Avaya J129 IP Phone and J100 Series SIP 1.1.0.0 Software

This file is the Readme document for the Avaya J100 Series SIP Release 1.1.0.0 software. This file describes the contents of the March 2017 release software distribution package.

Avaya J100 Series SIP Release 1.1.0.0 software is supported on the J129 IP Phone to be used with 3PCC (3rd party call control) platform, Avaya Aura® and IP Office™.

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.

Call Platform Compatibility

The Avaya J129 IP Phone using Avaya IP Phone SIP Release 1.1.0.0 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
- IP Office™ 10.0 SP2
- 3PCC (3rd party call control) Platform
    - Broadsoft Broadworks R21SP1
    - Zang Office R1.0
    - Edgewater Network device (Edgemarc 4550).

Note: The Avaya J100 Series SIP Release 1.1.0.0 software is generally available controlled introduction (GA-CI) for the J129 IP Phone deployed in the 3PCC environment.

New Features supported in J129 IP Phones with SIP 1.1.0.0 software

- Local contacts supported with IP Office. J129 will use the IP Office™ hosted Personal Directory as local contacts. J129 will backup/restore contacts with IP Office whenever contact is modified (on IP Office™ or J129 locally). *Requires minimum IP Office™ Release R10.1*
- Support of 3PCC Platform
  - Broadsoft Broadworks R21SP1
  - Zang Office R1.0
  - Edgewater Network device (Edgemarc 4550)
- MACADDR based setting file provisioning

Feature support in J129 IP Phones with SIP 1.1.0.0 software for 3PCC Platform

Below are the feature highlights for the J129 IP phone with SIP 1.1.0.0 software for 3PCC platform:

- One line phone, supports two concurrent calls
- 3 Context Sensitive Soft Keys
- Handsfree speaker and handset
- Codec support G.711, G.722, G.726, G.729, Opus
- Recent Call Log (100 entries)
- Contact List (250 entries)
- Built in volume boost control in Handset for Hearing Impaired
- Backup/restore of User Data:
  - Local Contacts
  - Feature settings (Call Forward, DND, Auto Answer)
  - Personal Settings (language, ring type, time zone, time format, date format)
- Local Features for 3PCC
  - Hold Reminder
  - Local Conference
  - Call Forward
  - Auto Answer
  - DND
  - Speed Dial
- Network (Server based) features
  - Call Hold/Unhold
  - Transfer (Unattended/Attended Transfer)
  - Call Waiting
  - Call Park/Unpark
  - Call Pickup
  - Meet-me Conference
  - Network Conference (3 party)(not applicable for Zang Office)
  - Voice Mail (VM Submit, VM Retrieve, Message Wait Indication)
- Mute Key with Mute Alerting
- Dual 10/100 Ethernet ports to support co-located PC

- Power over Ethernet Class 1
- HTTP/HTTPS
- VLAN
- LLDP
- Advanced diagnostics capability

Documentation for J100 Series SIP 1.1.0.0

The following documentation has been updated for this release:

- Using Avaya J129 IP Phone
- Avaya J129 IP Phone Quick Reference
- Installing and Administering Avaya J129 IP Phone
- IP Office™ Platform 10.0 - SIP Telephone Installation Notes
- Using Avaya J129 IP Phone in third-party call control setup
- Installing and Administering Avaya J129 IP Phone in third-party call control setup
- Avaya J129 IP Phone in third-party call control setup Quick Reference

J100 Series SIP 1.1.0.0 (1.1.0.0.15) Package Content

The J100 Series SIP 1.1.0.0 software package (J100-IPT-SIP-R1_1_0_0-022417.zip) contains all the files necessary to install the J129 IP phone.

