



Avaya J100 Series SIP Release 3.0.0.0 Readme

This file is the Readme for the Avaya J100 Series SIP Release 3.0.0.0 software (J100 SIP 3.0.0). This file describes the contents of the July 2018 (**3.0.0.0.20**) release software distribution package.

J100 SIP 3.0.0 software is supported on the Avaya J129, J139, J169, and J179 IP Phones used with Avaya Aura®, Avaya IP Office™, and select 3PCC (3rd party call control platforms). J100 SIP 3.0.0 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.

Compatibility

The Avaya J129, J139, J169 and J179 IP Phones using J100 SIP 3.0.0 software is supported with:

- Avaya Aura® Platform 6.2 FP4 (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
- 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
- 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
- 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
- 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
- IP Office™ 10.0 SP2 or later (J129 only)
- IP Office™ 11.0 or later (J139 requires 11.0 SP1 or later)
- Avaya Aura® Call Center Elite 7.0.1.0¹, 7.1.0.0¹, 8.0.0.0¹
- 3PCC (3rd party call control) Platform
 - Broadsoft Broadworks R21SP1
 - Zang Office R1.0
 - Edgewater Network device (Edgemarc 4550).

¹ J169/J179 IP Phone is supported with CC Elite. The J129/J139 IP Phone are not supported with CC Elite.

New Features in J100 SIP 3.0.0

Avaya J100 Series SIP Release 3.0.0 contains the following new features

New with this release	Description
Support for J139 IP Phone	J100 Release 3.0.0 introduces support for the J139 IP Phone.
Support for PushAPI	J100 Release 3.0.0 adds support for the following PushAPI functionality: <ul style="list-style-type: none">- receipt of audio push- receipt of topline push The following pushAPI functionality is NOT supported: <ul style="list-style-type: none">- receipt of WML push- recording of audio push
Improved DES (Device Enrollment Services) Capability	J100 Release 3.0.0 adds support for NEC (Numeric Enrollment Code).
Interworking with Broadsoft Device Management	BroadWorks Device Management provides the capability to generate automatically device-specific configuration files for download by the access device. Download capability for firmware and other resource files required by the device is also provided by this feature.

Documentation for J100 SIP 3.0.0.0

The following documentation is new for this release:

- [Using Avaya J139 IP Phone SIP](#)
- [Using Avaya J139 IP Phone SIP in third-party call control setup](#)
- [Avaya J139 IP Phone SIP Quick Reference](#)

The following documentation has been updated for this release:

- [Avaya J100 Series IP Phone Overview and Specifications](#)
- [Installing and Administering Avaya J100 Series IP Phone](#)
- [Installing and Administering Avaya J100 series IP Phone in third-party call control setup](#)
- [Using Avaya J169/J179 IP Phone SIP](#)
- [Using Avaya J169/J179 IP Phone SIP in third-party call control setup](#)
- [Avaya J169/J179 IP Phone SIP Quick Reference](#)
- [Using Avaya J129 IP Phone SIP](#)
- [Using Avaya J129 IP Phone SIP in third party call control setup](#)
- [Avaya J129 IP Phone SIP Quick Reference](#)

The following documentation has not been updated for this release and is included here for reference:

- [Using Avaya J169/J179 IP Phone SIP in a Call Center](#)
- [Using Avaya JBM24 Button Module](#)

These documents are available on <http://support.avaya.com> under "J100 Series IP Phones "
-> "SIP 3.0.x" -> Documents

J100 SIP 3.0.0.0 (3.0.0.0.20) Package Content

The J100 SIP 3.0.0.0 package (J100-IPT-SIP-R3_0_0_0-062218.zip) contains all the files necessary to upgrade Avaya new or previously installed Avaya J129/J139/J169/J179 IP Phones to the J100 SIP 3.0.0.0 software.

