



Avaya one-X® Deskphone SIP Release SIP 2.6.8 Readme

This file is the readme for the Avaya one-X® Deskphone SIP Release 2.6.8 software which runs on the following 9600 Series IP Deskphones:

- 9620, 9620C, 9620L
- 9630, 9630G
- 9640, 9640G
- 9650 and 9650C.

This document describes the contents of the Release 2.6.8 software distribution package that can be found at <http://support.avaya.com>

Avaya one-X Deskphone SIP 2.6.8 is software for the 9600 IP Telephones that supports the SIP protocol. This release is recommended for installations with Avaya Aura® Session Manager 6.0 and later, with an Avaya Aura Survivable Remote, Avaya Aura Communication Manager 6.0 and later and also with installations with SIP Enablement Services 4.x or 5.x with Communication Manager 4.x or 5.x.

Avaya one-X® Deskphone SIP 2.6.8 improves quality of SIP 2.6.8 and delivers Contact Caching feature and improved support of 2 SBM24.

This maintenance release supersedes all previous one-X® Deskphone SIP releases. Avaya recommends that all customers upgrade both new and installed IP Deskphones running one-X® Deskphone SIP 2.x software to this version at their earliest convenience.

New Features in Deskphone SIP 2.6.8

Feature	Feature description
Contact Caching	A local copy of contacts is always kept on the phone so that in non-failover scenarios when the phone receives a partial download from PPM, users will always see all their contacts. Users are able add/edit/delete their cached contacts.
Presence	Presence is enabled by default. Please refer to the user/admin guide for details how to use this feature

Fixed Issues in Deskphone SIP 2.6.8

Area	Defect ID	Defect Name
Performance	SIP96X0-7743	96x0 sometimes crashes after 10000 calls
	SIP96X0-7606	Phone reboots because of exception in tWindowManage task.
	SIP96X0-4963	Phone does not resolve the domain name given in Craft-LOG menu for SysLog server.
	SIP96X0-7665	When changing from SM1/SM2 to SM3/SM4 using SMGR the phone sometime freezes.
	SIP96X0-7612	The phone rebooted after pressing OK after getting the error "Unable to fetch a logo" message.
	SIP96X0-7352	The phone does not transmit gateway IP address in LLDP packet to switch.
	SIP96X0-7654	The phone reboots an additional time when loading file EXTEND_RINGTONE language KoreanRT8.xml.
	SIP96X0-7626	The phone crashes under heavy traffic.
	SIP96X0-7650	Call button on Phone does not work after unsuccessful call and when phone is on-hook.
	SIP96X0-7653	Sometimes phone is rebooted due to an exception in tDualUnicast task.
	SIP96X0-7609	The phone with 3 BM cannot use debug level for logging.
	SIP96X0-7604	Phone reboots every time when it receives a BYE from AAC behind a SBC.
	SIP96X0-7616 SIP96X0-7598	Resolved various audio issues, potential memory leaks, memory corruption issues.
	SIP96X0-6774	The phone has a memory leak.
	SIP96X0-5954	Memory leak while downloading HOME / IDLE WML UIs with parse error.
	SIP96X0-6017	Answering a call over SIP trunk using CTI application causes the phone to cash.
Configuration	SIP96X0-7719	Phone loses "MSGNUM" configured when use logs out and logs back in.
	SIP96X0-4731	When adding an invalid proxy server address statically and then deleting it, the top line shows failover icon.
	SIP96X0-7680	User is able to delete SIP proxy address when it received from settings file and PPM.
	SIP96X0-7661	Proxy order is not valid.
	SIP96X0-7657	9640 does not accept the DST settings when DSTSTAT value is higher than DSTSTOP value.
	SIP96X0-4547	PHY1_OPEATIONAL_MODE and PHY2_OPEATIONAL_MODE parameters are not present in MIB and phone still responds to PHY1SPEED and PHY2SPEED.
	SIP96X0-5747	User settings should be changed to default when logged in with another extension on third party controller.
	SIP96X0-7217	Phone doesn't restart after Phone IP address is changed followed by changes in SIP proxy settings (yellow triangle show up).
	SIP96X0-7470	96x0 SIP phone does not reboot after "IP address" change followed by change in "Debug" mode.
Audio	SIP96X0-7681	SRTP call becomes RTP after Hold and un-Hold with Network Call Redirection (NCR) and Music On Hold (MOH) enabled.
	SIP96X0-7595	Sometimes phone displays error message on RTCP statistics

