



Avaya IP Deskphone H.323 Release 6.4 Readme

This file is the Readme for the Avaya IP Deskphone H.323 Release 6.4 for the 9608, 9608G, 9611G, 9621G, and 9641G IP Deskphones. This file describes the contents of the software distribution package.

H.323 6.4 software is supported on the 9608, 9608G, 9611G, 9621G and 9641G IP Deskphones only and when used with Avaya Aura[®] Communication Manager. The H.323 6.4 software will not load or operate on any other models.

To upgrade your 9600-Series IP Deskphones:

1. Unzip the zip file to the root directory of your HTTP server (see package files names in section "H.323 6.4 Package Contents").
2. Update the 46xxsettings.txt file according to the setup environment.
3. Reset your Avaya 9600-Series IP Deskphone.

Avaya Aura[®] Communication Manager Compatibility

Although the 9608, 9608G, 9611G, 9621G and 9641G IP Deskphones are supported on Avaya Aura[®] Communication Manager 5.2.1 and later, Avaya recommends you to deploy the deskphones with the latest available Communication Manager release. See the "Communication Manager Software & Firmware Compatibility Matrix" at <http://support.avaya.com> for the supported software/firmware versions of the Media Server, Media Gateway, and circuit packs.

CM 6.2 is the minimum version required for *native* support of the 9608, 9608G, 9611G, 9621G and 9641G IP deskphones.

- For CM releases prior to 6.2, administer the 9608, 9608G, 9611G, 9621G and 9641G IP deskphone as a 9650 IP Deskphone.

CM 5.2.x is the minimum version required to support the following features:

- PE DUP functionality.
- Enhanced support for Station Lock feature to lock local applications like Contacts, Call Log, and A-Menu.
- Store incoming call numbers in the missed call log when they are redirected.

For more details see the H.323 configuration section in the Communication Manager Administration Guide which you can download at <http://support.avaya.com>.

New features in H.323 6.4

Phone functionality and administrative features:

- The top line on the phone will now display media quality indication for the following conditions:
 - Call uses wide band codecs
 - Phone detects poor network quality which impacts audio quality.
- Improved local contact pairing functionality in several scenarios such as hold/resume, call forwarding, agent DID calls, conference calls and bridged calls.
 - Note: In order to use this feature, you must enable the “contact pairing” feature in the phone settings menu.
- Added support for French on-screen keyboard. This feature is enabled when French locale is configured.
- Offline call log: With this feature enabled, the logged in user can see the call history of the duration when the phone was logged off.

In addition, the call log can be synchronized with the call log of a Avaya one-x communicator (H323).

This feature requires administration on CM. For more information, refer the CM Administration guide.

The following minimum versions are required:

- CM: 6.3.6
- Avaya one-x communicator: 6.2.0323
- Support for the new Global (Icons Only) version of the 9611 phone.

Administrative features:

- If craft menu password is not changed from the default password (PROCPSWD parameter), the DEBUG option is shown as read only. In previous releases the DEBUG option was not visible in this scenario.
- Added support for Service Level Agreement (SLA) monitor. This feature requires the installation of the Avaya Diagnostic Server (ADS).

SLA Mon™ technology is a patented Avaya technology embedded in Avaya products to facilitate advanced diagnostics. The technology works in a server-agent model where a SLA Mon server controls the actions of SLA Mon agents to execute advanced diagnostic functions consisting of:

Endpoint Diagnostics

- The ability to remotely control IP phones, to assist end users with IP Phone configuration and troubleshooting.
- The ability to remotely generate single and bulk test calls between IP phones.
- The ability to remotely execute limited packet captures on IP phones to troubleshoot and diagnose IP phone network traffic.

Network Monitoring

- The ability to monitor multiple network segments for performance in terms of packet loss, jitter, and delay.
- The ability to monitor hop-by-hop QoS markings for voice and video traffic.

***** Note:** In case the settings file already includes the trust certificates list in the TRUSTCERT setting file parameter, then the SLA monitor root certificate must also be added to the list and placed as the first certificate.

For example: SET TRUSTCERTS slamonRootCA.crt,rootCertRNAAD.cer

In case the ADS Server is installed with the default Avaya certificate, then the phone must use the default Avaya root certificate which is included as part of the phone package and is called slamonRootCA.crt.

Note that Avaya does not recommend the use of default certificate for ADS server as it is non-unique certificate and can expose the equipment to security risk. In order to avoid this risk, obtain and install unique certificates on both ADS server and relevant trusted certificate on the phone.

- It is now possible to query the phone hardware revision via SNMP.
- The settings file parameter IDLEFEATURES (enables a list of feature identifiers for softkey features in idle call state) is being saved in a non-volatile memory thus retaining the information after power down or reboot.
- SSH remote debug capability has been extended and now includes a wider commands set. In addition, there is a new option in the phone's DEBUG menu to enable the SSH on the fly, without performing a reboot. (works only when "SSH_ALLOWED" value is "2" in the settings file).
- Added a new configurable possibility for soft keys layout.
The new soft keys layout can be configured in the settings file by setting the value of AGTACTIVESK to "3", which means that the soft keys would be labeled as an active call in a non-call center environment from left to right: Hold, Conf, Transfer, Drop.
- Call Journaling feature. With this feature, the details of all incoming calls that were missed when the phone was logged out will be kept at the Communication Manager. These calls will be added to the call history as missed calls, as soon as the user logs in.

The following new system parameters are added to the list of 9600 Series H.323 customizable system parameters:

| Parameter name | Default Value | Possible values | Description |
|---------------------|---------------|-----------------|--|
| QLEVEL_MIN | 1 | 1 - 6 | Specifies the minimum quality level in which a low local network quality indication will not be displayed. |
| WBCSTAT | 1 | 0 or 1 | Specifies whether a wideband codec indication will be displayed when a wideband codec is being used. |
| AGENTGREETINGSDELAY | 700 | 0 - 3000 | Delay time (milli seconds) between call autoanswer and playing of an agent greeting. |
| CALL_LOG_JOURNAL | 0 | 0 or 1 | Triggers restore of call log journal. |

| | | | |
|---------------|-----------|--------------|---|
| PHNSCRCOLUMNS | 0 | 0 or 1 | Specifies whether the Phone Screen is presented with one (full-width) or two (each half-width) columns. |
| SLMCAP | 0 | 0, 1 or 2 | Specifies whether the SLA Monitor agent supports packet capture. |
| SLMCTRL | 0 | 0 or 1 | Specifies whether the SLA Monitor agent supports device control. |
| SLMPERF | 0 | 0 or 1 | Specifies whether the SLA Monitor agent supports performance monitoring. |
| SLMPORT | 50011 | 6000 - 65535 | Specifies the UDP port used to receive commands from the SLA Monitor server. |
| SLMSRVR | 0.0.0.0:0 | Any | Specifies the source IP address and, optionally, the source port number of valid discovery messages from an SLA Monitor server. |
| SLMSTAT | 0 | 0 or 1 | Specifies whether the SLA Monitor agent will be enabled. |
| SLMTEST | 50012 | 6000 - 65535 | Specifies the UDP ports used for RTP and traceroute tests. |

H.323 6.4 Known Issues

The following table includes deskphone known issues:

| ID | Issue Description |
|-------|---|
| 10967 | When disabling SLA monitor through setting file (SLMSTAT=0), but keeping SLMCAP and SLMCTRL enabled (SLMCAP=1 and SLMCTRL=2), then the craft menu buttons (Service mode control and Service mode record) would be shown as enabled, although SLA Monitor will actually not be working at all. Solution: in the setting file, set both SLMCAP=0 and SLMCTRL=0. |
| 10907 | When the SLMCAP parameter of the settings file is set to 1, the SLA client will not be able to capture packets. Solution: Set the value of SLMCAP to 2 to allow capturing of packets. |
| 10873 | Agent greeting of type VDN, will not work if VDN name configured on CM exceeds 15 characters. The phone supports 15 characters or less. Solution: Configure VDN names on CM to have 15 characters or less. |
| 10861 | Incoming calls display the caller name from the CM server, not the local contact name. when the information from the CM server contains an alphanumeric name. For example, if the caller information in the CM is "John 123" and the name is saved in the local contact list as "John W", the phone screen displays "John 123" during the call, and "John W" in the call history. |