- FW_S_J129_R3_0_0_0_20.bin – application binary file for J129
- FW_S_J139_R3_0_0_0_20.bin – application binary file for J139
- FW_S_J169_R3_0_0_0_20.bin – application binary file for J169
- FW_S_J179_R3_0_0_0_20.bin – application binary file for J179
- J100Upgrade.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_Arabic.xml
 - 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_J129_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_Turkish.xml
- Eight extended Korean ring tone files:
 - KoreanRT1.xml
 - KoreanRT2.xml
 - KoreanRT3.xml
 - KoreanRT4.xml
 - KoreanRT5.xml
 - KoreanRT6.xml
 - KoreanRT7.xml
 - KoreanRT8.xml
- One certificate file:
 - av_prca_pem_2033.txt – Avaya Product Root CA certificate with an expiration date of 2033
- Avaya-J100iPhone-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
- Avaya Global Software License Terms 102016v1.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](#).**

Advisements with J100 SIP 3.0.0.0 software

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/J169/J179 hardware which are pre-configured for interworking with Open SIP platforms such as Broadsoft and Zang Office. **When using J100 3.0.0 or later software, the “3PCC” hardware cannot be converted for use on Avaya Aura® or Avaya IP Office™.**

Limitations with IPv6

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

- Deskphones cannot assign their own IP address (SLAAC). Only DHCPv6 is supported.
- Deskphones cannot resolve domain names into IPv6 addresses (AAAA records). As a result, a FQDN cannot be used for server configurations.
- Extended rebind is not supported.
- HTTP/HTTPS over IPv6 is not supported. The deskphones must use HTTP/HTTPS over IPv4 to retrieve settings files, software files, audio files, and language files.
- The following functionality is only supported via IPv4
 - RTCP
 - WML and Push
 - Microsoft Exchange integration
 - SNTP
 - Syslog (In addition, the syslog file will not show IPv6 addresses)
 - SCEP
 - Avaya Diagnostic Server (ADS / SLAMon)
 - SSH / Telnet (used only by Avaya Support)
 - SNMP
 - Shared Control / Deskphone Mode
 - Interworking with CC Elite.
- FQDN cannot be used for server configuration

As a result of these limitations, deskphones will only be supported in a mixed IPv4/IPv6 environment.

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

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.

Removal of Avaya SIP Root CA Certificate

The Avaya SIP Root CA Certificate (av_sipca_pem_2027.txt) is not included in the installation package.

Support for OCSP

J100 SIP software supports OCSP (Online Certificate 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, and HTTPS.

FIPS 140-2 Cryptographic libraries

J100 SIP 1.5.0 and later software supports an administrator-configurable option to utilize FIPS 140-2 certified algorithms for cryptographic operations. The following features support secure operations when FIPS mode is enabled:

- The crypto random generator complies with [SP 800-90] DRBG specification
- Certificate signature authentication
- SIP signaling over TLS
- SRTP
- Downloads over HTTPS of settings, upgrade files, trusted certificates, and PKCS#12 files (Note that TLSSVR must be set and HTTPSRVR must be empty)
- OCSP

Microsoft Exchange integration uses a non-certified algorithm and must be disabled when in FIPS 140-2 mode.

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.

J169 IP Phone – Minimum Software Release

The J169 IP Phone may be upgraded from SIP R1.5.0.0 to SIP R2.0.0.0 software. However, downgrade from R2.0.0.0 to 1.5.0.0 is not recommended.

J179 IP Phone – Minimum Software Release

The J179 IP Phone may be upgraded from SIP R1.5.0.0 to SIP R2.0.0.0 software. However, downgrade from R2.0.0.0 to 1.5.0.0 is not recommended.

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

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 Domain Names (FQDN) is only supported in 3PCC 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"](http://support.avaya.com) which is available on <http://support.avaya.com> for known limitations.

Language support

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

Ringtone and Ringtone Wave Files

Ringtone wave files should be placed in the root directory of the HTTPSRVR. Additionally, numeric only conventions should be avoided with ringtone names.

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 3PCC 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 SLMSRV 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 IP Phone, 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.

"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.

² J129 does not support a headset

J129/J139/J169/J179 - Aliasing

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

Until Avaya Aura® Communication Manager provides native support of the J169/J179 IP Phones, there are JBM24 feature administration limitations. To preserve existing settings for a user, feature buttons will be ported the first time they login to the phone. This allows an existing user with a 9608/9608G/9611G/9641/9641G IP Phone and BM12/SBM24 to upgrade their phone to a J169/J179 with JBM24 and preserve their BM feature buttons. Any further changes via System Manager after the initial login will not be seen until Avaya Aura® Communication Manager provides native support of the J169/J179 IP Phones and corresponding JBM24 button module.

