Avaya 1692 IP Conference Phone H.323 Release 1.4 Readme

This file is the Readme for the Avaya 1692 IP Conference Phone H.323 Release 1.4. This file describes the contents of the **March 2012 Generally Available** software distribution package.

This Release supersedes all previous Avaya 1692 IP Conference Phone H.323 releases and service packs. Avaya recommends that all customers upgrade both new and installed 1692 IP Conference Phones to this version at their earliest convenience.

To upgrade your 1692 IP Conference Phone:

1. Unzip the zip file in the root directory of your HTTP server.
2. Make any adjustments required by your environment to your 46xxsettings.txt file.
3. Reset your 1692 IP Conference Phone.

Avaya Aura® Communication Manager (CM) Compatibility

Although the 1692 IP Conference Phone is supported on Avaya Aura® Communication Manager 4.0 and higher, we recommend using the latest Communication Manager release with this model. See the "Communication Manager Software & Software Compatibility Matrix" at http://support.avaya.com for the supported software/software versions of the Media Server, Media Gateway, and circuit packs.

CM 4.0 and above is required to support the following features:

- Setting Codecs G.711, G.722
- SRTP
- Feature buttons on a softkey

The 1692 IP Conference Phone supports only AES for SRTP (the 1692 does not support signaling encryption).

The 1692 is not natively supported. It is recommended that the 1692 be administered as a 4620SW.

For more details refer to the H.323 configuration section in the Communication Manager Administration Guide which can be downloaded from http://support.avaya.com.

Additional installation information is provided in the Avaya 1692 IP Conference Phone Quick Start Guide which can be downloaded from http://support.avaya.com.

Enhancements with H.323 Release 1.4

Administrative related capabilities:

- **Gatekeeper list** – Support for configuration of Gatekeepers through DHCP/46xxsetting.txt file (MCIPADD) and CM (RAS Gatekeeper lists). *The existing functionality has been expanded to allow up to 7 addresses for MCIPADD, and up to 23 addresses from Communication Manager. This increases the flexibility and reliability of the telephone to find alternative Gatekeepers in cases of network or equipment outages.*
- **Controlled access to configuration parameters** – Allows administration of local Mute Key Sequence procedures depending on the values set using the parameters PROCPSWD and PROCSTAT settings. *This improves security by allowing administrators so customize access to Local Procedures.*
- **GET filename** – Support for use of "GET filename" command in the 1692upgrade.txt and 46xxsettings.txt files. *This gives administrators some flexibility in assigning different parameter values to different telephones or different user communities.*
- **SNMP-MIB** support for newly added features/variables.

Improved interworking with Communication Manager:

- **Registration Request** (RRQ) – Support for software version, vendormanufacturerCode and callPresent reporting in RRQ message in CM acceptable prescribed format. *The first two changes provide more information to Communications Manager for station management. The third change improves the speed and reliability with which the phone can fail over in case of a network or gatekeeper outage.*
- **Unnamed Registration/Terminal Translation Initialization** (TTI) – Support for Unnamed Registration/Terminal Translation Initialization (TTI). *"Unnamed Registration" is when the telephone registers to Communications Manager with a null extension and password. This can be useful for telephones before they are assigned extensions, or when a user has taken over the extension of phone A while at phone B. The telephone that has registered with a null extension and password has only one call appearance, no feature buttons, and is given service by Communications Manager at the level of any TTI-administered telephone. Note that for Unnamed Registration to work, both the IP Conference Phone and Communications Manager must be administered to allow it.*
- **Unregistration Requests** (URQs) – Support for URQ messages with reason codes. *This improves security (since the telephone takes extra precautions to validate the URQ message) and can aid problem solving in that the telephone provides detailed messages explaining why the phone unregistered.*
- **Load balancing** - Support for load balancing. *With basic registration, potentially all H.323 IP telephones would attempt to get service from the same Gatekeeper, even if dozens of other Gatekeepers are available. Communication Manager Gatekeepers have long had the ability to "direct" registering telephones to other Gatekeepers that are under-utilized (hence improving availability and speed of registration by 'balancing the load' across multiple Gatekeepers). With Release 1.4, the 1692 IP Conference Phone no longer ignore this direction, but instead complies with it.*

Avaya 1692 IP Conference Phone H.323 Release 1.4 Package Content

The release package contains all the files necessary to upgrade Avaya new or previously installed 1692 IP Deskphones to Release 1.4.

