Avaya J100 Series SIP Release 4.0.2.0 Readme

This file is the Readme for the Avaya J100 Series SIP Release 4.0.2.0 software (J100 SIP 4.0.2.0). This file describes the contents of the July 2019 (**4.0.2.0.8**) release software distribution package.

J100 SIP 4.0.2.0 software is supported on the Avaya J129, J139, J169, and J179 IP Phones used with Avaya Aura®, Avaya IP Office™, and select OpenSIP platforms. J100 SIP 4.0.2.0 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 4.0.2 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)
- IP Office™ 11.0 or later for J129/J169/J179
- 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
- Avaya Aura® Call Center Elite 7.0.1.0[1], 7.1.0.0[1], 8.0.0.0[1]
- OpenSIP Platforms
  - Broadsoft Broadworks R22.0
  - Zang Office R1.0
  - Asterisk R16
  - 3CX R15.5
  - FreeSWITCH 1.8.5

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

- o   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.

New Features in J100 SIP 4.0.2

Avaya J100 Series SIP Release 4.0.2 contains the following new features.

New with this release	Description
Support for Crisis Alert Receive on Avaya Aura®	The ability to receive a Crisis Alert has been added to the J169/J179 IP Phones. Crisis Alert Receive is NOT supported on the J129/J139 IP Phone. **Note:** This functionality also requires Avaya Aura® 8.1.0.
Support for No Hold Conference on Avaya Aura®	The ability to use No Hold Conference has been added. **Note:** This functionality also requires Avaya Aura® 8.1.0.
Support for Busy Indicator on Avaya Aura®	The ability to use Busy Indicator has been added. **Note:** This functionality is supported on Avaya Aura® 7.1.3 but not Avaya Aura® 8.0/8.1.0. It is scheduled to be available with Avaya Aura® 8.1.1.
Additional capabilities for 3CX	Support has been added for TR-87 and Remote Provisioning Server (RPS).
Broadsoft – BLF Transfer	Allow a user to transfer a call to another user via the BLF key.
Web provisioning enhancements	Improvements to the web provisioning interface for pre-configuring keys.
Basic interworking with Netsapiens	The compatible OpenSIP vendors has been extended to include Netsapiens.

Documentation for J100 SIP 4.0.2.0

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".

J100 SIP 4.0.2.0 (4.0.2.0.8) Package Content

The J100 SIP 4.0.2.0 package (J100-IPT-SIP-R4_0_2_0-062519.zip) contains all the files necessary to upgrade Avaya new or previously installed Avaya J129/J139J169/J179 IP Phones to the J100 SIP 4.0.2.0 software.

- FW_S_J129_R4_0_2_0_8.bin – application binary file for J129
- FW_S_J139_R4_0_2_0_8.bin – application binary file for J139
- FW_S_J169_R4_0_2_0_8.bin – application binary file for J169
- FW_S_J179_R4_0_2_0_8.bin – application binary file for J179
- FW_JEM24_R1_0_1_0_8.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_J169_J179_Arabic.xml
    - Mlf_J169_J179_BrazilianPortuguese.xml
    - Mlf_J169_J179_CanadianFrench.xml

- o Mlf_J169_J179_CastilianSpanish.xml
- o Mlf_J169_J179_Chinese.xml
- o Mlf_J169_J179_Dutch.xml
- o Mlf_J169_J179_English.xml
- o Mlf_J169_J179_German.xml
- o Mlf_J169_J179_Hebrew.xml
- o Mlf_J169_J179_Italian.xml
- o Mlf_J169_J179_Japanese.xml
- o Mlf_J169_J179_Korean.xml
- o Mlf_J169_J179_LatinAmericanSpanish.xml
- o Mlf_J169_J179_ParisianFrench.xml
- o Mlf_J169_J179_Polish.xml
- o Mlf_J169_J179_Russian.xml
- o Mlf_J169_J179_Thai.xml
- o Mlf_J169_J179_Traditional_Chinese.xml
- o Mlf_J169_J179_Turkish.xml
- Eight extended Korean ring tone files:
  - o KoreanRT1.xml
  - o KoreanRT2.xml
  - o KoreanRT3.xml
  - o KoreanRT4.xml
  - o KoreanRT5.xml
  - o KoreanRT6.xml
  - o KoreanRT7.xml
  - o KoreanRT8.xml
- AvayaLanguageTool_SIP.xlsm
- One certificate file:
  - o 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
- 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 **.**

Advisements with J100 SIP 4.0.2.0 software

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

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".