- FW_S_J129_R1_1_0_0_15.bin – The SIP 1.1.0.0 application binary file.
- J100Supgrade.txt – This file is downloaded by the IP phone and instructs the phone on how to upgrade to this version of software.
- Fourteen 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_Russian.xml
- Eight extended Korean ring tone files:
    - KoreanRT1.xml
    - KoreanRT2.xml
    - KoreanRT3.xml
    - KoreanRT4.xml
    - KoreanRT5.xml
    - KoreanRT6.xml
    - KoreanRT7.xml
    - KoreanRT8.xml
- Certificate file: av_prca_pem_2033.txt – Avaya Product Root CA certificate with an expiration date of 2033
- release.xml
- SNMP MIB file: Avaya-J100IpPhone-MIB.mib
- A “signatures” subdirectory containing signature files and a certificate file.

System specific parameters should be entered into the 46xxsettings.txt file which is available for separate download at http://support.avaya.com. Refer to Appendix 3 for new or changed parameters with this release of software.

Advisements with J100 Series SIP 1.1.0.0 software for Avaya Aura® and IP Office™

Deploying the J129 with IP Office™ natively

The J129 is supported with IP Office™ both natively and in a failover in an Avaya Aura Centralized Branch configuration. IP phones that are supported with IP Office™ traditionally have their configuration files deployed from the IP Office™ itself. However, with IP Office™ 10.0 SP2 the configuration file (settings file) must be deployed from a file server (HTTP or HTTPS). In a future release of IP Office™ the settings file will be deployed from the IP Office™ itself.

Presence

The J129 does not display presence in an Avaya Aura® network. The J129 publishes presence information for other clients that support viewing presence.

VLAN separation

The J129 supports Full VLAN separation; it does not provide Partial VLAN separation. This is controlled via a parameter VLANSEPMODE. Avaya recommends that voice and data traffic be separated by VLANs and that voice traffic has its own VLAN.

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. The use of Fully Qualified Doman Names (FQDN) is not supported. This applies to all sources of the SIP_CONTROLLER_LIST parameter which includes DHCP, LLDP, and the 46xxsettings.txt file. It is not recommended that the SIP_CONTROLLER_LIST be manually configured on the phone but if required then the administrator must first go into the Admin menu and change the Proxy policy from "Auto" to "Manual".

Enhanced diagnostics

The J129 offers enhanced diagnostic capabilities such as port mirroring, sending the phone report to a server and the ability to change the SSH capabilities.

Survivability – Failover support

The J129 offers the same failover support as the 9600 Series with Deskphone SIP software product line when connected to Avaya Aura with SM, BSM, or IP Office as a backup server. With the support for connectivity in a native IP Office™ environment survivability is supported with two IP Office™ SIP proxies. In this case SIPREGPROXYPOLICY must be set to Alternate.

Features not supported on the J129 Phone

The following features are not supported by the J129: Exchange integration, WML browser, Contact Center (CCElite), URI dialing, Bridge Call Appearance (except MDA), simultaneous display of caller name and number, redial by list, conference roster list, missed call filtering, displaying presence, Push feature, shared control with 1XC communicator/ACW, downloadable ringtones, Favorites, Personalize labels, the following AST features: analog bridge appearance, Audix record, 3rd party MWI, auto intercom, autodial buttons, busy indicator, call pick-up, enhanced call forward, calling party name unblock, dial intercom, directed pick-up, exclusion, extended pick-up, Hunt group busy, limit call, priority calls, team buttons, whisper page.

Advisements with J100 Series SIP 1.1.0.0 software for 3PCC Platform

Deploying the J129 in 3PCC Platform

The J129 is supported with Broadsoft Broadworks R21SP1 and Zang Office R1.0. 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 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.

SIP Proxy configuration

Avaya recommends that the J129's SIP Proxy be configured via the settings file. Optionally, the SIP proxy may be configured manually through the phone's Admin menu. Configuring the SIP proxy manually requires the administrator to first go into the Admin menu and change the Proxy policy from "Auto" to "Manual".

SIP User Login Credentials configuration

J129 offers SIP User Login credentials via settings file for auto login. Optionally, the SIP User Credentials can be entered manually by user if not provided through settings file.