The following files are included in the release package:

- 1692_014000.bin (application file)
- Bootrom.ld (boot file)
- 1692upgrade.txt (Avaya upgrade script file)
- 000000000000.cfg (XML file required for BootServer process)
- sip_backup.cfg (Required for TLS and Localization)
- phone1_vcvr.cfg (Required for TLS and Localization)
- 1692Localization folder containing supported language folders.
    - French_Canada
    - French_France
    - German
    - Italian
    - Spanish
    - Chinese
    - Japanese
    - Portuguese
    - Russian
- av_prca_pem_2033.txt (Avaya Product Root CA certificate)
- av_csca_pem_2032.txt
- av_sipca_pem_2027.txt
- av_sipca_pem_2011.txt
- release.xml (Release content in .xml format)

The signatures in the signatures subdirectory of the .zip distribution package are only intended to be used by the file server, and the file server that is on the CM 6.x Utility Server is the only file server that currently support this.

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

Note that the “sip_backup.cfg” and “phone1_vcvr.cfg” files are required for TLS and Localization with the 1692 H.323 offering. These files must be present with every upgrade package.

The upgrade procedure requires that all files be placed in the file server folder.

The IP Conference Phone H.323 Release 1.4 package is available in the following versions:

- 1692-IPT-H323-R1_4-030712.zip
- 1692-IPT-H323-R1_4-030712.tar

Issues resolved with H.323 Release 1.4

Issues resolved in this release include:

WI/SSIP number	Issue
SSIP-8499	No audio path after CM reset while shuffling is enabled.
SSIP-8686	1692 phone sends GRQ to AGK list IP's in case of login attempt through unnamed registration Login Softkey.
SSIP-8681	1692 does not display the current active Gate keeper Address in SysInfo when it is registered to alternate Gate keeper.
SSIP-8678	Sometimes even with extension availability phone is performing Unnamed Registration request.
WI00961964 SSIP-8676	SR 1-2228964572 1692 phone running software 1.3 or 1.31 losses audio path when the CM Server with processor Ethernet is interchanged.
wi00974599 SSIP-8690	Newly logged in conference phone, 1692, does not show the administered button-labels.
SSIP-8663	Mute key, Volume Up and Volume Down keys are not working in active call when CM Server with processor Ethernet is interchanged.
SSIP-8667	Add support for GET filename command.
SSIP-8424	VLANTEST 0 does not result in correct phone behavior.
SSIP-8662	PROCSTAT and PROCPSWD features support in 1.4.
wi00969714 SSIP-8683	SR 1-2295357067 1692 phones reset the TCP connection
SSIP-8625	Auto-icom server feature displayed wrongly on phone.
SSIP-8624	Phone did not sent Release complete message in extension take over scenario.
SSIP-8603	Phones are not rebooting when do busy out/release station/port using CM6.0
SSIP-8312	Two Phones are getting register with same extension when unnamed registration is enabled on CM.

SSIP-8452	1692 ignores language settings in the station form and uses default.
SSIP-7960	Phone is unable to go to unnamed registration state.
SSIP-8665	1692 phones are rebooting during calls in 1.32 software.
SSIP-8575	1692 phone does not follow load balancing.
SSIP-8532	1692 does not reboot via CM reset ip-stations command
SSIP-8264	The station doesn't get the correct settings from the IP-NETWORK-Region (CoS settings)
SSIP-8498	Phone screen does not display properly in ideal state after conference call with reset system 4.
SSIP-8597	1692 phones cannot be loaded with new software remotely.
SSIP-8630	Phone is sending GRQ request and register back with CM again.
SSIP-8589	1692 Reregisters, drops call and unregister with TTI enabled.
SSIP-8635	1692 dropping active call.
SSIP-8637	1692 not trying the LSP in failover scenario.
SSIP-8531	Remote reset of the 1692 results in the phone freezing.
SSIP-8570	Phone displays the "No File server Address * to program" message even when it gets file servers IP from DHCP when do Mute Restart after Mute RESET.
SSIP-8301	Server feature cursor does not get updated when remove few features from ACM while the phone is registered to it.
SSIP-8373	Team button is displayed as {Null}, if logoff and login the phone after adding this server feature.
SSIP-7649	Gatekeeper Request (GRQ) Issue-: Phone is not sending GRQ request in consistent time gap, when it goes in discovery mode. *Note: Phone functionality does not have any impact and is more of a statistical observation.*