Avaya Aura® 8.0.1 provides native support of the J129/J169/J179 IP Phones. The J139 should be administered as J169.

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: 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.

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.	

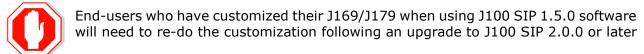| Avaya Aura® configuration	What to expect when a user moves/adds a feature / auto dial to a different line location on the phone	What to expect when a user deletes a feature / auto dial using customization.	What to expect when user does a Customization -> Restore
Avaya Aura® 7.1.3.2 or earlier (J100 aliased as a 9600)	Change will not show in SMGR (similar behavior as SIP 9600-series)	No change in SMGR.	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.
Avaya Aura® 7.1.3.3 (J100 aliased as a 9600)			
Avaya Aura® 8.0.1 or later (J100 native)		Feature favorite flag will be unchecked in SMGR	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.

Avaya Aura® – Busy Indicator – does not work on Aura 8.x

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.x when the J169/J179 are natively supported. It is expected that this will be resolved with Avaya Aura® 8.1.1.

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/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](#)

J129/J139/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.

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 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.

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

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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.

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 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.

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

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[2] J129 does not support a headset

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.

"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.
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)

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
- 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:
- Push API
- WML Browser

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
- 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 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
- 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
- WML Browser

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

The J129/J139J169/J179 are supported with Broadsoft Broadworks, Zang Office, Asterisk, and 3CX. 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.

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.

J100 4.0.2.0 Resolved Issues (since J100 4.0.1.0)

The following table includes issues which are resolved with this release of software compared to J100 4.0.1.0.11

External ID	Internal ID	Issue Description
		Avaya Aura®
	SIP96X1-31692	J139 – Call Forward Ringtone does not work.
1-14063335506	SIP96X1-52871	J169/179 are using same FNU code for hold/transfer/conf
1-14701490462	SIP96X1-57660	Cannot transfer in shared control mode - only phone reboot solves issue
		IP Office
	SIP96X1-52128	[BT CCMS]-Phone can't answer automatically the second incoming call after drop the first call
1-14656038012	SIP96X1-57292	J179 Shows wrong time with IPO
1-14097689272	SIP96X1-51144	J1xx Intermittently not ring heard when call transferred through Hunt Group in IPO
1-14516730532	SIP96X1-53960	EXCHANGE:J100 RTP delay from phone in IPO
		CCElite
1-14755106602	SIP96X1-60179	J179 phone does not show VDN info when CM Agent has auto answer enabled
		Open SIP
	SIP96X1-53985	Pre-configuration BLF phone key is still displayed after removing on web
	SIP96X1-53496	New BLF PHONEKEY displays with Number instead of Name
	SIP96X1-53140	Phone change audio path from headset to speaker when get incoming call alerting from PUSH
	SIP96X1-52990	3CX – After Ending BLF call, custom label is reset
	SIP96X1-51758	Broadsoft: [rare] Phone stuck with Limited Service icon (!), XSI event channel down , not recovering work-around: restart phone
	SIP96X1-51249	[Broadsoft] No voice speech path when far-end holds the call and then user activates call retrieve feature from different phone.
		All Platforms
	SIP96X1-38760	Phone firmware does not display on SLA Server after phone get discovered
		Web User Interface
	SIP96X1-53663	Some Pre-config keys are not immediately displayed when setting by Web *work-around: reboot the phone*
	SIP96X1-53767	Web Server does not respond after "SIP User Logout" is changed from Web UI
		Bluetooth

External ID	Internal ID	Issue Description
	SIP96X1-53027	Voice is not clear when using Plantronics PLT_VL60 headset
	SIP96X1-53792	BT headset going off-hook(feature button) while BCA alerting freezes phone UI

Unresolved issues in J100 4.0.2.0

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®
	SIP96X1-26755	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
	SIP96X1-30262	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.
	SIP96X1-36019	The pop-up entry doesn't disappear after using "Dial Intercom" to dial
	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