Enhanced diagnostics

The J129 offers enhanced diagnostic capabilities such as port mirroring, sending the phone report to a server and the ability to change the SSH capabilities.

Features not supported on the J129 Phone 3PCC Platform

The following features are not supported by the J129 3PCC Version: Presence, Exchange integration, WML browser, URI dialing, Bridge Call Appearance, MDA, simultaneous display of caller name and number, redial by list, conference roster list, missed call filtering, displaying presence, Push feature, shared control with 1XC communicator/ACW, downloadable ringtones, Favorites, Personalize labels, the following AST features: analog bridge appearance, Audix record, 3rd party MWI, auto intercom, autodial buttons, busy indicator, enhanced call forward, calling party name unblock, dial intercom, directed pick-up, exclusion, extended pick-up, Hunt group busy, limit call, priority calls, team buttons, whisper page.

Unresolved issues in J100 Series SIP 1.1.0.0 software

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®
	SIP96X1-23938	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.
	SIP96X1-23936	After entering Login credentials, sometimes there is no button effect for 8-10 seconds. It is observed once any other user logouts having 250 contacts and immediately other user tries to login.
	SIP96X1-23905	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
	SIP96X1-23878	"Emerg" softkey is displayed on "Idle logged out" screen when SIP_CONTROLLER_LIST is empty and PHNEMERGNUM is not configured
	SIP96X1-23863	J129 – Phone does not always update sip proxies list after changing order of sip proxies in SMGR
	SIP96X1-23676	call-bsyda feature fails on attempt to activate it
	SIP96X1-23508	J129 – "Lock unavailable when phone is in use" is not translated when local language is changed during active call.
	SIP96X1-23238	J129 - Phone displays conference icon and softkeys after other MDA user deactivates call-park feature
	SIP96X1-23199	J129 –Phone re-displays unpark activation state although phone already ended call after having conference with unparked call and other number.
	SIP96X1-23169	"Auto Answer": feature is working while user in Admin menu page and cannot switch the audio path or cannot drop the call. Workaround: Exit from Admin menu and perform the action on active call
	SIP96X1-22629	Occasionally the phone does not display IP Address in Admin mode when provisioning. Workaround: The correct IP is visible via the "Network Information" screen
	SIP96X1-22458	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
	SIP96X1-22229	Bridge on function does not work with E164 MDA extension phone
	SIP96X1-20779	Phone does not enable call-park feature when the feature is activated from another MDA user

External ID	Internal ID	Issue Description
	SIP96X1-20771	Phone fails to control local conference after attempting to add another participant while failed over to AudiCodes with OPUS.
	SIP96X1-20316	Server field ID display does not index to the right after clearing 8 characters
	SIP96X1-19647	The Call forward icon is not shown on the phone when there is a destination is configured for 3rd party Call forwarding
	SIP96X1-21738	J129: After failover/failback buttons [NewCall] and [Add] don't appear, icon [failover] doesn't disappear.
	SIP96X1-23792	Phone doesn't allow to add new contact after failover to non-AST environment (Teldat)
3PCC		
	SIP96X1-23840	J129 Provisioning through settings file– Phone remains in Locked state if User A locks the phone and after restart of phone, User B auto-login through settings file. In this case, User can unlock the phone using User B login password.
	SIP96X1-23559	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. Workaround: Edit contact again and perform Manual Backup
	SIP96X1-23211	Glare handling for retransmitted INVITE and 407 with different nonce Workaround: Change Timer T1
	SIP96X1-23183	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.
	SIP96X1-21314	Phone does not allow contacts to be added as speed dial entries until PPM update is complete
	SIP96X1-21296	The LastName field is always displayed when user inserts a symbol in the FirstName field in Contact Search screen
	SIP96X1-21275	Phone displays incorrect icon on UI for invalid number
	SIP96X1-23846	Localization not supported for few strings related to User data backup/restore.
IP Office™		
	SIP96X1-23804	J129 contacts with IP Office: Sometimes 'New' softkey is displayed after adding 250 contacts.
All Platforms		
	SIP96X1-23945	MAX_TRUSTCERTS parameter in not supported in SNMP MIB
	SIP96X1-23890	"@" 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.