		collection.
	SIP96X0-7718	A volume level for 6 event handset on wideband calls is 3dB too quiet and wideband equalizer is not properly applied.
	SIP96X0-7627	During the call phone transmits RTP to wrong address (stale RTP session); call ends up in one way conversation.
	SIP96X0-7629	Phone gets no audio.
	SIP96X0-7634	One Way talk path in attended transfer and conference without Music on Hold.
	SIP96X0-7518	Voice quality with 9650 phones.
	SIP96X0-6741	Conference leg doesn't have voice path in I55 and conference Initiator reboot.
	SIP96X0-7424	During shuffling 9620 does not transmits/receives RTP for approximate 0.6 seconds.
Failover\Failback	SIP96X0-7694	A phone doesn't failback after a call.
	SIP96X0-7514	Phone cannot failback from third controlled to the primary if SIMULTANEOUS_EGISTATIONS is 2 and failback was pended because of active call.
	SIP96X0-7466	Call Volume is increased automatically after failover from SM1.
Push\WML	SIP96X0-7755	Line text of "Alt" attribute cannot exceed 1 line (around 28 letters).
	SIP96X0-7735	Phone cannot display "Push" during Acquiring Service.
	SIP96X0-5185	Alerting is heard even though it has been made off, after after canceling an audio push when phone is getting incoming call.
	SIP96X0-7541	96x0 crashes when loading helloworld.wml.
	SIP96X0-7295	Can't stop loading WML file when pressing stop soft key.
	SIP96X0-4824	If the pushed language does not exist on the phone, then the phone language doesn't change to English by default.
PPM	SIP96X0-7594	Phone supports and processes additional PPM error codes.
	SIP96X0-7590	Phone incorrectly responds to Reload Configuration 1 and Reload Configuration 0.
	SIP96X0-7521	Problems with user data handling in PPM Adaptor.
	SIP96X0-7382	Phone displays two extension of call forward while only configured one extension on PPM.
SBM24	SIP96X0-7748	96x0 with 3 BM configured as Busy-Indicators and assigned to pick up group reboots after incoming call.
	SIP96X0-7722	Phone left with low free memory when connected to 3 SMs and has 3 SBM24 with 72 busy indicators.
	SIP96X0-7690	Optimization for button module performance.
	SIP96X0-7610	Phone with 3 SBM stops working in 1-3 days.
	SIP96X0-3854	The character "W" gets truncated in the middle if 13 characters are assigned as personalized label for a feature displayed on SBM24.
Contacts	SIP96X0-7711	Phone does not display number of H323/Digital/Analog contacts which are added from SMGR.
	SIP96X0-7625	Phone hangs and throws exception while invoking more than 400 contacts uploaded on phone though SMGR.
	SIP96X0-7588	A local copy of contacts is always kept on the phone's Flash so in case of 500 serve errors – uses will always be able to see their contacts.
	SIP96X0-7677	Cannot modify/delete contact when SM is unavailable
	SIP96X0-7676	Cannot move up/down in the contact list when SM is unavailable.
	SIP96X0-7614	Remove clearing of contacts cache in the phone in case of 500