External ID	Internal ID	Issue Description
	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 *work-around: press "Select" SK*
	SIP96X1-61417	Phone is shared control generates tSharedControlM coredump when cancel login user.
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
	SIP96X1-53328	[BT-CCMS]-Phone cannot hold the call using BT headset when phone receive call on BCA line *work-around: Press Hold SK on phone*
CCElite		
	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	J169/179 – CCElite - 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*
	SIP96X1-60280	Phone cannot reactivates Assist feature when join in conference.
Open SIP		
	SIP96X1-42012	L129 BT headset does not hear ringing tone after when ignore one of incoming calls (RTX 209114)
	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

External ID	Internal ID	Issue Description
	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
	SIP96X1-53711	[Asterisk] phone crashes after consultative transferring by pressing BLF line key *Work-around: use keypad to enter number instead of using line key.*
	SIP96X1-53909	Red led still glow after replacing feature with application by Web pre-configuration
	SIP96X1-56316	[Freeswitch 3PCC] Phone cannot detect the state of BLF phone when using an FQDN for the sip proxy list.
	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
	SIP96X1-60451	3PCC 3CX PnP Failed to Barge-in BLF call
	SIP96X1-61262	Intermittent - 3PCC - Broadsoft - Cannot Join conference after hold and resume 3rd party
	SIP96X1-61347	The toogle control icons in Shared should show the current state.
	SIP96X1-61421	Cannot add new BLF/Autodial key after removing BLF keys on Web *work-around: reset phone to default*
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
	SIP96X1-30715	Phone display "Feature not available" while customizing line keys
	SIP96X1-30948	Phone displays number of BCA incorrectly after moving in customize key.
	SIP96X1-57796	AUDASYS parameter works incorrectly *work-around: adjust volume to 0 on rocker*

External ID	Internal ID	Issue Description
	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
	SIP96X1-34870	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"
	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*
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*

External ID	Internal ID	Issue Description
	SIP96X1-57487	Phone doesn't enable WEB server automatically in 3CX mode *work-around:enable it from settings file*
	SIP96X1-59350	can add certificate but cannot delete if the certificate's name has a space on Web browser.
	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*
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.
	SIP96X1-59261	After Headset is restarted (after upgrade) or plugged out/in it doesn't auto-reconnect to J179 phone
	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*
Button Module (JEM24 and JBM24)		
	SIP96X1-43092	JEM24: Power cycle required message dissappears unexpectedly
	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

Appendix 1 – Supported Hardware and Minimum Software Release

J100 SIP 4.0.2.0 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	
700512969	J129 IP PHONE 3PCC W/O PWR SUPP	J129D01A	700513639
700513639	J129 IP PHONE 3PCC W/CERT	J129D02A	
700513916	J139 IP PHONE	J139D01A	
700513917	J139 IP PHONE 3PCC	J139D01A	
700513918	J139 IP PHONE GSA	J139D01A	
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	

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	3.0.0.1.6	N/A
J139D01A	01 to 06	3.0.0.0.20	N/A
J139D01A	07	3.0.0.1.6	N/A
J169D01A	01 to 05	1.5.0.0.15	6.7.0.02
J169D01A	06	3.0.0.1.6	6.8.0.03
J179D02A	01 to 04	1.5.0.0.15	6.7.0.02
J179D02A	05	3.0.0.1.6	6.8.0.03

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

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.

###################### 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
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:
J100 SIP R4.0.2.0 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
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
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
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

SET NAT_SIGNALING_KEEPALIVE_OVERRIDE_SEC 20

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 (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 (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 (J129 and J139 do not support this parameter)
SET SHORTCUT_ACTION_AUTODIAL 2
##
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.
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 (J129 and J139 do not support this parameter)
SET CALL_PARK_FAC #95
##
UPDATE_DIALED_NUMBER_ON_ANSWER specifies whether displayed dialed number is updated or not based the number provided in 200 OK after an answer.
Value Operation
0   Displayed dialed number is not updated based on 200 OK received after answer (default)
1   Displayed dialed number is updated based on the number provided in 200 OK after answer.
This parameter is supported by:
J100 SIP R4.0.2.0 and later
96x1 SIP R7.1.6.0 and later
SET UPDATE_DIALED_NUMBER_ON_ANSWER 1
##
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)
SET WAIT_FOR_CALL_OPERATION_RESPONSE 2

Changed parameters.

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
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

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 only for 3PCC environment when 3PCC_SERVER_MODE = 0 (generic) or 1 (BroadSoft)
SET ENABLE_DND 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)
SET WAIT_FOR_CALL_OPERATION_RESPONSE 2

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