External ID	Internal ID	Issue Description
	SIP96X1-23850	When downgrade fails, Upgrade info screen is blank
	SIP96X1-23802	802.1x - Although phone receives EAP-SUCCESS packet, phone is displayed message "Waiting for 802.1X authentication". Workaround: Reboot the phone
	SIP96X1-23791	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
	SIP96X1-23668	J129 – Phone doesn't download J100Supgrade.txt when setting SIG is 1 in DHCP. Workaround: Provision J100 via DHCP-242 with SIG 0
	SIP96X1-23611	J129: OPUS - Audio path is not clear during conference.
	SIP96X1-23210	J129: Speed dial is working while user is in active call.
	SIP96X1-23102	AUDASYS parameter is not supported
	SIP96X1-23058	Phone always displays *Generating report* after pressing Back button from Debug menu. Workaround: Press Back button 3-4 times to exit from Debug menu
	SIP96X1-22872	J129 – Phone displays wrong details in missed call log. Workaround: Go to another screen and back to Recents screen then the detail call log is displayed correctly.
	SIP96X1-22234	Intermittently phone may become stuck at restarting screen after manual clear or reboot Recovery Path: Plugging out the cable and plug-in again
	SIP96X1-22231	MIB browser displays value of "endptLANGINUSE" incorrectly
	SIP96X1-21775	Korean language is not labeled in Korean if phone language is Hebrew
	SIP96X1-21088	Phone does not display 802.1X Authentication failed screen after receiving EAP-Failure frame from switch
	SIP96X1-21210 SIP96X1-21081	Phone does not send EAPOL Start when setting DOT1XSTAT to 2 and DOT1XEAP to TLS in 46xxsetting file. Phone displays "Authentication failed." and pauses after receiving 46xxsettings.txt file.
	SIP96X1-20743	Phone reboots a second time after the user comments out the proxy address in settings file.
	SIP96X1-20372	Phone does not display SCEP notifications while it is downloading identity certificate from CA server

Resolved issues in J100 Series SIP 1.1.0.0 software

The following table includes isssues which are resolved with this release of software compared to SIP 1.0.0.0.43.

External ID	Internal ID	Issue Description
		Avaya Aura®
	SIP96X1- 22315	J129 – Phone is stuck at Starting screen forever when disable - enable Ethernet port after rebooted.
	SIP96X1- 22301	J129 – Phone gets stuck at Waiting for DHCP screen forever after phone got invalid parameter then clear phone setting again
	SIP96X1- 22269	Both users logged out while pulling of extension during active call.
	SIP96X1- 22255	Phone will not lock automatically if the PHONE_LOCK_IDLETIME expires before the phone has completed an automatic login
	SIP96X1- 22203	Speaker shouldn't be handled from admin password view
	SIP96X1- 21896	Soft keys are changed unexpectedly when digit is dialed "provisioning server invitation"dialogue box is shown
	SIP96X1- 21881	J129 – Zero Touch Provisioning – IP Address input format is inconsistent between Provisioning screen and other screens
	SIP96X1- 21794	DHCPLEASERENEW displays as 0 in the MIB browser
	SIP96X1- 21774	J129 –Phone re-displays Contacts Detail Screen and reboot after deleting that contact
	SIP96X1- 21744	MATCHTYPE parameter does not work as expected
	SIP96X1- 21611	J129: When phone prompts for "configure provisioning" when the "Phone" hard button is pressed and phone goes to an erroneous screen
	SIP96X1- 21557	J129 got stuck in "Waiting for DHCP..." after reset to default and visiting Admin menu to confirm valid IP address
	SIP96X1- 21552	Wrong message is displayed on Mercury and 96x1 when Identity certificate expires
	SIP96X1- 21341	Phone crash during multiple reboot
	SIP96X1- 21277	Not able to make a second call in a particular scenario with automatic dialing configured
	SIP96X1- 21274	Phone always displays "Dial or select Party" message on locked screen
	SIP96X1- 21251	Transfer upon Hang-Up doesn't work on J129 phones
	SIP96X1- 21219	Alphanumeric character is not replaced by an asterisk after 1 second for 802.1X Password Credentials
	SIP96X1- 21082	J129 with IPO: After FO-FB user is not able to make a new call until the existing preserved call is disconnected
	SIP96X1- 21077	J129 Personal ringtone does not survive reboot
	SIP96X1- 21043	Softkey 'Save' is displayed in Settings submenus even when pending changes are canceled
	SIP96X1- 20961	J129 - Redial works even when both CAs are busy