| ID | Issue Description |
|-------|---|
| 10859 | <p>Changes to the "Match Criteria" field while editing an agent greeting of type VDN/ANI is not saved.</p> <p>Solution: Editing any other field in addition to the match criteria will cause the values to be saved - for example change the greeting name by one letter.</p> |
| 10712 | <p>In case the settings file already includes the trust certificates list in the TRUSTCERT setting file parameter, then the SLA monitor root certificate must also be added to the list and placed as the first certificate.</p> <p>For example: SET TRUSTCERTS slamonRootCA.crt,rootCertRNAAD.cer</p> |
| 10409 | <p>The 96x1 H.323 firmware supports both the Avaya Communication Manager and the Avaya IP Office Call Managers. You must restore a phone to factory defaults before switching between the Communication Manager environment and the Avaya IP Office environment.. Otherwise it may cause various operational issues to appear on the phone.</p> <p>Solution: For any abnormal behavior of the phone after switching between CM and IPO, go to CRAFT menu and clear values.</p> |
| 10225 | <p>When changing MD5 to TLS and also changing user name on the server, the phone's authentication fails.</p> <p>Solution: Clear values through CRAFT menu.</p> |
| 9939 | <p>The 96x1 H.323 phones support SRTP AES 128 HMAC 80 cryptographic Suite only. other cryptographic suites (un-authenticated suites and HMAC 32 based suites) are not supported.</p> |
| 9194 | <p>When Cisco Access Control Server (ACS 5.2) is configured for re-authentication timeout, using TLS session tickets to renew authentication, the phone does not re-authenticate.</p> <p>Solution: reboot the phone</p> |
| 8897 | <p>If the phone is connected to the network using VPN, the BRURI parameter contains FQDN address and not IP Address and the phone is logged out and sleep mode is activated, backup/restore will stop working until the next phone reboot.</p> <p>Solution: Configure IP and not FQDN when the phone is behind VPN.</p> |
| 10118 | <p>By opening a second call appearance and start dialing, agents are able to go back to the first call appearance and disconnect the live call.</p> |
| 10642 | <p>When configuring the phone for large fonts and the display includes a call forward icon , a ringer off icon and more than 10 missed calls, the Media Quality Indicator would override the time field on the screen.</p> <p>Solution: use normal font size when configuring the phone to display a Media Quality Indicator.</p> |
| 10344 | <p>In some complicated VPN environments, when "NVVPNENCAPS" is configured as if NAT device is used, the phone might become unresponsive on "exchanging keys" stage during phase one rekey attempt.</p> <p>Solution: Set NVVPNENCAPS to "1" in the settings file (Note: this will disable UDP encapsulation method which is usually used if there is a NAT device between the phone and the security gateway).</p> |
| 10082 | <p>Manual restore does not work when there is a logged in contact center agent.</p> <p>Solution: Log out the contact center agent, and keep the phone logged in to CM. The manual restore will work correctly at this state. Once the operation completes, log in the contact center agent again.</p> |

| ID | Issue Description |
|------|---|
| 3609 | <p>If CM reboots while an agent is logged in and in Aux-work mode, the headset will remain in off hook mode although the headset led turns off.</p> <p>Solution: Make the first incoming or outgoing call and the LED will then start functioning.</p> |
| 7864 | <p>In some cases, when you move a phone between two CM servers that have the same station number with different configurations, the phone will not load the new configuration.</p> <p>Solution: Open the station details screen in CM and perform any update to the form. This triggers a configuration update to the phone and resolves the issue.</p> |
| 7894 | <p>Using the IP redirect feature, the deskphone does not display the correct redirect server URL during the bootup sequence. The deskphone shows the original server UR instead.</p> |
| 9542 | <p>If the phone is in off-hook idle mode in the following CM versions:</p> <p>CM 6.3</p> <ul style="list-style-type: none"> • CM 6.2 SP0 and later • CM 6.0.1 SP 10 and later • CM 5.2.1 SP 14 and later <p>Users may hear a short dial tone sound in the currently active audio device when a failover occurs.</p> |
| 9477 | <p>If Audio report feature is active, the phone responds slowly for a few seconds until the report is complete.</p> |
| 9702 | <p>When EAP-TLS authentication is configured in 802.1x and authentication fails, the user will be presented with a failure screen. Pressing the "Clear" or "Bksp" touch buttons (on 9641 and 9621 phone models) in that screen may cause the phone to hang and reboot.</p> <p>Solution: Use the default userID (MAC address), and do not press the "Clear" and "Bksp" buttons on 9641 and 9621 phone models.</p> |
| 6339 | <p>In case the phone backup file is manually removed from the backup server, and then the phone reboots, the next backup operation would backup default parameters only.</p> <p>Solution: To ensure that the backup operation uses the correct backup parameters, do not remove the phone backup file from the backup servers manually</p> |
| 6985 | <p>In a Call Center, the agent greeting icon appears only on the phone screen. The greeting works properly.</p> <p>Solution: Return to the phone screen to show the greeting indication.</p> |
| 7131 | <p>In a Call Center, switching audio devices (handset, headset or speaker) while agent greeting is being played causes the agent greeting to stop.</p> <p>Solution: Do not switch audio devices while the greeting is being played.</p> |
| 6576 | <p>Touch phones: User timer is not displayed on the status line while on an active call, when 'PHNSCRALL' is set 1.</p> |
| 8892 | <p>Call Center: When CALLCTRSTAT is 1 and agent is logged in, the value of HEADSYS will always be treated as 1.</p> |
| 7474 | <p>If OPSTAT is 00x or 0, OPTSTATCC and CALLCTRSTAT are set to 1, Advanced options functionality is not blocked.</p> |
| 7040 | <p>When 'Timerstat' is set to 1 and 'Timer on' softkey is chosen while on an active call, the user timer blinks when the seconds advance.</p> <p>Solution: Use 'Timer on' softkey during idle state (not during an active call).</p> |

| ID | Issue Description |
|------|--|
| 5782 | If a phone is ringing while a failover between primary and secondary CMs in a Processor Ethernet duplicate setup occurs, the ringing for the current call will stop. The phone will keep alerting silently until the call is answered or disconnected. The ringing will continue to function properly in the next call. |
| 5078 | If the HTTP server is down, and the user is registered by static IP address and modifies call server IP address, changes are not reflected on the phone. Solution: Verify that the HTTP server is up and running before making administrative changes. |
| 4505 | Arabic language is not supported on the 9608 desk phones. |
| 5697 | After performing a downgrade of software, changing CM configuration of Button Module parameters and then upgrading again, labels are not updated on Button Module. Solution: Perform a "CLEAR" operation. |
| 7143 | If you press a second call appearance while an auto answer greeting is being played, you hear the dial tone and the greeting at the same time. |
| 8872 | If you use barge-in audio push after normal audio push, the deskphone may reset. |
| 8902 | Phone does not support using DNS for WML host name or trusted push server Solution: Use IP address for WML host name or TPS host name. |
| 8996 | If the administrator executes "Disable nr-registration" in CM, and no other server in the alternate gate keeper list is available, the phones will remain in discover mode until the alternate gate keeper will become available and will not return to the main, even if "Enable nr-registration" is executed on it. Solution: When you use "disable nr-regisration" make sure you have at least one active ESS or LSP in the network. If you've already issued the command - the alternate gate keeper must be brought up or the phones need to be rebooted in order to get back to the main server. |
| 8812 | When several certificates are provided and the first is invalid, phone will not continue to download other certificate. Solution: Use valid certificates. |
| 9582 | Team button alert is shorter when using headset. |
| 9500 | In Hebrew or Arabic, the "Enter" and "Bksp" soft keys are switched on VPN startup. Solution: When using those system languages, after entering a password, press "Bksp" to Enter and vice versa. |
| 9525 | The agent greeting feature (including agent greeting recording) does not function with Bluetooth headsets. Only wired or DECT headsets are supported. |
| 9503 | When setting the Log to file parameter at the CRAFT menu to ON and changing the Serial Port to "CLI", SBM button modules stop working. Solution: Change "Serial Port" back to "Adjunct" once debugging is completed. You can do this through the craft menu. Note: The DEBUG option is available for use only if you change the default password to the craft menu through the PROCPSWD parameter. The new value of the PROCPSWD parameter must be 4 to 7 numeric digits, "0000" through "9999999". However if value of PROCPSWD is less than 4 digits after you install R6.2.4 or later, the value will be changed back to the default value of 27238. |
| 9586 | The deskphone does not support WML of Push display when its title contains "&" or "<". |

| ID | Issue Description |
|-------|---|
| 9667 | The deskphone supports getting only one HTTP file server from LLDP. Multiple HTTP file servers can be configured on many other ways such as settings file, DHCP, CRAFT menu etc. |
| 10033 | After playing an existing greeting and using the RESTART soft key to re-record the greeting, the agent greeting icon will not be updated. Solution: Go to agent greeting screen and play one of the greetings. The icon will be updated correctly from this point on. |
| 5870 | When moving between states "After call" and "Auto-in", an Incorrect status line information appears. This problem is observed on 9611G phones only. Solution: Change the extension's configuration on CM to be administered as a 9650. CM Fixes (MR defsw100727) were delivered to the following releases: <ul style="list-style-type: none"> • CM 6.0.1 SP04.00 • CM 5.2.1 SP10.00 |
| 8533 | Synchronization issue between phone and agent after a network outage that causes phone reset and login, causing audio issues when answering a call Solution: Communication Manager Fixes (MR defsw113036) were delivered in the following releases: <ul style="list-style-type: none"> • 5.2.1sp14.00 • 6.0.1sp10.00 • 6.2sp00.00 • cm6.3 base * Synchronization issue is that the device chosen (speaker/headset/handset) is not synchronized. Choosing that device several times will cause synchronization. |
| 9559 | When running against specific versions of CM - the Auto-In LED remains lit on phone after failover from Main to ESS Solution: Upgrade to CM version 6.2 FP2/3 or CM 6.0.1 SP 10. |
| 9667 | Phone supports getting only one HTTP file server from LLDP. Solution: Multiple HTTP file servers can be configured using other ways such as settings file, DHCP and CRAFT menu. |
| 9771 | For 9608 and 9611, in guest login screen, the softkeys will disappear after 2 wrong login attempts. This will prevent users from trying a third time. Solution: Press the "OK" key on the phone and the password field will appear. Enter the correct password and press the "Up" arrow button. All the keys will appear on the screen. |
| 8897 | If the phone is connected to the network using VPN, the BRURI parameter contains FQDN address and not the IP address, and the phone is logged out and sleep mode is activated, backup/restore will stop working until the next phone reboot. Solution: Configure IP and not FQDN when the phone is behind VPN. |

