x1 SIP R7.1.1.0 and later
## SET FORCE_SIP_USERNAME "7415"
## SET FORCE_SIP_PASSWORD "2222"
## SET FORCE_SIP_EXTENSION "741515"
##
## GET $MACADDR will request for the "MACADDR" file from the HTTP/HTTPS Server where "$MACADDR"
which will be replaced by the telephone's MAC address.
##  Note: This parameter is supported by J129 SIP  R1.1.0.0, J169/J179 R1.5.0, J100 SIP R2.0.0.0
and later, J139 SIP R3.0.0.0 and later, J159 SIP R4.0.3.0 and later, 96x1 SIP R7.1.1.0 and later
##        and Avaya Vantage Devices SIP R2.0.1.0 and later
## GET $MACADDR.txt
##



############################################################
##
# PRODUCT_LINE_SETTINGS
##
############################################################

IF $MODEL4 SEQ J129 GOTO SETTINGSJ100
IF $MODEL4 SEQ J139 GOTO SETTINGSJ100
IF $MODEL4 SEQ J159 GOTO SETTINGSJ100
IF $MODEL4 SEQ J169 GOTO SETTINGSJ100
IF $MODEL4 SEQ J179 GOTO SETTINGSJ100

############################################################
##
# PER_MODEL_SETTINGS
##
############################################################

# SETTINGSJ159
##
##########  Add settings for J159 telephones below  ##########
##
## AUDIOSTHS specifies the level of sidetone in the handset.
##  Value  Operation
##    0    NORMAL level for most users (default)
##    1    three levels softer than NORMAL
##    2    OFF (inaudible)
##    3    one level softer than NORMAL
##    4    two levels softer than NORMAL
```

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```
##    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
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J179 SIP  R1.5.0; J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET AUDIOSTHS 1
##
## AUDIOSTHD specifies the level of sidetone in the headset.
##  Value  Operation
##    0    NORMAL level for most users (default)
##    1    three levels softer than NORMAL
##    2    OFF (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
##  This parameter is supported by:
##       J169/J179 H.323 R6.7 and later
##       J179 SIP  R1.5.0; J100 SIP R2.0.0.0 and later, J159 SIP R4.0.3.0 and later
## SET AUDIOSTHD 1
##
############### End of J159 model-specific settings ############

############### END OF CONFIGURATION FILE ######################
################################################################################
##
## HISTORY TABLE
##

##  14-October-2019:
##      1. Update the description of ALLOW_LOGOUT_WHEN_LOCKED.
##      2. 169/J179 R6.8.3 H.323 and 96x1 H.323 R6.8.3:
##         a. Add IGNORESTAT.
##      3. Update the description of CALL_PICKUP_FAC, CALL_PICKUP_BARGEIN_FAC, CALL_UNPARK_FAC
and CALL_PARK_FAC.
##      4. J100 SIP R4.0.3.0 and later:
##         a. Add new parameters: IGNORE_CONTACT_HEADER_DISPLAY_NAME,
SCREENSAVER_IMAGE_SECONDARY, SCREENSAVER_IMAGE_DISPLAY_SECONDARY,
SCREENSAVER_IMAGE_SELECTABLE_SECONDARY,
##             BACKGROUND_IMAGE_SECONDARY, BACKGROUND_IMAGE_DISPLAY_SECONDARY,
BACKGROUND_IMAGE_SELECTABLE_SECONDARY, PROVIDE_PRIMARY_RINGTONE, PROVIDE_RING_REMINDER_RINGTONE,
##             PROVIDE_CALL_PARK_RINGTONE, PROVIDE_BLF_INCOMING_CALL_RINGTONE,
PROVIDE_BLF_CALL_PARK_RINGTONE, PROVIDE_PRIORITY_ALERT_RINGTONE, PROVIDE_ALTERNATE_NUM1_RINGTONE,
##             PROVIDE_ALTERNATE_NUM2_RINGTONE, RINGTONE_DEFAULT_CALL_FWD_RING,
RINGTONE_DEFAULT_PRIMARY, RINGTONE_DEFAULT_RING_REMINDER, RINGTONE_DEFAULT_CALL_PARK,
##             RINGTONE_DEFAULT_BLF_INCOMING_CALL, RINGTONE_DEFAULT_BLF_CALL_PARK,
RINGTONE_DEFAULT_PRIORITY_ALERT, RINGTONE_DEFAULT_ALTERNATE_NUM1,
RINGTONE_DEFAULT_ALTERNATE_NUM2,
##             RINGTONE_ALERTINFO, PHONE_SCREEN_MODE, SHORTCUT_ACTION_BLF_PARK, MP_ENABLED,
MP_CODEC, MP_PACKET_SIZE, MP_GROUPS_TO_LISTEN, MP_GROUPS_TO_SEND.