Demo Certificates – Avaya Aura® Session Manager 6.3.8 and newer



New installations of Avaya Aura® Session Manager Release 6.3.8 and newer 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. It is highly recommended to replace the demo Avaya certificates. Refer to the Session Manager Administrator Guide for more details.

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.

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
- 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)
- Call Pickup, Hunt Group Busy, Team Button, Enhanced Call Forward, Dial Intercom, Exclusion, LNCC, Priority Calls, Whisper Page
- Interworking with Contact Center Elite (CC Elite)

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

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 not supported by the Avaya J139 IP Phone when deployed on IP Office™:

Advanced Call Presentation / Handling:

- MADN
- Bridged Appearances
- Coverage Appearances
- User BLF
- Group BLF

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)

Features not supported on the J169 Phone

The following features are not supported by the J169 IP Phone with J100 software:

- WML browser, simultaneous display of caller name and number, Favorites

Features not supported on the J179 Phone

The following features are not supported by the J179 IP Phone with J100 software:

- WML browser, simultaneous display of caller name and number, Favorites

J169/J179 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

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 J169/J179 IP Phone when deployed on IP Office™:

Advanced Call Presentation / Handling:

- MADN
- Bridged Appearances
- Coverage Appearances
- User BLF
- Group BLF

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)

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

- Personalization (i.e. ability to reconfigure the button layout)
- Push API

Deploying the J129/J139/J169/J179 in 3PCC Platform

The J129/J139/J169/J179 are supported with Broadsoft Broadworks R21SP1, Zang Office R1.0 and Edgewater Network device (Edgemarc 4550). 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. J129 phone to work in 3PCC 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

Provisioning of File Server Address

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

- DHCP
- LLDP
- Device Interface

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.

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 HTTPSRRV "" in the settings file to override the HTTPSRRV value received from DHCP. Commenting out the HTTPSRRV parameter will not override the value received from DHCP.

J100 3.0.0.0 Resolved Issues (since J100 2.0.0.0)

The following table includes issues which are resolved with this release of software compared to J100 2.0.0.0.45

External ID	Internal ID	Issue Description
Avaya Aura®		
	SIP96X1-29616	SIP proxy server does not switch to IPv4 when invalid Phone(V6) is set in WebUI
	SIP96X1-30159	J179/J169 – App in Customize key is not translated to new language, which is different from English
	SIP96X1-30214	Characters of features on JBM24 are Bold and do not display clearly in Thai language
	SIP96X1-29927	Phone initiates call to voice mail on 2nd CA
	SIP96X1-22458	J129 - User does not receive forced log out message while the phone they are registered to is in a locked state and the user is active in the Admin Menu
	SIP96X1-31597	Seen twice - J129 reboots and generates coredump tContactManager when use extension with 250 contacts
	SIP96X1-31356	Oneway speech path after fake hold and resume (failover)
1-13719633022	SIP96X1-31856	Display issue when called picked up using FAC.
CCElite		
	SIP96X1-29804	[CCElite] Call transfer– Agent does not send the Avaya-User-Data and UI-Info data in the INVITE
	SIP96X1-29794	[CCElite] Call transfer - Phone does not display collect digit
	SIP96X1-30907	[CCElite] - Phone does not display Call Timer in half width mode
All Platforms		
1-13498522234	SIP96X1-30113	Jabra EHS headset does not recognize tones from phone causing the Jabra Base to remain off-hook when user terminates the call by depressing the headset button.
	SIP96X1-30248	transducer unexpectedly switches to Headset!
	SIP96X1-31369	J129/J169/J179 - No voice speech path after hold both phones and resume the call (SRTP call)
	SIP96X1-31033	Quick search appear and show empty after logout/login
Web Provisioning		
	SIP96X1-30771	J129 - Cannot back to provisioning screen 1 after enter to provisioning screen 2