H.323 6.4 Resolved Issues

The following table includes the resolved issues which are relevant when the phone is administered by a Communication Manager, Call Center or IP Office:

| ID | Issue Description |
|-------|--|
| 10430 | In some cases the lower parts of the text was truncated on button module 24 buttons. (SR: 1-4708646412) |
| 10368 | When a call center agent's phone is configured to ignore external disconnect messages for active headset, the "Mute" button doesn't work on first press, but works only on being pressed a second time. (SR: 1-4643625043) |
| 10360 | When upgrading from 6.2.1 to 6.3, the phone might not show the screen saver as configured in the settings file, but displays the Avaya screen saver instead. (SR: 1-4639842634) |
| 10324 | When logging out from one extension and logging in to another extension, the password's unconcealed digits are displayed for few milliseconds. (SR: 1-4555089071) |
| 10288 | In some scenarios such as busy state, call forwarding and Direct Inward Dialing, the phone does not pair the right name to the extension number. (SR: 1-4622032891) |
| 10260 | When missed calls are logged with a hyphenation sign, they might be unerasable from call history's "all calls" and "missed" lists. (SR: 1-4589576261) |
| 10143 | The time indication field "Seconds Elapsed" was not increased at the DHCP discover messages, as sent out by the phone. (SR: 1-4408594886) |
| 10137 | In rare cases, when the phone is used by a call center agent, the recorded agent's greeting is played slowly and garbled. (SR: 1-4433634868) |
| 10128 | In call center environment, the agent greeting playback is delayed by two seconds after the call is answered. (SR: 1-4474975557) |
| 10388 | When backing up a greeting file which is shorter than 10 seconds and then restoring that file, the greeting playback might be duplicated. |
| 10064 | During an agent login on the 9608 phone, a long tone is played for two seconds instead of the 3 short beeps. (SR 1-4469805575) |
| 10032 | During boot, when the phone receives a 200 OK message with an HTTP Response-Header Server field which includes the string "HTTP" (e.g. "MY_HTTP_SERVER") as the value, the phone does not request for 46xxsettings file. (SR: 1-4424514097) |
| 10026 | When the administrator disables the nr-registration on the main Communication Manager, the phone might not find the ESS server and without manual intervention the phone is stuck in the discovery mode. (SR: 1-4432048015) |

| ID | Issue Description |
|----------------|---|
| 10007 | When a phone extension's password is changed at the Communication Manager, the contacts list is deleted as if the extension's name would have been changed. |
| 9930 | When increasing the volume to the maximum level, the headset audio quality is slightly decreased. (SR: 1-4342845742) |
| 9717 | During a live call with an active speaker, when the user is dialing digits, duplicated digits are sent out towards a conference server or an answering machine. (SR: 1-4196202452) |
| 9553 | When the call log is enabled and a call is completed, the phone sends four updates to the HTTP server instead of one. (SR: 1-4191578357) |
| 10710 10676 | Using the "Apploader" program, arbitrary shell commands could be executed with root privileges, allowing remote potential attackers to make arbitrary changes to the phone's application and kernel. |
| 10104 | When calling through Avaya one-X Communicator, having two live calls, once the caller drops an un-muted call, the other call would be changed to un-muted as well. |
| 10094 | When using SRTP with a conference server, callers using 9600 series phones which join the conference before the moderator arrives, would hear a noise as soon as the moderator joins the call. |
| 10770 | When starting a call before midnight and ending it after 24:00, the call record would be shown at the top of the history log, as the last call of the next day. (SR-1-4995915332) |
| 10659 | The phone doesn't support 802.1x "Fail Fallback" feature, so when no RADIUS server exists, the phone wouldn't pass the 802.1x authentication by receiving a faked success from a Juniper switch. (SR 1-4778568847) |
| 10167 7478 | When working with the phone as a call center environment, the menu entry "Option & settings" is not displayed for the following settings file parameters values: CALLCTRSTAT = 1 and OPSTAT = 7. (SR 1-4557089616) |
| 10017 | When a call center agent logs in, the soft keys layout is changed from "Hold/Conf/Transfer/Drop" to "Hold/Conf/Release/More". * The fix requires to set "AGTACTIVESK" to "3" (SR:1-4344189521) |
| 9942 | In a call center environment, when calling an agent Direct Inward Dialing number, the phone doesn't perform the pairing and the caller ID shows up instead of the contact's name. (SR-1-4317249123) |
| 9777 | During a call between two phones, when the remote side is a 9641 uses a speaker, the local side would hear an acoustic echo of itself. |
| 9763 | When forwarding an incoming call the callee doesn't perform the pairing so the caller extension number is displayed instead of his contact name. |
| 9640 | When a 9611 phone is being unplugged from a tagged network and plugged back to an untagged one, the phone gets stuck on DHCP process. (SR 1-4221502279) |

H.323 6.4 Package Contents

The H.323 6.4 package contains all the files necessary to upgrade Avaya new or previously installed IP deskphones to H.323 6.4 load.

The following files are included in each package:

- S9608_11HALBR6_4_0_14_V452.tar - The 6.4 H.323 phone application tar file for 9608 and 9611G models.
- S9621_41HALBR6_4_0_14_V452.tar - The 6.4 H.323 phone application tar file for the 9621G and 9641G models.
- - The 6.4 H.323 Kernel and root file system tar file.
- 96x1Hupgrade.txt - This file is downloaded by the 9608/9611G/9621G/9641G IP deskphones and instructs the phones on how to upgrade. DO NOT EDIT this file. You MUST USE the 96x1Hupgrade.txt file included in this package to upgrade H.323 software.
- 18 predefined language files for phone display:
 - mlf_S96x1_V87_arabic.txt
 - mlf_S96x1_V87_chinese.txt
 - mlf_S96x1_V87_dutch.txt
 - mlf_S96x1_V87_english_large.txt
 - mlf_S96x1_V87_french_can.txt
 - mlf_S96x1_V87_french_paris.txt
 - mlf_S96x1_V87_german.txt
 - mlf_S96x1_V87_hebrew.txt
 - mlf_S96x1_V87_italian.txt
 - mlf_S96x1_V87_japanese.txt
 - mlf_S96x1_V87_korean.txt
 - mlf_S96x1_V87_polish.txt
 - mlf_S96x1_V87_portuguese.txt
 - mlf_S96x1_V87_russian.txt
 - mlf_S96x1_V87_spanish.txt
 - mlf_S96x1_V87_spanish_latin.txt
 - mlf_S96x1_V87_template_english.txt
 - mlf_S96x1_V87_turkish.txt
- av_csca_pem_2032.txt (Avaya Call Server Root Certificate)
- av_prca_pem_2033.txt (Avaya Product Root CA certificate)
- 96x1mibDRAFT.txt for reference
- AvayaMenuAdmin.txt template for reference
- release.xml

The signatures in the signatures subdirectory of the .zip distribution packages are only intended to be used by the file server, and the file server that is on the CM6.0 Utility Server is the only file server that currently supports this.

System specific parameters should be entered into the 46xxsettings.txt file which is available for separate download at <http://support.avaya.com>

The H.323 6.4 package is available in the following versions:

- Versions with encryption enabled
 - 96x1-IPT-H323-R6_4_0_14-040314.zip
 - 96x1-IPT-H323-R6_4_0_14-040314.tar
- Versions with encryption disabled
 - 96x1-IPT-H323-R6_4_0_14U-040314.zip
 - 96x1-IPT-H323-R6_4_0_14U-040314.tar

Previous releases:

The following paragraphs include the features and changes of releases prior to 6.4.

Features in H.323 6.3.1.16

This release was created in order to support the new 9608G phone model, which is an improved 9608 with a Gigabit Ethernet port, as well as the new Global (Icons Only) version of the 9608 phone.

The software is compatible with the 9608, 9608G, 9611G, 9621G, and 9641G.

New Features and changes in H.323 6.3.0.37

- Turkish and Polish language support for IP Office
- VPN support for IP Office
- EAP-TLS support for network authentication, including Single Reboot Deployment. EAP-TLS is a more secure authentication method than EAP-MD5.
- HTTP Redirection supported for faster file downloading.

HTTP Redirection allows multiple servers to be used to download files to IP deskphones without the need to configure different values of HTTPSRVR (or TLSSRV) for different groups of deskphones. Use HTTP Redirection to speed up and balance firmware upgrades to many deskphones spread over multiple sites.