		server errors.
User Interface	SIP96X0-7544	Sometimes the phone does not switch immediately from the "Login" screen to the "Logging in..." screen after entering login and password.
	SIP96X0-7729	Limited phone service when double pressing the "Enter" soft key quickly.
	SIP96X0-7669	Energy Saver display activates without user intervention.
	SIP96X0-7205	96x0 SIP phone displays wrong incoming call number if call is coming from CS1K SIP Phone.
	SIP96X0-768	Prompt line has empty white line after returning from screensaver.
	SIP96X0-7646	Phone is not logged in but if the user presses the phone button then it goes to phone screen as if the phone is registered.
	SIP96X0-7584	Topline isn't cleaned when a phone switches on Login screen.
	SIP96X0-5635	Prompt line is in yellow color when null.
	SIP96X0-6553	Top line is not refreshed in the failover scenario.
	SIP96X0-6284	Phone displays a "name" that is the same as the dialed "outside" number.
	SIP96X0-4948	Call appearance becomes non-operational for sometime if we are dialing only @ character.
	SIP96X0-6881	Phone should not show "Limited Phone Service" if it is not registered.
	SIP96X0-2154	Server Link Down obscures screen; call cannot be dropped.
	SIP96X0-4484	Login screen - Password text entry box shows exceeded characters.
	SIP96X0-7403	Only 56 characters are allowed while default maximum length is 200.
	SIP96X0-7513	Change Avaya copyright notes in About Avaya one-X
	SIP96X0-7525	Text doesn't get truncated properly when the language is Hebrew and text entry is done.
	SIP96X0-3643	The soft key text exceeds its border length if given a long soft key label.
	SIP96X0-5987	Previously entered characters in exchange password are displayed for Hebrew language on entering new character.
Signaling	SIP96X0-7679	The phone does not register if the Settings File has 2 SM Controllers and Use Profile has only one Controller.
	SIP96X0-5504	Re-use the authentication of 401 of REGISTER request for "all" subsequent SIP requests (SUBSCRIBE, INVITE, PRACK and BYE etc.).
	SIP96X0-7503	When SIP96x0 receive an IPv6 address in a INVITE/re-INVITE it replies with 200OK (need to reply with a 488 response).
	SIP96X0-7774	Phone fails to fall over from SAVP to AVP when 480 response received after 181 call is being forwarded.
Conference\Transfer	SIP96X0-7624	Phone attempts transfer before dialing the transferee target.
	SIP96X0-5186	Conference initiator is always forwarding the received DTMF with volume level 15 if any key is pressed on conference leg.
	SIP96X0-7523	Conference issue on 96xx phone.
Features	SIP96X0-7662	Bridge appearance doesn't work after off hook/on hook.
	SIP96X0-7583	Telephone doesn't follow dial plan rules after editing contact.
	SIP96X0-6622	Phone should compare dialed string against Dial plan even if dial using Edit Dialing.
	SIP96X0-5757	Hold call timer: Time is displayed more than 9:59:59.

	SIP96X0-3168	Call Time is not behaving as required.
	SIP96X0-7543	PTI key definitions don't exist for 9620L.
	SIP96X0-7528	Phone with configured emergency number sometimes does not allow dialing it in the locked state and displays 'this telephone is locked'.
	SIP96X0-7524	SIP phone has issues with EC500.
	SIP96X0-7660	96x0 repetitively downloads the logo.
	SIP96X0-5920	Exchange integration fails, calendar SK not displayed, exchange server IP is 0.0.0.0.
	SIP96X0-7483	Speaker auto turn on when On Hook and Off Hook consecutively.
	SIP96X0-7480	After pressing emergency button at login screen, phone screen is shown if there are no active controllers.
	SIP96X0-7417	Incorrect BCA behavior when there are multiple bridged appearances.
	SIP96X0-7468	"vclass" should be "Avaya.Phone" to align with presence model document.
	SIP96X0-7511	Presence should be enabled by default.
Call Log	SIP96X0-6016	First & Last name reversed on add contact from call log.
	SIP96X0-7451	SIP phone log caller's (transferring) name and transferred phone number.
Languages	SIP96X0-7536	Latest language file should be used.
	SIP96X0-7510	Some options in Audio Parameters are not translated to Hebrew language.
EAP	SIP96X0-7512	Phone does not renew certificate from certificate authority as the value set in MYCETENEW parameter.
	SIP96x0-7495	Phone authorization EAP-TLS failed.