Unresolved issues in J100 3.0.0.0.20

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®		
1-12671341066	SIP96X1-24119	ELD Rules not working. User is not able to dial 11 digit national calls from history.
	SIP96X1-17977	Configuration file includes SET_ENABLE_G729 2. User A and User B are in call. Then on User B, press "Transfer" softkey and dial to User C. During User B and User C are in call, User A holds its call. At that time, User B completes his transfer call to User C by pressing "Complete" softkey. The call is transferred successfully. When User A unholds the call, there is no speechpath.
	SIP96X1-20240	Phone B is configured with EC500 feature with cell phone number. Make call from phone A to phone B and answer the call on Cell phone. Press bridge soft key on phone B, then from phone B make unattended transfer to phone C. Answer the call on C -> No voice speech path between phone A and C.
	SIP96X1-23977	Deskphone is configured for FIPS mode and has a valid identity certificate. If an attempt is made to install a new certificate which is not FIPS-compliant, then the Deskphone does not display a "certificate rejected" message but does continue to use the original valid certificate.
	SIP96X1-23938	J129 Phone doesn't display "Non-AST/Fail-Over" icon after failover to non-AST environment when you are not on the idle screen and failover happens.
	SIP96X1-23905	J129 phone does not play ringing tone for second call when active call is emergency call
	SIP96X1-23883	J129 - Phone does not generate call log of Emergency call in Locked state when Emergency number was added in Contacts
	SIP96X1-23863	J129 - Phone does not always update sip proxies list after changing order of sip proxies in SMGR
	SIP96X1-23238	J129 - Phone displays conference icon and softkeys after other MDA user deactivates call-park feature
	SIP96X1-20779	J129 - Phone does not enable call-park feature when the feature is activated from another MDA user
	SIP96X1-20316	J129 - Server field ID display does not index to the right after clearing 8 characters

External ID	Internal ID	Issue Description
	SIP96X1-26755	J129 - Local call forward does not appear on Feature screen for using after failover to Audiocodes
	SIP96X1-28106	Phone does not display NewCall SK for making the second calls after completing failover to BSM in active call
	SIP96X1-28113	Guest Login: History LED keeps on lit, on primary user even after guest login logs out
	SIP96X1-28284	Navigation does not work for the first pressing in "EAP-MD5 Authentication failed" screen
	SIP96X1-28651	Unexpected Softkeys after Cancel consult with 3rd party
	SIP96X1-29823	J179 phone does not display added contact during limbo state
	SIP96X1-29855	Intermittent - Phone is stuck at PKCS12 installation screen during boot up in Wifi mode
	SIP96X1-29926	J129- One way speech path after holding/resuming the call with codec OPUS-WB20k and media encryption 3-srtp-aescm128-hmac80-unauth.
	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.
	SIP96X1-31276	Phone displays "No match found" when searching contact which created with only Last name on Exchange Server
	SIP96X1-32363	J179/J139 - Speaker LED is still ON when pressing cancel an audio push
	SIP96X1-34721	Phone displays incorrectly after recording audio.
	SIP96X1-34404	Speaker/ headset keep turning on when the call is dropped
	SIP96X1-34780	J139 phone display "Drop" soft key in Ad-hoc/ Meet me conference
	SIP96X1-35021	Contact is created with two numbers, the second number is E.164 format. The same number is configured in SMGR. Phone is configured for exact match. When an incoming call is made to the phone, the name from SMGR is shown instead of the local contact name.
	SIP96X1-31692	J139 - Call Forward Ringtone does not work.
IP Office		
	SIP96X1-23804	J129 contacts with IP Office: Sometimes 'New' softkey is displayed after adding 250 contacts.
	SIP96X1-33129	J139 - Push server got 500 error when perform a normal audio receive push with CCMS environment