- Identity Certificate Authentication for TLS. You can use TLS (HTTPS) to help protect private user data contained in backup/restore files, to encrypt the files during transmission, and require authentication to read or write the files.
- Single Sign-on for Local Device API Support. The deskphone can now detect a PC application that is connected through the phone PC port and receive log in and logout commands through a PC application. The PC application can be developed to automatically log in the phone when the PC is logged in or log out when the PC is logged out. Contact Avaya DevConnect to find business partners that have developed client applications for use with the deskphone. Development documentation for customers/partners that wish to develop their own client application is also available under: <https://downloads.avaya.com/css/P8/documents/100174196>. For further questions please contact Oliver Bengtsson, at bengtss@avaya.com.
- For wireless headsets connected through the ESH cables, you can now disable headset ringing and permit on hook/off hook operations only. Refer the corresponding admin and user guides for more information.
- Support of new World Online API posted in August 2013 to enable full detail presentation in the World Clock feature.
- The following new system parameters have been added to the list of 9600 Series H.323 Customizable system parameters:
 - SYSAUDIOPATH: To set the default audio path to the speaker or the headset, or allow the call center agent to select the audio path.
 - CLOGOUTIDLESTAT: To configure the headset LED to remain on after the call center agent logs out.
Note: The craft menu access procedure was changed; in order to check whether the headset LED is ON, the agent must press the mute button twice.
You can gain access to the craft menu when phone is on headset off-hook idle state.
 - PHY2_AUTOMDIX_ENABLED: To configure automatic recognition of crossover or straight Ethernet cables on the deskphone PC port (Auto MDIX). By default, Auto MDIX is enabled on the PC port. Disabling Auto MDIX prevents accidental network loops when both the deskphone line interface and the PC interface are connected to a network switch with no loop prevention mechanisms, for example, Spanning Tree.
 - DOT1XWAIT specifies whether the phone will wait for 802.1X to complete before proceeding with startup and initiating DHCP. Default value is 0, which means that when the 802.1X Supplicant is started, startup will continue without waiting for 802.1X authentication to complete, =1 Startup will not continue
 - New parameter LOCALZIPTONEATT was added to control the local phone ziptone heard when using AUTOANSSTAT 1. Note – If Auto answer is configured on the CM and not by using the AUTOANSSTAT parameter setting, this parameter does not influence that zip tone volume. The parameter name is LOCALZIPTONEATT and valid range is between 0-95 (0 being the loudest, 95 lowest).
 - New LEDMODE parameter added to support changed LED behavior. OLD behavior would be maintained as default (LEDMODE 0) and would mean that the red led is controlled locally by the phone. New behavior may be obtained by setting LEDMODE to 1 and would mean the buttons red LEDs are controlled by CM (this will align the 96x1

phone behavior to the 16xx behavior). For example, Button module and phone LED changes are aligned according to call state etc.

- SSO_ENABLED: To implement the Single Sign-On (SSO) feature. Additional related parameters added are – SSO_REGISTERED_MODE, SSO_LOCK_SYNC, SSO_DISCONNECT_ACTION, SSO_DISCONNECT_FACS, SSO_CLIENT_CERT. Details and explanation of Single Sign-on and API usage can be found in the Admin guide and Single Sign-on application note on the Avaya support site.
 - LLDP_XMIT_SECS specifies the rate in seconds at which LLDP messages will be transmitted. Valid values are 1 through 3600; the default value is 30. Main usage is for the SSO application to discover the phone faster.
 - AGTSPKRSTAT: Modified to allow the call center agent to use the SPEAKER button to release an ongoing call.
- Voice Initiated Dialing (VID) is no longer supported.

Changes in H.323 6.2.4

The DEBUG option is available for use only if you change the default password to the craft menu through the PROCPSWD parameter.

The new value of the PROCPSWD parameter must be 4 to 7 numeric digits, "0000" through "9999999". However if value of PROCPSWD is less than 4 digits after you install R6.2.4 or later, the value will be changed back to the default value of 27238.

New Features in H.323 6.2.3

Support for option to activate mute when using the phone in Shared Control mode with One-x Communicator causes the speakerphone to go off-hook.

An application that uses Shared Control mode can cause the phone to go off-hook on speakerphone even though the user is not at the phone. Since this can be a privacy concern, CM and the deskphones now support an option by which the speakerphone will be automatically muted in this scenario, and will remain muted unless mute is deactivated by a user at the deskphone.

The functionality is available only for customers using CM 6.2 FP1 and above.

Customers using CM 6.2 FP1 who had previously configured the 96x1 aliased to 96x0, will need to re-program the deskphones to avail of this functionality.

Support for improved activation of Agent Greetings

To improve the operation of the Agent Greeting feature, CM and the deskphones now support a mechanism by which the Agent ID is automatically downloaded to the phone after an agent logs in. Hence, agent personalized greetings will be available in cases where they have not been previously, such as logins via Computer-Telephony Interface (CTI) applications.

The functionality is available only for customers using CM 6.2 FP1 and above.

Certificate renewal clarification:

Certificate renewal is initiated when the date and time of the deskphone's local clock reaches (or exceeds, if the deskphone starts up after) the date and time that corresponds to the expiration of a percentage of the certificate's valid lifetime (as specified in the certificate's Validity object) that is equal to the value of MYCERTRENEW.

Dates and times specified in certificates are in Greenwich Mean or Universal Time (GMT/UT) but the deskphone does not know the offset of its local clock relative to GMT/UT, so renewal is initiated assuming that there is no offset, which may result in renewal starting up to 12 hours earlier or later than the specified time. Since certificate lifetimes are typically on the order of a year or more, ignoring the offset relative to GMT/UT is expected to be well within the precision of the value of MYCERTRENEW. However, if a certificate with a very short lifetime is used for testing purposes, it should be assumed that the local time is GMT/UT.

Even though the value of MYCERTRENEW is allowed to be set to values that correspond to from 1% to 99% of the certificate's lifetime, SCEP servers typically do not expect, and may reject, renewal requests that are received before at least 50% of the certificate's lifetime has expired. Setting MYCERTRENEW to values less than 50 is supported for testing purposes, but corresponding server administration may be required.

New Features in H.323 6.2.2

Provide user option for bi-directional headset identification

The SP1 Release for H323 6.2 introduced support for ringing on certain wireless headsets from Jabra and Plantronics. You can enable ringing on the wireless headset using this feature. To enable and use this feature, the base unit of the wireless headset should be connected to the phone and you must configure SET HEADSETBIDIR 1 in the settings file.

In this Release, the deskphones expand that functionality by giving the user the same selection capability. The new user option is listed under the Call Settings menu, and is called "Headset Signaling". The default is the current value of HEADSETBIDIR is Off. On means the headset is assumed to be bi-directional; Off means the headset is assumed not to be bi-directional. The user has normal access to this new option, can toggle back and forth from Off to On, and the final choice takes effect when the option is Saved.

Phone-based conditional auto-answer

You can have the deskphone automatically answer incoming calls, or a subset of incoming calls, independently of the auto-answer setting on Communication Manager. This functionality requires at least one new settings file parameter, AUTOANSSTAT, and may require either or both of two others, AUTOANSALERT and AUTOANSSTRING, as noted below.

If AUTOANSSTAT is set to 0, the deskphone never auto-answers. This is the default value.

If AUTOANSSTAT is set to 1, the deskphone auto-answers all calls, unless AUTOANSSTRING is non-null.

If AUTOANSSTAT is set to 2, the deskphone auto-answers all calls on primary call appearances unless AUTOANSSTRING is non-null, but never auto-answers calls on bridged call appearances.

If AUTOANSSTAT is set to 3, the deskphone auto-answers all calls only if the user has logged into a call center as an agent, unless AUTOANSSTRING is non-null. If the user has not logged into a call center, the deskphone does not auto-answer any calls.

If AUTOANSSTAT is set to 4, the deskphone auto-answers all calls on primary call appearances only if the user has logged into a call center as an agent, unless AUTOANSSTRING is non-null, but never auto-answers calls on bridged call appearances. If the user has not logged into a call center, the deskphone does not auto-answer any calls.

In the statements above, "AUTOANSSTRING" refers to a new settings file parameter. If it is null (the default), it is ignored and only AUTOANSSTAT is used to determine if a call is auto-answered or not. If AUTOANSSTRING is non-null, the incoming call is auto-answered if both AUTOANSSTAT allows the call to be answered, AND if the incoming call's display message includes the contents of AUTOANSSTRING anywhere in the message. In general, it is expected (but not required) that AUTOANSSTRING would be non-null when AUTOANSSTAT is 3 or 4.

The deskphone will NOT auto-answer by AUTOANSSTAT if the user is dialing, or is already on an active call. This means the user will not be interrupted if the phone is being used. The deskphone will auto-answer by AUTOANSSTAT if all calls appearances are in Idle or some of the call appearances are in Held but there are still Idle call appearances. The deskphone will auto-answer with the current "audio path" device (under Call Settings screen).

Given that the deskphone auto-answers the call, if audible alerting would normally be provided for the incoming call, if AUTOANSALERT has value "1", the deskphone provides normal audible alerting for the auto-answered call. If AUTOANSALERT has value "0" (the default), the deskphone does NOT provide audible alerting for the auto-answered call, even if audible alerting would otherwise be provided.

New Features in H.323 6.2.1

Bidirectional wireless headset support

The SP1 Release for H323 6.2 includes support for ringing on certain wireless headsets from Jabra and Plantronics. You can enable ringing on the wireless headset using this feature. To enable and use this feature, the base unit of the wireless headset should be connected to the phone and you must configure SET HEADSETBIDIR 1 in the settings file.

When you switch on the base unit the either of the following scenarios may result:

1. When user goes off-hook with the headset or switches from non-headset device to headset, the wireless headset will be activated.
2. When user goes on-hook with headset or switches from headset device to non-headset, the wireless headset is deactivated.