```

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```
##         b. Mark BACKGROUND_IMAGE, BACKGROUND_IMAGE_DISPLAY and BACKGROUND_IMAGE_SELECTABLE as
supported by J139/J159 SIP R4.0.3.0 and later.
##         c. Mark DIRENABLED_PLATFORM, DIRSRVR, DIRSRVRPRT, DIRTOPDN, DIRSECURE as supported by
J139, J159, J169, J179 SIP R4.0.3.0 and later. Add new parameters: DIRSEARCH_FIELDS,
##             DIRSHOW_FIELDS, DIRNAME_FIELDS, DIRNUMBER_FIELDS.
##         d. Mark the following parameters as supported by J159 SIP R4.0.3.0 and later:
ENABLE_OOD_RESET_NOTIFY, TEAM_BUTTON_RING_TYPE, GUESTLOGINSTAT, GUESTDURATION, GUESTWARNING,
##             ENABLE_MLPP, MLPP_NET_DOMAIN, MLPP_MAX_PREC_LEVEL, ENABLE_PRECEDENCE_SOFTKEY,
DSCPAUD_FO, DSCPAUD_FL, DSCPAUD_IM, DSCPAUD_PR, DSCPMGMT, ENABLE_REDIAL_LIST, L2Q,
##             L2QVLAN, L2QAUD, L2QSIG, VLANSEP, VLANSEPMODE, PHY2VLAN, PHY2PRIO, PHY2TAGS,
DSCPAUD, DSCPSIG, WBCSTAT, QLEVEL_MIN, DHCPSTD, DHCPSTDV6, VLANTEST, REUSETIME, DNSSRVR,
##             DOMAIN, SHOW_LAST_EXTENSION, SIPDOMAIN, SIP_CONTROLLER_LIST,
SIP_CONTROLLER_LIST_2, ENABLE_UDP_TRANSPORT, SIPREGPROXYPOLICY, SIMULTANEOUS_REGISTRATIONS,
CONNECTION_REUSE,
##             ENABLE_PPM_SOURCED_SIPPROXYSRVR, CONFIG_SERVER_SECURE_MODE, VOLUME_UPDATE_DELAY,
ENABLE_AVAYA_ENVIRONMENT, SMGR_AUTO_FAVORITE, DIALPLAN, PHNNUMOFSA, SNTPSRVR, SNTP_SYNC_INTERVAL,
##             GMTOFFSET, DSTOFFSET, DSTSTART, DSTSTOP, WAIT_FOR_REGISTRATION_TIMER,
REGISTERWAIT, WAIT_FOR_UNREGISTRATION_TIMER, WAIT_FOR_INVITE_RESPONSE_TIMEOUT,
OUTBOUND_SUBSCRIPTION_REQUEST_DURATION,
##             NO_DIGITS_TIMEOUT, INTER_DIGIT_TIMEOUT, FAILED_SESSION_REMOVAL_TIMER,
TCP_KEEP_ALIVE_STATUS, TCP_KEEP_ALIVE_TIME, TCP_KEEP_ALIVE_INTERVAL, CONTROLLER_SEARCH_INTERVAL,
##             ASTCONFIRMATION, FAST_RESPONSE_TIMEOUT, RDS_INITIAL_RETRY_TIME,
RDS_MAX_RETRY_TIME, RDS_INITIAL_RETRY_ATTEMPTS, SIP_TIMER_T1, SIP_TIMER_T2, SIP_TIMER_T4,
FORBIDDEN_SESSION_REMOVAL_TIMER,
##             CONFERENCE_FACTORY_URI, EVENT_NOTIFY_AVAYA_MAX_USERS, ENABLE_PRESENCE,
PRESENCE_ACL_CONFIRM, ALLOW_DND_SAC_LINK_CHANGE, DND_SAC_LINK, AWAY_TIMER_VALUE, AWAY_TIMER,