Known Limitations of Release 2.6.8

❖ Recommended SIP Transport Protocols

SIP Proxy / Phone Connection Type	UDP	TCP	TLS
SES (as primary controller)	Not Recommended	Recommended	Recommended
Avaya Session Manager (as primary controller)	Not Recommended	Recommended	Recommended
Avaya Secure Router 2330 and 4134	Not Recommended	Recommended	Recommended
AudioCodes MP-series analog and BRI gateways (as secondary controller)	Not Recommended	Recommended	Not Recommended
Cisco 2811 ISR (as secondary controller)	Not Recommended	Recommended	Not supported
I55 (as secondary controller)	Not Recommended	Recommended	Not supported
Teldat Vyda gateway	Not Recommended	Recommended	Not Recommended

Table of Phone Connection Types for Survivability Configurations

'Recommended' means minimal latency in the detection of a Failover condition

- The Audiocodes SIP gateways tested for interoperability are the MP114 and MP118; the recommended firmware revision is 5.60A.010.005 or later.
- The Cisco gateways tested for interoperability are the Cisco [ISR] ; the minimum firmware revision is c2800nm-ipbasek9-mz.124-20.YA2.bin
- ❖ When the phone is failed over to a non-Avaya secondary controller (AudioCodes/Cisco): We expect all the phones in the environment are operating in same transport protocol. There may be a few call-based issues if the phones are using different transport protocols.
- ❖ SRTP
 - The Avaya Gateway firmware version must support SDES SRTP
 - Crypto suites must match
 - Requires CM 4.0.1 or greater for SM and CM 5.1 or greater for SES
 - Always provision 'None' encryption (setting 9) as one of the encryption selections in the settings file and on CM
- ❖ Configuration with 3 BMs and few dozen Busy Indicators \ Bridged Appearances administrated
 - The phone may registered in non-AST mode
 - The Phone may crash under heavy traffic
- ❖ NAT
 - 96xx SIP telephones should not be provisioned behind a NAT with private network addresses when the CM and SES or SM switching fabric are provisioned on a different network.
- ❖ Recommended Failback Policy

Admin failback not recommended in R2.6 and Auto failback policy is recommended.

Known Issues in Release 2.6.8

Area	Defect ID	Issue	Workaround
Failover \ Failback	SIP96X0-7798	If presence enabled and one of the SIP proxy is AudioCodes proxy then phone will crash if SIMULTANEOUS_REGISTRATIONS count only SM servers	If presence isn't required then turn it off (ENABLE_PRESENCE 0) Otherwise SIMULTANEOUS_REGISTRATIONS should show the number of all SIP proxy (including AudioCodes SIP proxy). Example: if SIP_CONTROLLER_LIST: SM1, SM2, AudioCodes then SIMULTANEOUS_REGISTRATIONS should be 3 and not 2 (this workaround will not work if you have 3 SMs or more).
Upgrade \ Downgrade	NA	Deskphone SIP 2.6 can only be loaded onto a 9620C or 9620L running H.323 firmware if either - The 9620C or 9620L is running Deskphones H.323 3.0 or later; or 96x0Hupgrade.txt includes hb96xxua3_00.bin	
Configuration	SIP96X0-3992 SIP96X0-1337 SIP96X0-2339	Any configuration parameter that is set through the CRAFT menu to its default value will be ignored and the value set via DHCP or 46xxsettings file will be used.	To explicitly set default values for parameters, set them in DHCP or the 46xxsettings file.
Features	SIP96X0-7640	Hospitality: Function of hard buttons cannot be changed	This is not supported by 2.6.8 and earlier service packs.
The Avaya one-X® Communicator Deskphone Mode	SIP96X0-7803	The Avaya one-X® Communicator Deskphone mode will experience log in issues if we set PPM_SOURCED_PROXY_SERVER 0 in 46xxsettings file of the Phone.	To avoid this issue do not change the default value of PPM_SOURCED_PROXY_SERVER in 46xxsettings file.
	SIP96X0-7740	96x0 SIP phone that has a 1 BM configured and One-X communicator logged in a shared control mode goes out of memory at the traffic (One-X issue).	Do not use high feature traffic like as Busy Indicators off\on.
	SIP96X0-7726	One-X communicator in a shared control mode with 96x0 phone, that has a 3 BM configured does not work (One-X issue).	Use 1 SBM24 only.
EAP	SIP96X0-7752	Authentication is failed with unicast mode.	When user modifies EAP settings on the phone (or using 46xxsettings.txt) then need to reboot the phone manually.
SBM24	SIP96X0-7751	Phone does not clear 3 SBM24 (Busy Indicators) while failover\failback to 3rd party proxy.	
WML	SIP96X0-7746	Phone restarts when loading WML file smaller 250KB.	Do not use more than 100 cards in the single WML page.