External ID	Internal ID	Issue Description
	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.
	SIP96X1-33444	J1XX : "Reset to defaults" fails on J179 while registered to Branch IPO
	SIP96X1-31066	J179 - Headset call - make hold/unhold call goes to handsfree on speaker
CCElite		
	SIP96X1-31004	Phone does not display full AgentID when system language set to English and phone set to Arabic
	SIP96X1-32816	[CCElite] Call transfer - Phone displays collect digit, when drop call then phone still displays collect digit
	SIP96X1-33409	[Intermittent - CCElite] Phone doesn't display SO information on Agent Information Line
	SIP96X1-35017	[CCElite] VuStats' led is blinking incorrectly when phone send re-SUBSCRIBE and received NOTIFY event avaya-cm-cc-info
3PCC		
	SIP96X1-23559	J129 - User_Store - Phone is not making PUT request after receiving 404 in response of a GET query after adding first contact if HTTP Authentication is enabled in server and Authentication is ignored by user when it prompts. <i>Workaround: Edit contact again and perform Manual Backup</i>
	SIP96X1-23211	J129 - Glare handling for retransmitted INVITE and 407 with different nonce <i>Workaround: Change Timer T1</i>
	SIP96X1-23183	J129 - User_Store - If user changed a parameter that triggered a Backup. If the backup is in progress and the user logs out and new user is logging immediately may have their "restore" impacted and it may not work.
	SIP96X1-32258	Phone has no configuration (i.e. direct from factory or after a "Clear" operation) and the web interface is used to configure the SIP controller list and user information. The phone does not login when rebooted. <i>Workaround: Configure the SIP controller list and save the settings, then configure the SIP user information and save the settings.</i>
All Platforms		
	SIP96X1-22231	J129 - MIB browser displays value of "endptLANGINUSE" incorrectly

External ID	Internal ID	Issue Description
	SIP96X1-23890	J129 - "@" character is not supported for User ID, Contacts. If "@" character is configured, phone will ignore all the characters after "@" including "@" character. For CLI display, phone will not display name or number after "@" including "@" character.
	SIP96X1-21314	J129 - Phone does not allow contacts to be added as speed dial entries until PPM update is complete
	SIP96X1-23850	J129 - When downgrade fails, Upgrade info screen is blank
	SIP96X1-23791	J129 - 802.1x – Phone is not displaying 802.1x credential screen if phone receives EAP-Failure packet. This case happens only if credentials are changed in Radius Server during working environment. Workaround: Reboot the phone
	SIP96X1-20743	J129 - Phone reboots a second time after the user comments out the proxy address in settings file.
	SIP96X1-20372	J129 - Phone does not display SCEP notifications while it is downloading identity certificate from CA server
	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
	SIP96X1-30715	Phone display "Feature not available" while customizing line keys
	SIP96X1-30948	Phone displays number of BCA incorrectly after moving in customize key.
	SIP96X1-31080	Pressing key "0" for the first input in contact group name then press other key, other key is doubled
	SIP96X1-33023	Unable to gracefully recover phone when lost WIFI on active call
	SIP96X1-33403	Phone doesn't show "Restarting..." while coming back to main screen after setting file server from Admin Menu
	SIP96X1-35019	J169/J179 Phone switches speech path to speaker when pressing "Headset" button and then pressing "Redial" SK
	SIP96X1-33286	J100 - After press Restart SK, phone is not display Restarting Screen.
	SIP96X1-34870	When adding a new group in Contact groups, the "NAME" prompt is not translated to some languages.
Web User Interface		
	SIP96X1-30510	Web UI displays SIP User ID and Authentication User ID inconsistently after Guest User is logged out.
	SIP96X1-34938	Web UI displays wrong ethernet gateway when WIFISTAT is set to 2

External ID	Internal ID	Issue Description
	SIP96X1-34440	If a phone is on an active call and the Web UI is used at the same time to upgrade software, then voice path is lost.
	SIP96X1-34441	Phones do not reboot automatically after upgrading software via Web UI.
	SIP96X1-30908	Web UI - Need to clarify item Old Web Admin Password in Phone Admin Password menu
	SIP96X1-32880	Web UI – If only the ENABLE_3PCC_ENVIRONMENT value is changed, then it does not take affect. <i>Workaround: Change any other parameter to a non-default value at the same time.</i>

Appendix 1 – Supported Hardware

J100 SIP 3.0.0.0 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	Note
700512392	J129 IP PHONE	J129D01A	
700513638	J129 IP PHONE NO PWR SUPP		
700512969	J129 IP PHONE 3PCC W/O PWR SUPP	J129D01A	
700513639	J129 IP PHONE 3PCC W/CERT		
700513634	J169 IP PHONE NO PWR SUPP	J169D01A	Must use J100 1.5 or later software.
700513635	J169 IP PHONE GSA	J169D01A	Must use J100 1.5 or later software.
700513636	J169 IP PHONE 3PCC	J169D01A	Must use J100 2.0 or later software.
700513569	J179 IP PHONE NO PWR SUPP	J179D02A	Must use J100 1.5 or later software.
700513629	J179 IP PHONE GSA	J179D02A	Must use J100 1.5 or later software.
700513630	J179 IP PHONE 3PCC	J179D02A	Must use J100 2.0 or later software.
700513916	J139 IP PHONE	J139D01A	Must use J100 3.0 or later software.
700513917	J139 IP PHONE 3PCC	J139D01A	Must use J100 3.0 or later software.
700513918	J139 IP PHONE GSA	J139D01A	Must use J100 3.0 or later software.