##             EXCHANGE_SERVER_LIST, PROVIDE_EXCHANGE_CONTACTS, PROVIDE_EXCHANGE_CALENDAR,
EXCHANGE_USER_DOMAIN, EXCHANGE_AUTH_USERNAME_FORMAT, EXCHANGE_EMAIL_DOMAIN,
ENABLE_EXCHANGE_REMINDER,
##             EXCHANGE_REMINDER_TIME, EXCHANGE_SNOOZE_TIME, EXCHANGE_REMINDER_TONE,
EXCHANGE_NOTIFY_SUBSCRIPTION_PERIOD, SPEAKERSTAT, MUTE_ON_REMOTE_OFF_HOOK, SDPCAPNEG,
ENFORCE_SIPS_URI,
##             100REL_SUPPORT, PLAY_TONE_UNTIL_RTP, PHONEKEY, TEAM_BUTTON_RING_TYPE,
LOCALLY_ENFORCE_PRIVACY_HEADER, ENABLE_SIP_USER_ID, ENABLE_STRICT_USER_VALIDATION,
BRANDING_VOLUME,
##             ENABLE_OOD_MSG_TLS_ONLY, TEAM_BUTTON_REDIRECT_INDICATION, ENABLE_BLIND_TRANSFER,
USE_CONTACT_IN_REFERTO, PROVIDE_KEY_REPEAT_DELAY, HEADSET_PROFILE_DEFAULT, HEADSET_PROFILE_NAMES,
##             ACOUSTIC_EXPOSURE_PROTECT_MODE_DEFAULT, HANDSET_PROFILE_DEFAULT,
HANDSET_PROFILE_NAMES, PHNEMERGNUM, PHNMOREEMERGNUMS, ENABLE_SHOW_EMERG_SK,
ENABLE_SHOW_EMERG_SK_UNREG,
##             LOG_DIALED_DIGITS, WIFISTAT, WLAN_MAX_AUTH_FAIL_RETRIES,
ENABLE_NETWORK_CONFIG_BY_USER, WLAN_COUNTRY, WLAN_ENABLE_80211D, WLAN_ESSID, WLAN_SECURITY,
WEP_DEFAULT_KEY,
##             WEP_KEY_LEN, WEP_KEY1, WEP_KEY2, WEP_KEY3, WEP_KEY4, WLAN_PASSWORD,
WLAN_WPA2E_EAP_METHOD, WLAN_WPA2E_EAP_PHASE2, WLAN_WPA2E_IDENTITY, WLAN_WPA2E_ANONYMOUS_IDENTITY,
##             WLAN_L2QAUD, WLAN_L2QSIG, WLAN_DSCPAUD, WLAN_DSCPSIG, HEADSYS, SKILLSCREENTIME,
UUIDISPLAYTIME, ENTRYNAME, CC_INFO_TIMER, RECORDINGTONE_INTERVAL, RECORDINGTONE_VOLUME,
##             TRUSTCERTS, MAX_TRUSTCERTS, ENABLE_PUBLIC_CA_CERTS, TLSSRVRID, FQDN_IP_MAP,
SERVER_CERT_RECHECK_HOURS, CERT_WARNING_DAYS, DELETE_MY_CERT, BLOCK_CERTIFICATE_WILDCARDS,
##             TLS_VERSION, HTTPPROXY, HTTPEXCEPTIONDOMAINS, STUN_SERVER_ADDRESS,
STUN_UDP_INITIAL_TIMEOUT_MSEC, STUN_UDP_MAX_TRANSMISSIONS, STUN_UDP_MAX_MEDIA_TRANSMISSIONS,
##             NAT_SIGNALING_KEEPALIVE_ENABLED, NAT_SIGNALING_KEEPALIVE_OVERRIDE_SEC, MYCERTURL,
MYCERTCN, MYCERTDN, MYCERTCAID, MYCERTKEYLEN, MYCERTRENEW, SCEPPASSWORD,
##             SCEPENCALG, PKCS12URL, PKCS12_PASSWD_RETRY, DOT1XSTAT, DOT1X, DOT1XEAPS,
FIPS_ENABLED, OCSP_ENABLED, OCSP_ACCEPT_UNK, OCSP_NONCE, OCSP_URI, OCSP_URI_PREF,