When an incoming call arrives:

- If the phone is in idle state, a ringing tone is heard in the wireless headset.
- If the phone is already in a headset call, a short beep is heard in the wireless headset.

Controlling headset by settings file

The HEADSYS parameter is re-introduced.

HEADSYS specifies whether the deskphone will go on-hook if the headset is active when a Disconnect message is received.

The default value of HEADSYS is tied to the value of CALLCTRSTAT.

If CALLCTRSTAT has value 1 (meaning the phone is a call center phone), then the default of HEADSYS is 1; If CALLCTRSTAT has value 0 (meaning the phone likely not a call center phone), then the default of HEADSYS is 0. In either case, though, the administrator would have the normal ability to override defaults by explicitly setting HEADSYS in the settings file – in non-call center environment only (when CALLCTRSTAT is 1 and agent is logged in, the value of HEADSYS will always be treated as 1).

Note: Customers using Call center features without configuring "CALLCTRSTAT" should set the "HEADSYS" parameter to "1".

Enhancements in H.323 6.2

- Bluetooth support for 9641G
 - May be turned off by settings file
 - See documentation for more information.
- Wireless headset ring. Ringing is supported on wireless headsets. It is supported with Plantronics WO1 Base + AVP 62-1 cable + Hydra WH100 headset.
 - Jabra (cable # 14201-33) doesn't support the incoming call alert.
- Local Timer stop watch feature.

The desk phones now support a local timer (stop watch) feature. The agent is able to select the

Local Timer Feature and to start a stop watch.

- WML buttons: The capability to define WML buttons with functions is extended to allow the implementation of call control capabilities.
- Visiting user profile: A user will be able to log into any 9600 series Deskphone within the configured network of systems using their unique user identifier and password.
- The following new system parameters have been added to the list of 9600 Series H.323 Customizable System Parameters
 - AGTCALLINFOSTAT - Automatically invokes the Call-info permission flag; for Call Center use only.
 - AGTFWDBTNSTAT - Disables/enables the Forward button permission flag; for Call Center use only.
 - AGTGREETINGSTAT - Sets the agent greeting permission flag; for Call Center use only.
 - AGTLOGINFAC - Indicates the Feature Access Code to be used by agents when logging in to a Call Center; for Call Center use only.
 - AGTSPKRSTAT - disables/enables the speakerphone permission flag; for Call Center use only.
 - AGTTIMESTAT - suppresses the date/time presentation flag; for Call Center use only.
 - AGTTRANSLTO - Translation for "to" as used on the call server; for Call Center use only.
 - AGTTRANSLCLBK - Translation for "callback" as used on the call server; for Call Center use only.
 - AGTTRANSLPRI - Translation for "priority" as used on the call server; for Call Center use only.
 - AGTTRANSLPK - Translation for "park" as used on the call server; for Call Center use only.
 - AGTTRANSLICOM - Translation for "icom" as used on the call server; for Call Center use only.
 - CALLCTRSTAT - Call center functionality flag; for Call Center use only.
 - OPSTATCC - OPSTAT override flag; for Call Center use only.
 - TIMERSTAT - Controls whether the User Timer Timer On and Timer Off softkeys display on the Deskphone or not.
 - ADMIN_HSEQUAL - specifies handset audio equalization standards compliance
 - DEFAULTRING - specifies the default ring tone
 - BLUETOOTHSTAT - specifies whether the user is given an option to enable Bluetooth
 - HEADSETBIDIR specifies whether an alerting tone is generated on the headset interface
 - RECORDINGTONE specifies whether Call Recording Tone will be generated on active calls
 - RECORDINGTONE_INTERVAL specifies the number of seconds between Call Recording Tones
 - RECORDINGTONE_VOLUME specifies the volume of the Call Recording Tone in 5dB steps
 - SSH_ALLOWED specifies whether SSH is supported
 - SSH_BANNER_FILE specifies the file name or URL for a custom SSH banner file
 - SSH_IDLE_TIMEOUT specifies the number of minutes of inactivity
 - LANGLARGEFONT specifies the name of the language file for the display of large text
- The Management Information Base (MIB), available on the Avaya support site has been updated for several new parameters and to increase the number of syslog event messages in the endptRecentLog and endptResetLog parameters.
- The deskphone display now combines the Title line and the Prompt line shown on other 9600

Series IP Deskphones into one Status line. The Status line appears as the second display line, under the Top line.

- Headset RX gain (volume) levels were increased by average of 2-5dB.
- Weather and clock indications.

Powered by “World Weather Online” <http://www.worldweatheronline.com> and “The Weather Channel” <http://www.weather.com/>

Call Center enhancements

- Agent greeting support:
 - Greeting maximum size is 10 seconds, and about 6 greetings can be saved.
 - Manual recording of agent greetings is supported.
 - Agent Greetings can be saved on the phone between logins by the same Agent.
 - Automatic prompting for greeting.
 - Greetings can be automatically triggered per event such as time of day, answering a call, ANI, VDN and prompted digits.
- Agent login and logout detection via CTI, Abrv-dial or FAC/autodial.
- User Timer extended after call is over.
- Call appearance is now 15 characters wide for VDN name for touch phones, for button phones it must be 12 or 24 characters wide, depending on screen width.
- The Agent can press the UUI-INFO button to get the UUI information displayed. If this information does not fit on the allocated display area the content will scroll on the line.
- The Agent can use the Caller info button to view secondary information about the ongoing call (collected digits, qcalls, q-time, trk-id, VuStat display, and uui-info).
- Call Recording Tone administrable.
- Dual headset adapter support.

The agent desk phone, with the dual headset adapter box, provides two headset ports for the agent and the supervisor using the standard RJ9.

H.323 6.0/6.1/6.2/6.2.1/6.2.2/6.2.3/6.2.4/6.3 resolved Issues

Issues resolved in previous releases and service packs, which are also included in this version:

H.323 6.3 Resolved Issues

H.323-6.3 release provides continued improvement and optimization to stability, robustness and reliability.

The following table includes the resolved issues which are relevant when the phone is administered by a Communication Manager, Call Center or IP Office:

| ID | Issue Description |
|----------------------|---|
| | After August 2013 - World Clock feature does not display the correct data. World Online API will no longer work after August 2013, beginning September 1st a new API is required to obtain data for the World Clock phone feature. When using the World Clock feature on the IP phone, after API support expires and on firmware without the new API change, the user will see an error with a message: "Could not access internet service". |
| 73663218 | Jitter Statistics in RTCP packets do not show the correct value. |
| 46922778 | The ringtone volume is also changed whenever the user changes the speaker volume. |
| 27869645 | Slow internet rate for a PC which is working in 1GBps while the phone network port is connected to a 100Mbps switch port (inactive flow control). |
| 93915362 | VPN tunnel establishment fails due to a long duration of certificate confirmation. |
| 55940782 | Two phones which are logged in via VPN, receive the same duplicated IP address if the first two bytes of their MAC addresses are the same. |
| 18920196 34967756 | Button module labels disappear after a network outage. Even though the BRURI address isn't defined in the settings file, 9608 displays "retrieval failure" during login, instead of ignoring the restore operation. Occurs in IP Office mode only. |
| 90931678 | Phone does not display umlauts on the third row when working as a call center agent. |
| 12972554 | When calling back to a missed call with a 2 digits extension, the whole log of missed calls is cleared, instead of clearing only the missed call entry. |
| 87971315 | When speaker is configured to "one way talk path", headset and handset get stuck on mute mode even though the mute LED isn't glowing. |
| 72961508 | Phone reports incorrect uptime in SNMP query responses. |
| 63978409 | Red and green LEDs do not function according to call states. Applies to 9608 and 9611 only. |

| ID | Issue Description |
|----------|---|
| 25978445 | New LEDMODE parameter added to support different LED behavior. OLD behavior would be maintained as default (LEDMODE 0). New behavior may be obtained by setting LEDMODE to 1 (this for example will align Button module and phone LEDs in case of team button used, red led changes according to call state etc). |
| 07957398 | If the phones are configured to work with 802.1X authentication, but the network switches are configured with "Force Authorize" (no Authentication) - then the phone will refuse to connect to the network. |
| 23961523 | Incorrect sysuptime reported in SNMP MIB 2: (iso.org.dod.internet.mgmt.mib-2.system.sysUpTime.0) |
| 35976045 | Chinese underscore letters in WML application are shown as strikethrough |
| 54957424 | When changing the password on CM during an active call, the call gets dropped in case a headset was in use. |
| 25983456 | SCEP failure timeout was increased. The phone will now allow longer time for SCEP to complete. |
| 25952985 | MYCERTCN parameter support for serial Number ("\$SERIALNO"). |
| 25766985 | Phones do not register to CM after VPN tunnel has been established over SR 4134 , Avaya VPN Gateway 3050/3070. |
| 25972045 | When using 1XC in shared control mode with 96x1, setting - Turn On Mute for Remote Off-hook Attempt is set to 'y' on station form along with a specific CM patch where CM sends mute twice – the phone turns off mute. |