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

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=P1661

New parameters.

There are no new parameters.

Changed parameters.

```
## 1. J100 SIP R3.0.0.0 and later (J129, J139, J169, J179):
## a. J139 is supported by J100 SIP R3.0.0.0. Mark the following as supported by J139 in J100 SIP R3.0.0.0: BAKLIGHTOFF,
ENABLE_MLPP, MLPP_NET_DOMAIN, MLPP_MAX_PREC_LEVEL,
## ENABLE_PRECEDENCE_SOFTKEY, DSCPAUD_FO, DSCPAUD_FL, DSCPAUD_IM, DSCPAUD_PR, DSCPMGMT, PHY1STAT,
PHY2STAT, PHY2_AUTOMDIX_ENABLED, HANDSET_PROFILE_DEFAULT,
## HANDSET_PROFILE_NAMES, ENABLE_IPOFFICE, MEDIA_PRESERVATION, PRESERVED_CALL_DURATION,
ENABLE_SHOW_EMERG_SK, ENABLE_SHOW_EMERG_SK_UNREG, LANGUAGES, SYSTEM_LANGUAGE,
## SIP_CONTROLLER_LIST, SIMULTANEOUS_REGISTRATIONS, SIPREGPROXYPOLICY, TRUSTCERTS, SPEAKERSTAT, L2Q,
L2QVLAN, L2QAUD, L2QSIG, VLANSEP, VLANSEPMODE, PHY2VLAN, PHY2PRIO,
## PHY2TAGS, DSCPAUD, DSCPSIG, WBCSTAT, QLEVEL_MIN, DHCPSTD, VLANTEST, REUSETIME, DNSSRV, DOMAIN,
ENABLE_UDP_TRANSPORT, SIP_CONTROLLER_LIST_2, SIPDOMAIN,
## ENABLE_PPM_SOURCED_SIPPROXYSRV, ENABLE_AVAYA_ENVIRONMENT, DIALPLAN, PHNNUMOFS, SNTPSRV,
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, MUTE_ON_REMOTE_OFF_HOOK,
## SDPCAPNEG, ENFORCE_SIPS_URI, 100REL_SUPPORT, PLAY_TONE_UNTIL RTP, LOCALLY_ENFORCE_PRIVACY_HEADER,
ENABLE_SIP_USER_ID, ENABLE_STRICT_USER_VALIDATION,
## BRANDING_VOLUME, ENABLE_OOD_MSG_TLS_ONLY, PHNEMERGNUM, PHNMOREEMERGNUMS, LOG_DIALED_DIGITS,
MAX_TRUSTCERTS, ENABLE_PUBLIC_CA_CERTS, TLSSRV, FQDN_IP_MAP,
## SERVER_CERT_RECHECK_HOURS, CERT_WARNING_DAYS, DELETE_MY_CERT, TLS_VERSION, HTTPPROXY,
HTTPPEXCEPTIONDOMAINS, MYCERTURL, MYCERTCN, MYCERTDN, MYCERTCAID,
## MYCERTRENEW, SCEPPASSWORD, PKCS12URL, PKCS12_PASSWD_RETRY, DOT1XSTAT, DOT1X, DOT1XEAPS, OCSP_ENABLED,
OCSP_ACCEPT_UNK, OCSP_NONCE, OCSP_URI, OCSP_URI_PREF,
## OCSP_TRUSTCERTS, OCSP_HASH_ALGORITHM, OCSP_USE_CACHE, OCSP_CACHE_EXPIRY, ENABLE_WEBSERVER,
WEBSERVER_ON_HTTP, WEB_HTTP_PORT, WEB_HTTPS_PORT, FORCE_WEB_ADMIN_PASSWORD,
## ENABLE_PHONE_LOCK, PHONE_LOCK_IDLETIME, ENABLE_G711A, ENABLE_G711U, ENABLE_G722, ENABLE_G726,
G726_PAYLOAD_TYPE, ENABLE_G729, ENABLE_OPUS, OPUS_PAYLOAD_TYPE,
## SEND_DTMF_TYPE, DTMF_PAYLOAD_TYPE, SYMMETRIC_RTP, MATCHTYPE, PHNMUTEALERT_BLOCK, PROCSTAT, PROCPSWD,
ADMIN_PASSWORD, ADMIN_LOGIN_ATTEMPT_ALLOWED, ADMIN_LOGIN_LOCKED_TIME,
## SNMPSTRING, SNMPADD, LLDP_ENABLED, LOGSRV, LOCAL_LOG_LEVEL, LOG_CATEGORY, SSH_ALLOWED,
SSH_BANNER_FILE, SSH_IDLE_TIMEOUT, EASG_SITE_CERTS, EASG_SITE_AUTH_FACTOR,