External ID	Internal ID	Issue Description
	SIP96X1- 20958	J129 – Phone still displays main menu after pressing speaker to redial.
	SIP96X1- 20935	When pressing voicemail key on phone, the screen is stuck in same screen
	SIP96X1- 20927	J129 with IPO: Phone momentarily displays limited service icon during volume adjustment
	SIP96X1- 20921	Phone rebooted during Failover testing
	SIP96X1- 20829	ENABLE_CONTACTS does not survive reboot when J100Supgrade.txt is not found
	SIP96X1- 20758	J129 with IPO: Phone is displaying (for 2 second) “Join” soft key after select the “Merge” soft key.
	SIP96X1- 20554	J129 phone is not working as per ‘Button_Mapping’ parameter.
	SIP96X1- 20536	J129. The parameter L2QAUD does not take effect until a 2nd reboot
	SIP96X1- 20475	J129 : Back button plays error tone if pressed on NewCall screen
	SIP96X1- 20398	Bogus change of Admin->IP Configuration->IPv4->DHCP causes unexpected reboot
	SIP96X1- 20395	Phone displays "Dial or Select Party" when phone pressed New Call softkey on locked screen
	SIP96X1- 20386	Phone does not display fully and correctly information of a forwarded call after answering or ignoring the call even though all related information sent to the phone via Invite message from SM.
	SIP96X1- 20350	Conference name is not updated when a point to point session promoted to a conference session
	SIP96X1- 20308	J129 phone still shows lock screen after exiting CRAFT menu even though it was forced logout.
	SIP96X1- 20253	'0+' shown after long press on '0' when phone is locked out
	SIP96X1- 20120	Phone freezes at Verifying Credentials screen while logging because phone detects No SIP proxy
	SIP96X1- 20069	J129 – Cannot use Left Right navigation button for changing the UTC offset in time zone menu
	SIP96X1- 19851	Mercury phone allows to add local contact after login to a non-AST environment
	SIP96X1- 19677	Phone doesn’t display “No Ethernet” message at starting screen
	SIP96X1- 19464	J129 – Phone shouldn’t display Save soft key while IP address field in ADDR menu is blank
	SIP96X1- 19433	J129 'Dropped 7 log lines due to congested socket' avaya_phone.log missing logs
	SIP96X1- 19369	Incorrect name and values of Pass-thru parameter
	SIP96X1- 19048	Detection of default Admin password needs to be updated in Admin/Debug screen

Appendix 1 – Supported Hardware

J100 Series SIP 1.1.0.0 software is supported on the following models of IP phones.

Comcode	Short Description	Model	Note
700512392	J129 IP Phone	J129D01A	Ships with SIP 1.0.0.0 software.
700512969	J129 IP PHONE 3PCC	J129D01A	Ships with SIP 1.1.0.0 software.

Appendix 2 – Release History

The following table provides a history of the J100 SIP 1.x software releases.