OCSP_TRUSTCERTS,
##             OCSP_HASH_ALGORITHM, OCSP_USE_CACHE, OCSP_CACHE_EXPIRY, TPSLIST, SUBSCRIBELIST,
PUSHCAP, PUSHPORT, ENABLE_WEBSERVER, WEBSERVER_ON_HTTP, WEB_HTTP_PORT, WEB_HTTPS_PORT,
##             FORCE_WEB_ADMIN_PASSWORD, BAKLIGHTOFF, SCREENSAVERON, SCREENSAVER_IMAGE,
SCREENSAVER_IMAGE_DISPLAY, SCREENSAVER_IMAGE_SELECTABLE, BACKLIGHT_SELECTABLE, WMLIDLETIME,
##             WMLIDLEURI, ENABLE_PHONE_LOCK, PHONE_LOCK_IDLETIME, ENABLE_G711A, ENABLE_G711U,
ENABLE_G722, ENABLE_G726, G726_PAYLOAD_TYPE, ENABLE_G729, ENABLE_OPUS, OPUS_PAYLOAD_TYPE,
```

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```
##             SEND_DTMF_TYPE, DTMF_PAYLOAD_TYPE, SYMMETRIC_RTP, PHNMUTEALERT_BLOCK, MATCHTYPE,
SIG, PHY1STAT, PHY2STAT, PHY2_AUTOMDIX_ENABLED, PROCSTAT, PROCPSWD, ADMIN_PASSWORD,
##             ADMIN_LOGIN_ATTEMPT_ALLOWED, ADMIN_LOGIN_LOCKED_TIME, SNMPSTRING, SNMPADD,
LLDP_ENABLED, LOGSRVR, LOCAL_LOG_LEVEL, LOG_CATEGORY, ENABLE_RECORDING, WARNING_FILE,
##             SSH_ALLOWED, SSH_BANNER_FILE, SSH_IDLE_TIMEOUT, EASG_SITE_CERTS,
EASG_SITE_AUTH_FACTOR, CERT_WARNING_DAYS_EASG, AUTHCTRLSTAT, SLMSTAT, SLMCAP, SLMCTRL, SLMPERF,
##             SLMPORT, SLMSRVR, ENHDIALSTAT, PHNCC, PHNDPLENGTH, PHNIC, PHNLD, PHNLDLENGTH,
PHNOL, ELD_SYSNUM, AGCHAND, AGCHEAD, AGCSPKR, AUDIOENV, RINGTONESTYLE, RINGTONES,
##             RINGTONES_UPDATE, PROVIDE_CF_RINGTONE, HTTPSRVR, HTTPDIR, HTTPPORT, AUTH, TLSSRVR,
TLSDIR, TLSPORT, DES_STAT, RTCPMON, RTCPMONPORT, RTCPMONPERIOD, ICMPDU, ICMPRED,
##             BRURI, AUDASYS, MSGNUM, DHCPSTAT, PRIVACY_SLAAC_MODE, DUAL_IPPREF, IPV6STAT,
SIGNALING_ADDR_MODE, MEDIA_NEG_PREFERENCE, MEDIA_ADDR_MODE, IPV6DADXMITS, CALLFWDSTAT,
##             CALLFWDDELAY, CALLFWDADDR, ENABLE_AUTO_ANSWER_SUPPORT, AUTO_ANSWER_MUTE_ENABLE,
ENABLE_DND, DND_PRIORITY_OVER_CFU_CFB, HOLD_REMINDER_TIMER, RTCP_XR, MEDIAENCRYPTION,
##             ENCRYPT_SRTCP, ENABLE_IPOFFICE, MEDIA_PRESERVATION, PRESERVED_CALL_DURATION,
SUBSCRIBE_LIST_NON_AVAYA, USER_STORE_URI, ENABLE_3PCC_ENVIRONMENT, 3PCC_SERVER_MODE,
##             BLF_LIST_URI, CALL_PICKUP_FAC, CALL_PICKUP_BARGEIN_FAC, CALL_UNPARK_FAC,
CALL_PARK_FAC, ALLOW_BLF_LIST_CHANGE, SHORTCUT_ACTION_BLF, SHORTCUT_ACTION_CONTACT,