SSIP-8323/SSIP-7800	Redial Key is not retaining the last dialed number call made by Directory feature.
SSIP-7730	Need to do Mute sequence two times to get the required screen in certain scenario.
SSIP-8309	'Password error' and 'Extension in use' message for all the supported languages other than English is not properly translated.
SSIP-8311	Phone hangs in extension take over scenario and required hard reboot to make operational.
SSIP-8312	Two Phones are getting registered with same extension when unnamed registration is enabled on CM
SSIP-8251	1692 is not taking DIFFSERV/TOS PARAMETERS in the network region from CM
SSIP-7961	Phone is showing "Resource Unavailable" when registered with a CM with CLANS configured.
SSIP-8314	Phone does not display the ringing call on the phone UI when phone is on log off screen
SSIP-8315	Phone displays two feature is highlighted at the time on the screen.
SSIP-8318 SSIP-7715 (Linking issue)	Phone does not generate the error tones.
SSIP-8319	Phone always contact to last ip address given in the list.
SSIP-8340	Sometimes phone hangs on the login screen in a particular scenario.
SSIP-8409	Phone doesn't return to ideal state after conference call while one call is dropped by 1692
SSIP-8264	The station doesn't get the correct settings from the IP-NETWORK-Region (CoS settings)
SSIP-8575	1692 phone's does not follow load balancing.
SSIP-8579	1692 does not support TTI/Unnamed registration

SSIP-8542	Discovered new vlan configuration screen sometimes is displayed momentarily at the booting up screen when L2QVLAN and L2Q set in settings file and Vlan Name TLV is defined.
SSIP-7840	Phone does not attempts to contact all the next DNS Server in the list as defined in DHCP when the first DNS server responds back with "no such name" message. *Note: Working as per the DNS standard specification. 1692 is designed as per that specification.*
SSIP-8324	Phone accepts more than 13 digits on the dialing screen

Unresolved issues in H.323 Release 1.4

WI/SSIP number	**Issue**
SSIP-8689	Oasis 1692: When user goes off-hook and then starts accessing Menu>>Ringing Patterns (without dialing any number) simultaneously listening the ring tones while dial tone is being played out notices an increase in dial tone volume. Note: Corner scenario of an issue trivial in nature and will not fix in R1.4 cycle.
SSIP-8256	*Phone is retaining the Time server IP even if that parameter is deleted from the DHCP server.* *Note: Phone functionality does not have any impact as this is not a mandatory parameter. Phone anyways takes time from ACM.*
SSIP-8295	*LLDP: Phone is not losing its connectivity with the CM if change the VLAN in Network policy TLV and remains registered after dropping the call.* *Note: Phone functionality does not get impacted.*
SSIP-8304	*Phone is not displaying the 802.1x Failure screens if disabling the ID on the authentication server when in Restart confirmation screen.* *Note: More of a cosmetic issue and also a rare scenario where a phone successfully authenticated to the 802.1x server with EAP ID and password is disabled only when the phone is in Restart confirmation screen.*
SSIP-8535	Phone displays enter extension screen if do Mute ADDR# after authenticating the phone with a different 802.1x user id and password.
wi00973379 SSIP-8685	1-2307490335 / PEA 1-12BDE27 - SET HTTPDIR not retained during boot up
SSIP-8684	SR 1-2316459151 1692 phone - language files not downloaded when using https as fileserver type
SSIP-8252	1692 is not taking signalingChannelRecovery values from CM ip-network region form
SSIP-8247	1692 with software version R1.0 1692 is drawing the maximum power on a POE switch and that its class type not identifiable.
SSIP-7963	Phone is getting a GRJ with a Reject Reason "Security Denial" during registration with definite configuration on ACM

SSIP-7959	Phone shows confusing messages.
wi00974613 SSIP-8691	The 1692 (1.31) phones are showing the auth code after a redial on the screen.
wi00979512 SSIP-8698	Phone goes to 'Unnamed Registration' after pressing and deleting one dial-pad button in EXT prompt and leaves for one minute.
wi00979516 SSIP-8699	The 'unnamed registration' timer resets if pressing button other than '0-9 or #'.
wi00974841 SSIP-8693	1692 does not work with Unicode setting on CM station page.