Presence	SIP96X0-7539 SIP96X0-7538 SIP96X0-7535 SIP96X0-7534	On 96x0 endpoint presence status doesn't changed after PS restart (server side issue).	Logout and login the user.
SRTP	SIP96X0-7787	Phone lockup or get noise after answer a call with MEDIAENCRYPTION mismatch with CM	Do not use configuration mismatch between phone and CM

Feature Compatibility Matrix for Communication Manager (CM) and SIP Enablement Server (SES)

A number of the new features in Release 2.6 and later are available only on recent CM and SES versions as shown in the table below.

	Minimum CM Version Required	Minimum SES Version Required
SBM-24 button module – Feature button assignments and operation	5.0	5.1
SBM-24 button module – Autodial assignments and operation	5.0	5.1.2
SBM-24 button module – Bridged Call Appearance (BCA) and Call Appearance (CA) operation	5.1.2	5.1
Auto Answer	5.0	5.0
Deactivation of Call Forward On Busy Do Not Answer	5.2	5.2
Support for up to ten emergency numbers	6.2	N/A (only a single emergency number is supported)

Features from prior SIP phone releases that require recent CM and SES versions are shown in the table below:

	Minimum CM Version Required	Minimum SES Version Required
Visiting User (Applicable for SES environment only)	5.0	5.0
Extend Call	5.0	5.0
SRTP	4.0.1	n/a
All features other than those noted in this these tables	4.0	4.0

96xx SIP Release 2.6.8 (SIP 2.6.8.4) Package Contents

The package contains the readme file and all the files necessary to upgrade Avaya 9600 IP telephones to 96xx SIP Release 2.6.8. The following files are included in each package