H.323 6.2.4 Resolved Issues

| ID | Issue Description |
|----------|--|
| 21894913 | Tuning frequencies used with bi-directional feature. This change includes support for Jabra ESH Cable firmware 1.5 |
| 87943172 | Blank screen appears in split screen mode on specific scenarios. |
| 98938654 | When six feature identifiers are defined via IDLEFEATURES parameter and two of the feature identifiers are not valid in CM, the sixth feature identifier doesnot appear in the soft key. |
| 78913448 | In case of adding SSH_ALLOWED to the settings file with a different value than default and removing it without reverting it - the SSH_ALLOWED parameter do not take default value. |
| 78891032 | On specific cases - agent greeting plays out in slower speed. |
| 78800123 | Issues seen after extensive continues one hour testing of audio push every 5 seconds. |
| 78928345 | 9611 in half screen mode does not display caller info on top line. |
| 78921143 | Issues with flow control. |
| 67934232 | Button module is not registered when changing the module type (BM12 and BM24) |

| ID | Issue Description |
|----------|---|
| 56753343 | Rare delays with phones connecting to call servers upon CM upgrades |
| 56772632 | When using IPO - Issues seen on active calls in case the connection to IPO is lost temporarily and retrieved. |
| 67920356 | Changing the speaker volume changes the ringtone volume as well. |
| 56877156 | When using IPO - local DTMF duration when dialing last digit is sometimes too long. |
| 56904032 | RTCP field population is sometimes missing. |
| 45924843 | Button phones: Topline does not toggle in Half Screen mode. |
| 56941168 | A delay of toggling the timer on topline when phone in half-width and length of topline info is precisely at the end of the line(26-30 characters). |
| 65827222 | Audible Call-Pkup alerting not working on phones if a "rich" ringing tone is chosen. |
| 67817245 | Issues seen when VID is enabled when having network issues. |
| 43863043 | In German language the call log history displayed number overlaps time information. |
| 78833956 | Issues when connecting 2 button modules (SBM24) to the phone when multiple interaction occurs with the BM (for example - multiple team buttons, features and calls configured on the BM). |
| 65914658 | Issues seen with BM12 when multiple interaction occurs with the BM (for example - multiple team buttons, features and calls configured on the BM). |
| 89849876 | When the phone has DNS server defined and IP address configured for WMLHOME, the WML page loading is slower. |
| 98829843 | UI issues with first line appearance when phone password is reseted from CM during phone operation. |
| 19910232 | Issues with failover during active call or headset on (for example auto in state) [Internal issues fixed- Active shuffled call drops immediately when LSP failover happens]. |
| 67881467 | Debug - log to file parameter in debug mode was with wrong default when clearing values. |
| 43839654 | Backlight soft key position is not fixed with a specific configuration. |
| 56920032 | Issues with ring ping on IPO. |
| 12823876 | Random issues retrieving greetings. |

H.323 6.2.3 Resolved Issues

| ID | Issue Description |
|----------|---|
| 23738234 | 'Drop' softkey appears on active ACD/DAC calls when AGTACTIVESK is set to 2 and no manual greeting is recorded. |
| 54714054 | Transport Company. Issue: 96x1 VPN does not upgrade while NVHTTPSRRV has DNS and not IP. |
| 54768645 | Connectivity is lost for PCs which are connected to the phones, when PCs are swapped. |
| 43758232 | Issues when receiving multicast packets of specific kind. |
| 34749232 | 9611 Redial not showing all digits. |
| 23585332 | Backlight does not go off when 96x1 has only Bridge Appearances. |

| ID | Issue Description |
|----------|--|
| 34501533 | 96x1 Phones are not renewing the downloaded CA certificate as per the value set in MYCERTRENEW parameter. |
| 34729543 | 9611G speaker volume is dropping for a second or two while 2nd call appearance is ringing. |
| 45785432 | Issues with backup restore against the Avaya utility server. |
| 76437276 | Not friendly to Transfer a call when using SpeedDial button. |
| 32565221 | Unwanted line display on the top of the screen. |
| 45747722 | Cities outside of United states and Canada are showing up with states instead of country. |
| 45758054 | phone doesn't validate certificate time in WML. |
| 34781543 | intermittent Hyperactivity on 96x1 sets. |
| 46724568 | 1st call appearance get stuck when the 2nd call appearance is dropped. |
| 48644038 | Issues with Button modules on 9608 on rare occasions. |
| 90780122 | When disabling nr-regisration phone doesn't jump to the first ESS. |
| 98788908 | 9608 IP phone is tagging L2Q COS Priority 5 but advertising a value of 6 via LLDP from the phone. |
| 98811529 | Robustness in case of very high rates of broadcast and multicast. |
| 87814454 | Support of certificate key length of 2048. |
| 98797234 | Ringtone volume is too loud at its lowest level. |
| 78794798 | Synchronization issues between the phone and Jabra headset . |
| 30781500 | Resilience against phone hyperactivity usage. |
| 91839982 | Certificate key length change from 1024 to 2048 will not trigger certificate download. |
| 98834876 | 96x1 phones not able to renew the identity certificate issued through SCEP. |
| 87845586 | Redial softkey is not always presented in CC environment. |
| 76844875 | When using phone auto answer local feature, Zip tone is too loud. |
| 23825908 | During secure backup restore invocation, intermittent stability issues. |
| 78859576 | One way talk path issues when seeing issues against the DHCP server. |
| 87825976 | When issuing specific CM commands, while second call appearance is in use, phones reset intermittently. |
| 89915234 | Users which obtained 6.2.3.12 version and are using 9621G or 9611G with the Serial Number format: yy5Nxxxxxxxxxx and yyFCxxxxxxxxxx, where yy is 12 or 13, may notice on 9611G lack of sharpness and on 9621G they may experience upside down display due to a software bug. |

H.323 6.2.2 Resolved Issues

| ID | Issue Description |
|----------|---|
| 65619189 | When the AGTTRANSLTO is set to "to" and the language on CM is not English, the associated VDN greeting is not played. |
| 69651553 | Certificate is not downloaded with a 46xxsettings file of small size. |
| 98655987 | On specific scenario soft key misplacement. |
| 65648523 | Issues with WML proxy authentication. |

| | |
|----------|--|
| 89650043 | Call Log issues when using abrv-dialing. |
| 13710000 | While user is on active call on 1st CA, the next incoming call also gets auto-answered. |
| 43642732 | Bluetooth interoperability improvements. |
| 87677423 | Issues in case of broadcast flooding or multicast flooding (starting with 01). |
| 12539943 | Fix of AGTVUSTATID behavior in case of multiple VU-STAT buttons administrated. |
| 44672209 | The active CA is not highlighted when transferring a call using a busy indicator FB button. |
| 98651033 | Issues with station call during recording of agent greeting. |
| 98683923 | Sometimes when using rich ringtone, short alert heard at speaker of agent's 9621 prior to agent greeting on incoming customer call. |
| 98650233 | Issues with agent greeting after putting a call on hold, dropping a call or when an incoming call is received on second call appearance. |
| 98709934 | White noise in headset and handset. |
| 99662233 | After admin presses clear values on the phone, the message: "clearing values" is not presented on the phone. |
| 87498532 | Desk phone sends out GRQ instead of KARRQ once CM6 PE duplicate doing soft interchange with "sever -i". |
| 43673723 | Agent greeting issues on DID call scenario. |
| 23539932 | AGTVUSTATID parameter was previously known as AGTIDVUSTAT. |
| 92501510 | When the value of MYCERTRENEW parameter is set at the settings file, Phones do not renew at accurate times. |
| 72705921 | When changing a device after agent login, agent greeting does not play. |
| 02703353 | Implement Stop button during greeting playing in Call Center. |
| 87715623 | while recording a greeting mute functionality is disabled |
| 87667532 | Button phones - Call Screening programmed on button modules should have no visual indication after IP office reboot. |
| 98709976 | White noise in headset and handset on specific scenario. |
| 76725564 | Soft keys are not displayed correctly with AGTACTIVESK. |
| 65680134 | Issues with DHCP under specific network conditions. |
| 76437289 | Enhancement when transferring a call using speed dial button. |

6.2.1 Resolved Issues

| ID | Issue Description |
|----------|---|
| 56634582 | White line border on screen saver. |
| 36628959 | 9608G running 6.2 issues with DHCP option 242. |
| 12617865 | On rare occasions, audio is heard on both the headset and the handset. |
| 35630173 | When using BT headset with 9641G phone calls are disconnected. |
| 58605145 | When using Russian language, incorrect speed is shown in network information menu (100M instead of 1G). |