## CERT_WARNING_DAYS_EASG, AUTHCTRLSTAT, SLMSTAT, SLMCAP, SLMCTRL, SLMPEF, SLMPORT, SLMSPRV, ENHDIALSTAT,
PHNCC, PHNDPLENGTH, PHNIC, PHNLD, PHNLDLENGTH, PHNOL,
## AGCHAND, AGCHAD, AGCSPKR, AUDIOENV, HTTPSRV, HTTPDIR, HTTPPORT, AUTH, TLSSRV, TLSDIR, TLSPORT, DES_STAT,
RTCPMON, RTCPMONPORT, RTCPMONPERIOD, ICMPDU, ICMPRED,
## BRURI, MSGNUM, IPV6STAT, SIGNALING_ADDR_MODE, MEDIA_NEG_PREFERENCE, MEDIA_ADDR_MODE, IPV6DADXMITS,
CALLFWDDELAY, CALLFWDADDR, ENABLE_AUTO_ANSWER_SUPPORT,
## AUTO_ANSWER_MUTE_ENABLE, ENABLE_DND, ENABLE_DND_PRIORITY_OVER_CFU_CFB, HOLD_REMINDER_TIMER,
CONFERENCE_TYPE, RTCP_XR, MEDIAENCRYPTION, ENCRYPT_SRTCP,
## SUBSCRIBE_LIST_NON_AVAYA, ENABLE_3PCC_ENVIRONMENT, USER_STORE_URI, PROVIDE_OPTIONS_SCREEN,
PROVIDE_NETWORKINFO_SCREEN, PROVIDE_LOGOUT, DISPLAY_SSL_VERSION,
## ENABLE_CALL_LOG, ENABLE_REDIAL, CONTACT_NAME_FORMAT, DAYLIGHT_SAVING_SETTING_MODE, RTP_PORT_LOW,
RTP_PORT_RANGE, SIG_PORT_LOW, SIG_PORT_RANGE, INGRESS_DTMF_VOL_LEVEL,
## 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,
## AUDIOTH, AUDIOTH, ENABLE_BLIND_TRANSFER, PROVIDE_KEY_REPEAT_DELAY, HEADSET_PROFILE_DEFAULT,
HEADSET_PROFILE_NAMES, HEADSYS, ENTRYNAME, ENABLE_RECORDING,
```

```
## RECORDINGTONE_INTERVAL, RECORDINGTONE_VOLUME, SCREENSAVERON, BACKLIGHT_SELECTABLE, WARNING_FILE,
ELD_SYNSUM, RINGTONESTYLE, PROVIDE_CF_RINGTONE,
## CONF_TRANS_ON_PRIMARY_APPR, AUTO_SELECT_ANY_IDLE_APPR, LANGLARGEFONT, AUDASYS
## b. As J139 does not support Wi-Fi/BT module the following parameters were marked as not supported: WIFISTAT,
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_L2QUAD,
WLAN_L2QSIG, WLAN_DSCPAUD, WLAN_DSCPSIG.
## c. Exchange Integration parameters are not supported: 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
## d. Unsupported customization features: RINGTONES

##### PUSH INTERFACE SETTINGS #####
##
## TPLIST (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 TPLIST 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
## 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
## SET TPLIST 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
## 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
## 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
## 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.
## 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**
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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