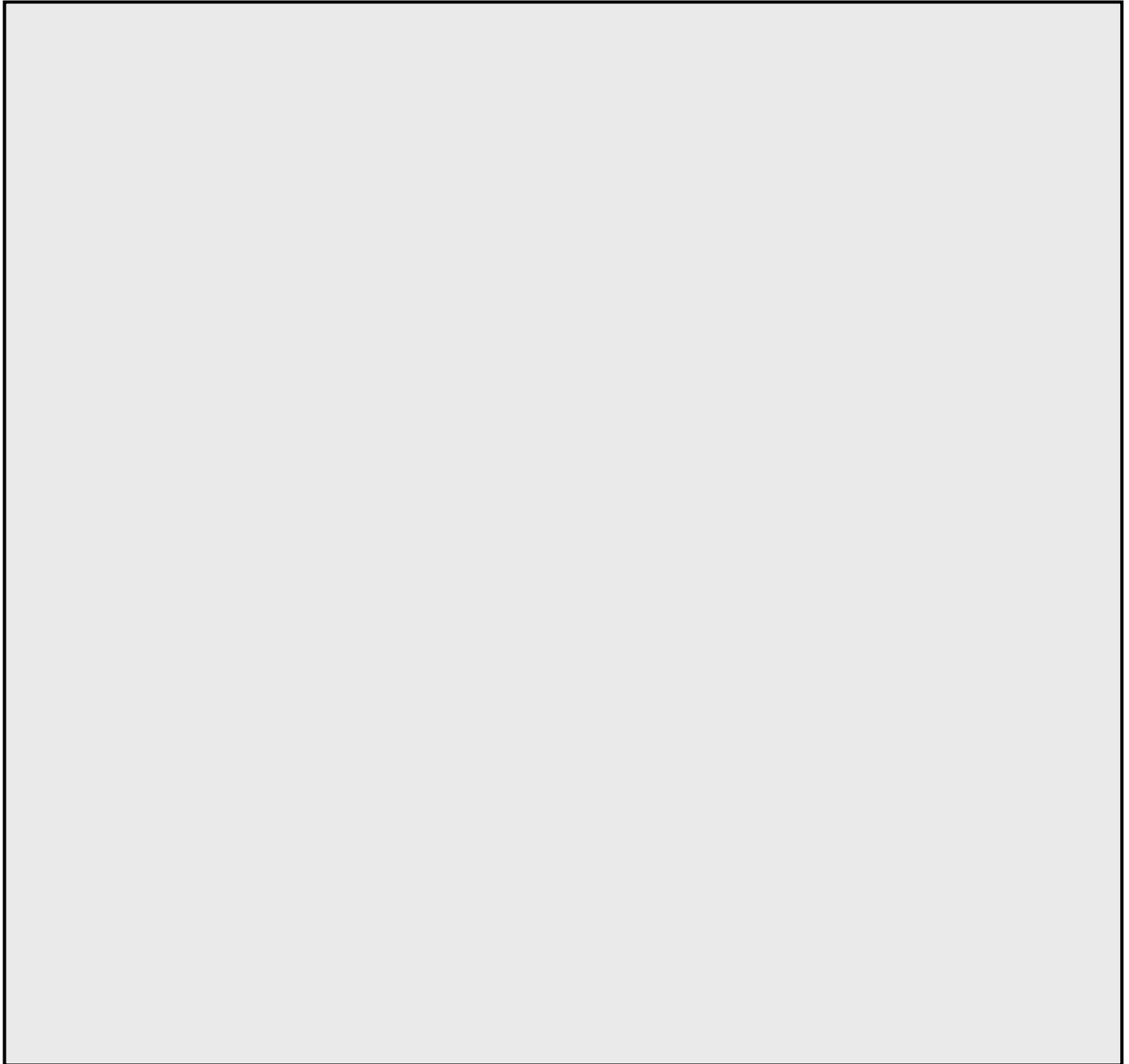
- ❖ hb96xxua3_00.bin – The 9600 R2.6 SIP boot application
- ❖ SIP96xx_2_6_8_4.bin – The 9600 R2.6.8 SIP phone application
- ❖ 96xxupgrade.txt – This file is downloaded by the Avaya 9600 IP telephones and instructs the telephones on how to upgrade. DO NOT EDIT this file. You MUST USE the 96xxupgrade.txt file included in this package to upgrade to R2.6.8
 - **Note that this file is designed for environments in which all 9620, 9620C, 9620L 9630, 9630G, 9640, and 9640G endpoints will be upgraded to the 9600 R2.6.8 SIP phone application.**
 - **Alternative 96xxupgrade file for mixed H.323 and SIP 96xx deployments** - Available at support.avaya.com the alternate_96xxupgrade.txt file is designed for environments that will support 96xx upgrades for both the H323 and SIP modes of operation. See the Avaya support website for additional information for how to edit this file and setup the SIG setting (the Signaling Procedure) in the 96xx phones.
- ❖ release.xml – An XML-format text file that is designed to be used with Avaya Integrated Management (AIM) through the Software Update Manager (SUM).
- ❖ Fifteen predefined language files:
 - Mlf_Arabic.xml
 - Mlf_BrazilianPortuguese.xml
 - Mlf_CanadianFrench.xml
 - Mlf_CastilianSpanish.xml
 - Mlf_Chinese.xml
 - Mlf_Dutch.xml
 - Mlf_English.xml
 - Mlf_German.xml
 - Mlf_Hebrew.xml
 - Mlf_Italian.xml
 - Mlf_Japanese.xml
 - Mlf_Korean.xml
 - Mlf_LatinAmericanSpanish.xml
 - Mlf_ParisianFrench.xml
 - Mlf_Russian.xml
- ❖ Eight extended Korean ring tone files:
 - KoreanRT1.xml
 - KoreanRT2.xml
 - KoreanRT3.xml
 - KoreanRT4.xml
 - KoreanRT5.xml
 - KoreanRT6.xml
 - KoreanRT7.xml
 - KoreanRT8.xml
- ❖ Four certificate files:
 - av_csca_pem_2032.txt – Avaya Call Server CA certificate with an expiration date of 2032

- av_prca_pem_2033.txt Avaya Product Root CA certificate with an expiration date of 2033
- av_sipca_pem_2027.txt Avaya SIP Root CA certificate with an expiration date of 2027

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

To upgrade your Avaya 9600 IP Telephones to 96xx SIP Release 2.6.8:

- ❖ Unzip executable file in the root directory of the outbound file directory of your HTTP server.
- ❖ Make any adjustments required by your environment to your 46xxsettings.txt file.



- ❖ Reset your Avaya 9600 IP Telephones.

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