| ID | Issue Description |
|----------|---|
| 63642585 | <p>Agent greetings cannot be cleared from the phone using the menu. Details: SP1 provides support for a new setting file parameter as detailed below:</p> <ul style="list-style-type: none"> • Adding new parameter AGTGREETLOGOUTDEL, with a default value of 0. • If AGTGREETLOGOUTDEL equals 1, then the agent greeting will be removed upon agent logout.(This implies that it is obtained from the server during each login). <p>If AGTGREETLOGOUTDEL equals 0 or is not present in settings file, agent file data will not be removed.</p> |
| 75575846 | Desk phones delayed boot on specific cases. |
| 59568782 | Irregular behavior of status line on desk phones on main screen during agent login/logout or when moving between agent states. |
| 99627349 | US Zip codes not working with the new Weather API. |
| 75630486 | Wrong German translation for "History". Subject to user interpretation. |
| 78633286 | Screen saver image stays in one place and no longer moves around. |
| 41625235 | Weather application issue. |
| 15635045 | Station file on the HTTP server gets overwritten with agent greeting data in a certain scenario. |
| 65634985 | Voice quality issues, echo, feedback and noise in specific scenarios. |
| 75649325 | Agent greeting is playing to active call when another call arrives on bridged appearance. |
| 15582212 | Headset does not turn off after agent logs off from a 9641G deskphone, without making any call. |
| 96638852 | <p>Ability to change configuration such that additional release button would not appear in the first page of soft keys. Details: The AGTACTIVESK parameter in the settings file can be used to configure the appearance of the Release and Transfer softkeys on the Agent's deskphone. This feature is applicable only when the Agent's deskphone is in the Active Call state.</p> <p>When AGTACTIVESK is set to 0 the Release button will appear on the first row of softkeys and Transfer button will be appear on the second row of softkeys.</p> <p>When AGTACTIVESK is set to 1 the Release button will be appear on the second row of softkeys and Transfer button will appear on the first row of softkeys.</p> <p>When AGTACTIVESK is set to 2, the Release button is not displayed at all. Only the Hold/Conf/Transfer/Drop softkeys are displayed.</p> <p>Note: The default value of AGTACTIVESK is 0.</p> |
| 58436956 | Missing authentication when using APS created Secure Backup Application. |
| 68599369 | Contact names are not fully displayed when using Korean language. |
| 35588378 | When VPNCODE is configured with 7 digits, user is not able to go into VPN menu by giving access code in A menu. |

| ID | Issue Description |
|----------|--|
| 73610075 | Button phones: Topline is not getting updated on the phone screen and on Feature screen when phone screen width is half and user switches from phone screen to feature screen and vice versa. |
| 67630045 | When using BT headset with 9641G phone and answering via speaker or handset, calls are disconnected. |
| 85385975 | 9621G and 9641G in VPN mode, do not respond to PING command (FW 6.016S / 6.016T) when used in VPN mode. |
| 85429345 | Phone automatically logs in to USB credentials even if the user does not press Login button. |
| 65542445 | Unsaved greetings are displayed in agent greeting list screen. |
| 33585258 | When the extension of a registered phone with BM12 is taken by another phone, the "Login" soft key is missing and the phone cannot register by any name again. |
| 86633666 | EEPROM values incorrect leading to incorrect handset and headset audio levels. |
| 95634045 | PC port of the phone is not working after undocking the laptop. |
| 75613086 | Ring-Ping does not work with rich ring-style. |
| 48589825 | Text Entry Request is missing when trying to customize Set Absence Text on 9621G Deskphone. |
| 59612354 | HEADSYS parameter does not function in Release 6.2, as it had worked previously. HEADSYS specifies whether the deskphone will go on-hook if the headset is active when a Disconnect message is received. The default value of HEADSYS is tied to the value of CALLCTRSTAT. If CALLCTRSTAT has value 1 (meaning the phone is a call center phone), then the default of HEADSYS is 1; If CALLCTRSTAT has value 0 (meaning the phone likely not a call center phone), then the default of HEADSYS is 0. In either case, though, the administrator would have the normal ability to override defaults by explicitly setting HEADSYS in the settings file – in non call center environment (when CALLCTRSTAT is 1 and agent is logged in, the value of HEADSYS will always be treated as 1). Note: Customers using Call center features without configuring "CALLCTRSTAT", should have "HEADSYS" parameter set to "1". |
| 45595276 | Several translation issues for Japanese and Large text. |
| 36448687 | Corrupted XML push messages are not handled properly by the phone. |

6.2 Resolved Issues

| ID | Issue Description |
|----------|---|
| 46357777 | The call-forward icon for Team Button is not displayed. |
| 31331036 | For phones administered as a team, the on hook/off hook status of a member phone is not updated on the 9621G & 9641G deskphones unless Phone button is pressed. |
| 89379447 | On 9608 deskphones, in some cases, the display screen may appear to shift a line downwards causing the softkeys to disappear from view. |
| 23315823 | Sometimes there is short delay in DTMF tones played when digits are dialed. |

| ID | Issue Description |
|----------|--|
| 99333887 | In rare cases, while using the speakerphone mode for dialing, the DTMF tone for last digit double chirps, that makes the end user feel that the number has been misdialled. |
| 32478014 | The phone does not reset its IP address, after it receives DHCPNAK message from the server. |
| 29341954 | In rare cases the phone stops responding while in screen saver mode. |
| 49494945 | Phone fails to establish call while register in Standard Local Survivability (SLS). |
| 41460834 | On rare occasions, the phone might stop responding or reboot while configuring the CRAFT menu. |
| 90451047 | When a user tries to delete characters entered in the login screen, the display is not updated though the characters are deleted. This only occurs on the button phones (9608 and 9611G). |
| 64466050 | QTest is not supported. |
| 23403673 | If automatic exclusion is enabled on the phone and you disable exclusion while on a call, the Conference, Hold, Transfer and Drop soft keys are replaced by a Bridge button. Note: The new CM release will fix this issue. |
| 65505605 | In TTS and non-TTS environments where there LSP servers have different gatekeeper IDs, the phone only tries to connect to the first LSP. |
| 18490835 | In 9608/9611G deskphone on the Contacts page, during add or edit contact, when user returns again to the page of adding new symbols, the cursor might disappear. |
| 93519818 | Display username does not work when the username starts from I such as in Irene. |
| 86560457 | A non administrated button in the CM is ignored instead of being placed as an empty button on the phone screen. |
| 58561821 | Headset RX volume is not loud enough. Gain levels should be increased by 2 to 5dB. |
| 23562791 | If the user removes the headset and starts a fresh discovery after successful Bluetooth pairing, BTPI_EVT_SEARCH_RES is received 10 times (shows the same MAC address in every event) and then btpid receives fatal signal 11. |
| 27591464 | No voice path after 200 simple calls. |
| 22453438 | Corrupted Prompt Line. |
| 71429592 | If the phone reboots while in visiting user mode, the phone does not log in automatically after startup. |
| 35468849 | Auto dial labels on BM connected to 9608 deskphone are shown incorrectly |
| 12252327 | Deskphones do not respond to RSVP requests. |
| 77277777 | Deskphone time is drifting without external sync messages. |
| 28278264 | Deskphones SBM button updates problem after power-cycle. |
| 21342890 | The UI is not refreshed in the CC Environment. |
| 78360719 | Posted messages do not work properly with 9608/9611 in split screen mode. |
| 24386278 | Deskphone subscribes to push server when registering in unnamed mode |

| ID | Issue Description |
|-------------|--|
| 10390748 | Deskphone parameter "PERSONALWAV" default value is 9. It should be 1. |
| 36391387 | Softkey label is truncated in WML browser. |
| 89408911 | AGC settings file values are stored in HTTP backup file even though no change was made by the user. |
| 45412438 | The 9641G deskphone requests an 802.1X password even though the 802.1X authentication is disabled on the switch. |
| 28425867 | The 9621/9641G deskphone truncates the second digit from weather temperature. |
| 38428679 | On the 9608 deskphone functions are missing, e.g. transfer, as the display has been shifted down by one line. |
| 17430827 | Delay in dialed digits being displayed on phone. |
| 27431180 | The DTMF feedback and display of digits are delayed in split screen mode on 9611G deskphone. |
| 39431266 | Sluggish response of key press in power save mode. |
| 92432416 | The deskphone doesn't prefer to take DHCP Offer with Option 242 |
| 82452077 | The PHONE screen is not changed to main menu screen in response to an off-hook or an incoming call. |
| 47467692 | WML proxy authentication doesn't work. |
| 17470384 | 9611 deskphone cannot save contacts to USB flash drive. |
| 22493589 | The ZipTone is not heard as audio cut-through is almost 400ms. |
| 17493659 | AGC parameters in setting file do not have any effect on the phone. |
| 50497173 | After 6.020S installation, when the guest login time expires during a call, the phone will reboot and will drop the call. |
| 12510889 | Distorted dial tone after an off hook (outgoing calls are enabled). |
| 19513476 | The phone boots up with a blank screen. |
| 26522937 | Very slow performance of WML on deskphones with R6.0. |
| 51527170 | Pulsating dial tone and fragmented rtp audio. |
| 48537183 | Calls answered on speaker have low volume. |
| 33548368 | User hears "click" when shuffling is turned on. |
| 28598190 | Contacts Backup/Restore fails on 6.2. |
| 32491637 | Call Center environment - Users must preview the complete greeting from the start to the end. Stopping the preview in the middle prevents the user from playing this greeting during a call. |
| 21436379 | Call Center environment - the Release soft key button is missing while the agent is in ring back state while establishing a call. This only happens if the parameter RINGBKFEATURES is set to null and agent greetings are not used. |
| 89467363 | Call Center environment - reviewing an agent greeting during a call results in no audio path for the call. |
| IPT00057491 | Call Center agents do not hear zip tones for incoming ACD call. |
| 27425994 | Bad audio on IP Office call server. |
| 34366474 | Cannot register a 9608 deskphone to B5800/IP Office with French character in button label. |
| 32514257 | When the DHCP server is down, reusing the previous received IP address from the server is not supported. |

| ID | Issue Description |
|----------|---|
| 62470049 | Using Sleep mode on a phone over VPN might cause to the phone to reboot or not operate correctly. |