Release	ID	Date	Link to Readme file
1.0.0.0	1.0.0.0.43	December 2016	https://support.avaya.com/css/P8/documents/101033485
1.1.0.0	1.0.0.0.15	March 2017	https://support.avaya.com/css/P8/documents/101037079

Appendix 3 – New 46xxsettings.txt parameter

The following settings file parameters are new for the J100 Series SIP 1.1.0.0 release.

Parameter name	Default value	Description	Source
ENABLE_UDP_TRANSPORT	1	0 - SIP Transport Protocol allows 'UDP', 'TCP' and 'TLS' to configure from User Interface 1 - SIP Transport Protocol allows 'TCP' and 'TLS' to configure from User Interface.	Settings file (3PCC only)
ENABLE_SIP_USER_ID	0	0 - No UI option is presented for SIP USER ID in ADMIN --> SIP --> SIP Global Settings. 1 - UI option is presented for SIP USER ID in ADMIN --> SIP --> SIP Global Settings to enable/disable	Settings file (3PCC only)
USER_STORE_URI		Defines the backup/restore path for User data	Settings file (3PCC and IP Office)
ENABLE_STRICT_USER_VALIDATION	0	0 - Strict validations of “To” header and “Request-URI” is not done against AOR and “contact” header published by phone in REGISTRATION respectively. 1 - Strict validations of “To” header and “Request-URI” will be done against AOR and “contact” header published by phone in REGISTRATION respectively.	Settings file (3PCC only)
ENABLE_DND	1	0 - No UI option is presented for DND feature. 1 - UI option is presented to user for activate/de-activate DND.	Settings file (3PCC only)
ENABLE_DND_PRIORITY_OVER_CFU_CFB	0	0 – Call Forward All/Busy is given priority over DND. 1 – DND is given priority over Call Forward All/Busy.	Settings file (3PCC only)
ENABLE_AUTO_ANSWER_SUPPORT	0	0 - No UI option is presented for Auto Answer feature. 1 - UI option is presented to user for activate/de-activate Auto Answer.	Settings file (3PCC only)
AUTO_ANSWER_MUTE_ENABLE	1	0 – Mute is disabled when auto answered. 1 - Mute is enable when auto answered.	Settings file (3PCC only)

Parameter name	Default value	Description	Source
HOLD_REMINDER_TIMER	0	Specifies the timer interval for Hold Reminder. Range = 0-999 seconds. Timer = 0 means Hold Reminder is not supported.	Settings file (3PCC only)
SIP_TIMER_T1	500	SIP Timer T1 is an estimate of the Round Trip Time (RTT) and is defined in milliseconds. Valid values are 500 through 10000 milliseconds	Settings file (3PCC only)
SIP_TIMER_T2	4000	SIP Timer T2 is maximum retransmit interval for non-INVITE requests and INVITE responses and is defined in milliseconds. Valid values are 2000 through 40000 milliseconds	Settings file (3PCC only)
SIP_TIMER_T4	5000	SIP Timer T4 is maximum duration a message will remain in the network and is defined in milliseconds. Valid values are 2500 through 60000 milliseconds	Settings file (3PCC only)
FORCE_SIP_USERNAME	Blank	FORCE_SIP_USERNAME replaces user field entered by user during Login	Settings file
FORCE_SIP_PASSWORD	Blank	FORCE_SIP_PASSWORD replaces password entered by user during Login	Settings file
FORCE_SIP_EXTENSION	Blank	FORCE_SIP_EXTENSION replaces User ID entered by user during Login	Settings file (3PCC only)
GET $MACADDR.txt		GET $MACADDR will request for the "MACADDR" file from the HTTP/HTTPS Server where "$MACADDR" which will be replaced by the telephone's MAC address.	Settings file
TLSSRVR		Server used to download configuration files	Settings file
TLSDIR	Blank	Specifies the path name to prepend to all file names used in HTTPS GET operations during startup	Settings file
TLSPORT	443	Sets the port used for HTTPS file downloads	Settings file

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