Advisement – Upgrades/Downgrades to/from H.323 Release 1.0

Upgrades from H.323 Release 1.0 and H.323 Release 1.3 or later using HTTPS will fail due to changes in the default port from 443 to 411.

Workaround:

- Change TLSPORT via DHCP-SSON and set explicitly to 411.
- Upgrade from H.323 Release 1.0 to Release 1.2. Perform MUTE Clear on Release 1.2, use a static IP address and then upgrade to Release 1.3 or later.

When downgrading to H.323 Release 1.0, an "Error loading <mac>.cfg" error is displayed.

Workaround:

- Reboot the 1692 and re-enter the IP address.

License Agreements

Notice

While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document may be incorporated in future releases.

Documentation disclaimer.

Avaya Inc. is not responsible for any modifications, additions, or deletions to the original published version of this documentation unless such modifications, additions, or deletions were performed by Avaya.

Link disclaimer

Avaya Inc. is not responsible for the contents or reliability of any linked Web sites referenced elsewhere within this Documentation, and Avaya does not necessarily endorse the products, services, or information described or offered within them. We cannot guarantee that these links will work all of the time and we have no control over the availability of the linked pages.

License

USE OR INSTALLATION OF THE PRODUCT INDICATES THE END USER'S ACCEPTANCE OF THE TERMS SET FORTH HEREIN AND THE GENERAL LICENSE TERMS AVAILABLE ON THE AVAYA WEBSITE AT http://support.avaya.com/LicenseInfo/ ("GENERAL LICENSE TERMS"). IF YOU DO NOT WISH TO BE BOUND BY THESE TERMS, YOU MUST RETURN THE PRODUCT(S) TO THE POINT OF PURCHASE WITHIN TEN (10) DAYS OF DELIVERY FOR A REFUND OR CREDIT. Avaya grants End User a license within the scope of the license types described below. The applicable number of licenses and units of capacity for which the license is granted will be one (1), unless a different number of licenses or units of capacity is specified in the Documentation or other materials available to End User. "Designated Processor" means a single stand-alone computing device. "Server" means a Designated Processor that hosts a software application to be accessed by multiple users. "Software" means the computer programs in object code, originally licensed by Avaya and ultimately utilized by End User, whether as stand-alone Products or pre-installed on Hardware. "Hardware" means the standard hardware Products, originally sold by Avaya and ultimately utilized by End User.

License Type(s):

Designated System(s) License (DS). End User June install and use each copy of the Software on only one Designated Processor, unless a different number of Designated Processors is indicated in the Documentation or other materials available to End User. Avaya June require the Designated Processor(s) to be identified by type, serial number, feature key, location or other specific designation, or to be provided by End User to Avaya through electronic means established by Avaya specifically for this purpose.

Copyright

Except where expressly stated otherwise, the Product is protected by copyright and other laws respecting proprietary rights. Unauthorized reproduction, transfer, and or use can be a criminal, as well as a civil, offense under the applicable law.

Third-party Components

Certain software programs or portions thereof included in the Product may contain software distributed under third party agreements ("Third Party Components"), which may contain terms that expand or limit rights to use certain

portions of the Product ("Third Party Terms"). Information identifying Third Party Components and the Third Party Terms that apply to them is available on Avaya web site at: http://support.avaya.com/ThirdPartyLicense/

Trademarks

Avaya, Avaya Aura, Communications Manager and the Avaya logo are registered trademarks and/or service marks of Avaya Inc. in the U.S. and certain other jurisdictions.

All other trademarks are the property of their respective owners