##             SHORTCUT_ACTION_AUTODIAL, SHORTCUT_ACTION_BLF_PARK, XSI_URL, XSI_CHANNEL_DURATION,
XSI_HEARTBEAT, FORCE_XSI_USER_ID, FORCE_XSI_WEB_PASSWORD, PRIMARY_LINE_TYPE,
##             PRIMARY_LINE_BARGE_IN_ENABLED, SCA1_ENABLED, SCA2_ENABLED, SCA3_ENABLED,
SCA1_MAX_CALL_APPEARANCES, SCA2_MAX_CALL_APPEARANCES, SCA3_MAX_CALL_APPEARANCES,
##             SCA1_SIPUSERID, SCA2_SIPUSERID, SCA3_SIPUSERID, SCA1_USERNAME, SCA2_USERNAME,
SCA3_USERNAME, SCA1_PASSWORD, SCA2_PASSWORD, SCA3_PASSWORD,
##             SCA1_EXTENSION, SCA2_EXTENSION, SCA3_EXTENSION, SCA1_BARGE_IN_ENABLED,
SCA2_BARGE_IN_ENABLED, SCA3_BARGE_IN_ENABLED, SCA_LINE_SEIZE_DURATION,
PROVIDE_SHARED_LINE_CONFIG,
##             BW_ENABLE_DIR, BW_ENABLE_DIR_ENTERPRISE, BW_ENABLE_DIR_ENTERPRISE_COMMON,
BW_ENABLE_DIR_GROUP, BW_ENABLE_DIR_GROUP_COMMON, BW_ENABLE_DIR_PERSONAL, BW_ENABLE_DIR_CUSTOM,
##             BW_DIR_ENTERPRISE_DESCRIPTION, BW_DIR_ENTERPRISE_COMMON_DESCRIPTION,
BW_DIR_GROUP_DESCRIPTION, BW_DIR_GROUP_COMMON_DESCRIPTION, BW_DIR_PERSONAL_DESCRIPTION,
##             BW_DIR_CUSTOM_DESCRIPTION, BW_DIR_ENTERPRISE_EXTENSION,
BW_DIR_ENTERPRISE_COMMON_EXTENSION, BW_DIR_GROUP_EXTENSION, BW_DIR_GROUP_COMMON_EXTENSION,
##             BW_DIR_PERSONAL_EXTENSION, BW_DIR_CUSTOM_EXTENSION, DOWNLOADABLE_DIRECTORY,
PROVIDE_OPTIONS_SCREEN, PROVIDE_NETWORKINFO_SCREEN, PROVIDE_LOGOUT, DISPLAY_SSL_VERSION,
##             ENABLE_CALL_LOG, ENABLE_REDIAL, ENABLE_CONTACTS, CONTACT_NAME_FORMAT,
SYSTEM_LANGUAGE, LANGUAGES, COUNTRY, TIMEFORMAT, DAYLIGHT_SAVING_SETTING_MODE, RTP_PORT_LOW,
##             RTP_PORT_RANGE, SIG_PORT_LOW, SIG_PORT_RANGE, INGRESS_DTMF_VOL_LEVEL,
CONF_TRANS_ON_PRIMARY_APPR, AUTO_SELECT_ANY_IDLE_APPR, WAIT_FOR_CALL_OPERATION_RESPONSE,
##             DISCOVER_AVAYA_ENVIRONMENT, PSTN_VM_NUM, ENABLE_REMOVE_PSTN_ACCESS_PREFIX,
LOCAL_DIAL_AREA_CODE, PHNLAC, FORCE_SIP_USERNAME, FORCE_SIP_PASSWORD, FORCE_SIP_EXTENSION,
##             AUDIOSTHS, AUDIOSTHD.
##############################################################################################
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