6.0 Service packs Resolved issues

| ID | Issue Description |
|------------|--|
| wi00920894 | Upgrading the 9621G to different service packs while using CM Utility server with HTTPS is now possible. |
| wi00920421 | In rare cases, an incoming call would cause the 9641G phone to ring continuously and then reboot. The issue is now resolved. |
| wi00920911 | Transfer of audio and text to 9611G and 9608 phones possible now with ANS push server. |
| wi00920945 | The phone now displays the corresponding name for an incoming call from an existing contact. |
| wi00920625 | A call to an idle phone after its backlight turns off, results in continuous ringing and a reboot sometimes. The issue is now resolved. |
| wi00920514 | Incorrect EHS timing values were causing issues with some headsets. The issue is now resolved. |
| wi00920973 | Contacts pairing was not working on incoming calls if a + sign was used with the number, example +919850881431. The issue is now resolved. |
| wi00920746 | When the user presses the contacts button while on "Extension in use #to register" screen, a continuous error beep was heard. The issue is now resolved. |
| wi00919541 | Configuring a 9608 phone having a station name greater than 16 chars long, would result in the topline displaying the call timer with last digit missing. The issue is now resolved. |
| wi00920524 | On an incoming call to a phone having the Send All Calls (SAC) feature activated, timer gets stuck on Top line Status Message (TSM), if SAC button is pressed immediately after ending call. This issue is now resolved. |
| wi00920339 | On 9641G sets, the weather application information can now be saved. |
| wi00919048 | On the web browser of the 964G touch screen phones, the alignment for the check boxes and option buttons is now correct. |
| wi00920796 | For 9608 and 9611G phones, call timer on TSM would be overlapped by Caller info. The issue is now resolved. |
| wi00920980 | After logout, the Extension field was not be pre-populated with last used extension, if parameter QKLOGINSTAT is set to \"0\" in the 46xxsettings file. The issue is now resolved. |
| wi00921082 | Audio quality for narrow band handset in 9641G has been improved. |
| wi00920804 | The contacts edit number field has been increased to display up to 18 numbers on the 9621G & 9641G phones. |
| wi00920985 | H.323 6.0 SP4.1 TTS causes a registration issue where the Tcp socket is not yet completed. |
| wi00921023 | The Phone screen was not being updated properly when SBMs (Button Modules) are attached sometimes. The issue is now resolved. |
| wi00920902 | The 964G phone updated with H.323 R6.0 SP4.1 would sometimes stop responding and reboot after last call of several calls. The issue is now resolved. Related issue wi00919714: After setting personal ring-tones and rebooting, another ring-tone would be active. The issue is now resolved and personal ringtone is played as per the selection. |

| ID | Issue Description |
|------------|--|
| wi00921124 | A resource leakage that led to outage of descriptors has been fixed. |
| wi00920604 | On phones, personalized ringtone would not be saved when the phone was logged out and then logged in from another extension. The issue is now resolved. |
| wi00920808 | AUDIOSTHS and AUDIOSTHD parameters are now working. |
| wi00921125 | You can now perform HTTP backup if port other than 80 is set. |
| wi00925034 | 9641G using H.323 6.0 SP4.1 (6.016T) with Button Module attached and with setting of PHNSCRALL1 would lock up and reset during transfer. The issue is now resolved. |
| wi00920220 | Phone user configured volume settings are now retained between reboots. |
| wi00921079 | Improved AGC response now maintains comfortable voice levels for contact center phones. |
| wi00921178 | The World clock and Weather application working now in VPN scenario on setting WLMPORT value other than 8080. |
| wi00919760 | On receiving a call from a softphone in shared control, the phone does not display redial option for call although call was displayed in history. The issue is now resolved. |
| wi00920697 | SNMP query (MIB walk) working properly when device is registered for SNMP at CM. |
| wi00924478 | Contact name can now display more than 12 characters. |
| wi00920806 | For second incoming call, ring alert was too short. The issue is now resolved. |
| wi00921165 | In some configurations, the user would experience high level of jitter influencing voice quality. The issue is now resolved. |
| wi00929219 | When user presses headset button to answer call, the headset equalizer profile was activated resulting in audio quality issue. The issue is now resolved. |
| wi00921080 | Improved AGC for phones. |
| wi00924755 | 9608 deskphone fails to upgrade over VPN when VLAN tagging is used. |
| wi00920306 | 9608 deskphones would truncate some Korean words specified in mlf_xxx_korean.txt file. The issue is now resolved. |
| wi00925787 | VPN NAT-T inter-operability issue with Cisco ASA 8.2 is now resolved. |
| wi00924494 | On the 9641G, feature buttons were not displayed for administered options where 2 Button Modules were connected. |
| wi00932018 | The AUDIOENV parameter in settings file now works as configured to manage the transmit gain of headset audio. |
| wi00928107 | 9608 and 9621G phone would stop responding if WMLIDLE TIME and URI parameters were set in 46xxsetting file. The issue is now resolved. |
| wi00932894 | The phone would ring continuously even after dropping the call when using priority calling FAC. The issue is now resolved. |
| wi00929218 | In some cases, after using the phones for long period, the user would experience slowness in response time. The issue is now resolved. |
| wi00932612 | Improved interoperability with Plantronic wired headsets (Models AWH450N & AWH55) in on/off hook state and with on hook signals. |
| wi00930715 | Topline push now working properly in phones through Avaya push demo server. (only applicable for patch 6.017C users) |
| wi00927313 | Incoming alert is now displayed on UI for incoming call while screen saver is on. (only applicable for patch 6.017C users) |

| ID | Issue Description |
|-------------|--|
| wi00927605 | The phone plays the selected personalized ring only applicable for patch 6.017C users. |
| 39305254 | Upgrading 9621G deskphone may fail while using HTTPS in certain cases. |
| 80328689 | The phone enters a loop if both LLDP MED Network policy TLV & VLAN NAME TLV are advertised in the Extreme switch simultaneously. |
| 31356847 | In some cases, softkeys are not updated after pressing drop softkey |
| 52369137 | Incoming ringtone alert for 1st CA gets stopped at all, when user toggles 2nd CA while the first CA is busy in incoming call. |
| 71297037 | There is about 1 second of speech drop-out in the phone every ten seconds. The RTP is clean. |
| IPT00057229 | Bad audio quality after upgrade from 12S to 12Y. |
| 22298658 | Garble on shuffled calls (Bad timestamp issue). |
| 42262258 | Date and time freezes on the display. |
| 21264849 | Stutter dial tone after upgrade from SP2 to 12t in Extended Rebind state. |
| IPT00057417 | The same SSRs are used by more than one phone within minutes of each other |
| IPT00055467 | 9608 & 9611G: Noise on active call, when second call is ringing at 2nd appearance. |
| 51205348 | No dialtone on 9608/11 phones. |
| IPT00056118 | 9611G deskphone with 802.1x authentication enabled does not switch to voice VLAN from data VLAN. |
| 34235456 | Phone loops and reboots after displaying \"Retrieval in Progress\" |
| IPT00057473 | Phone does not contact the HTTP server if the DHCP has HTTPSRVR entry as FQDN.15 |
| IPT00057398 | ARP sent every 40 seconds at customer site. |
| IPT00056561 | \"USB Device Detected\" warning displayed on phone screen intermittently, even if no USB device is attached. |
| IPT00057384 | Complete softkey button sometimes not seen during transfer operation. |
| IPT00057390 | When vlanTest set to less than 10, phones move from voice VLAN to Data VLAN. |
| 62378447 | Deskphone fails on ESS interchange scenario. |
| 67306649 | Deskphones SDES information in RTCP packets contains some non-printable characters. |
| 21287449 | Calls establishment fail on 9608 deskphone in SLS mode in G430, on 6.0 SP3. |
| IPT00057011 | No ARQs are sent to ESS when TCP link lost while user is on call. |
| 55524638 | Deskphone loses track of TCP-KA failures when failover happens while on a Call + user presses Switch FBs. |
| IPT00056489 | Audio quality needs an improvement - Audio parameter modifications for speaker device. |
| IPT00056494 | H.323 DHCP release for the data VLAN is broadcast but not unicast. |
| IPT00057367 | DTMF tone stuck during dialing. |
| IPT00057618 | Last feature button is \"duplicated\" and unusable. |

| ID | Issue Description |
|-------------|--|
| IPT00055882 | While transferring or making a conference call the highlight of 2nd call-appr changes to 1st call-appr when CM audit runs. |
| 23171621 | Deskphones have stutter dial tone when going off hook with receiver or SPEAKER. |
| IPT00056619 | Improper call log history display for long numbers. |
| 43241649 | Login fails when restoring truncated _96xxdata.txt |
| 21281549 | Network performance goes down when PC with 1 Gbit port is connected to the PC port of 9641G phone |
| 33307369 | Deskphones do not update ringer after transfer for external calls. |
| IPT00056036 | 9641G deskphones fwd icon not shown on the topline. |
| 32247179 | Very slow performance of WML on deskphones running R6.0. |
| 21290449 | On 9608 deskphones, Japanese Unicode label is not shown correctly with Phone Display Language set to English. |
| 98307269 | \ "enter\" is not seen on VPN login screen using 9608 with German translation |

License Agreements

The 96x1 H.323 6.4 Third Party Terms document is available under the following path:

<https://support.avaya.com/helpcenter/getGenericDetails?detailId=C200922314304731046>

(please scroll to the 96x1 H.323 section)