Date: March 02, 2012

Avaya one-X® Deskphone SIP for 9600 Deskphones Release SIP 2.6.7 (2.6 SP7) Readme

This file is the readme for the Avaya one-X® Deskphone SIP release 2.6 Service Pack 7 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 Service Pack 7 software distribution package.

Avaya one-X Deskphone SIP 2.6 Service Pack 7 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 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 Service Pack 7 improves quality of SIP 2.6 SP6 and delivers improved Hearing Aid Compatibility and support for special handling of up to ten Emergency Numbers. Service Pack 7 also has new built-in “Best Fit” memory management that reduces memory fragmentation in VxWorks 5.5.1.

New Features in Deskphone SIP 2.6.7

Feature	Feature description
Hearing Aid Compatibility (HAC)	Hearing Aid Compatibility (HAC) allows for audio performance when IP Deskphones are used in conjunction with hearing aids. Providing an acceptable level of audio quality for users without a hearing aid may conflict with meeting the stricter HAC criteria. With this service pack, a configurable option has been introduced which allows the choice of tuning the handset to meet either TIA 810/920 audio standards or FCC Part 68 HAC standards.
Support for up to Ten Emergency Numbers	If your Deskphone is connected to Session Manager 6.2 and your administrator has configured the settings file for your Deskphone or System Manager for your extension with the emergency numbers, you can dial up to ten emergency numbers even if the Deskphone is unregister or locked. Anytime the emergency button is displayed, pressing it will dial the first emergency number provisioned by System Manager or the number in the PHNEMERGNUM specified in the settings file. Note that more emergency numbers can only be provisioned thru PPM.

New Features in Deskphone SIP 2.6

Setup Environment	Feature	Feature description
Common Feature List Available with all versions of CM, SES and SM.		
Support for Presence	Presence	Starting Service Pack 6 and above, the phone will be able to track the presence status of individuals through the Contacts list
SDP Negotiation	Payload size negotiation	Enables the phone to negotiate payload size with Communication Manager for the RTP packets.
All CM/SES/SM setup	Updated LLDP support	Updated handling for LLDP MED Network Policy TLVs with Application types 1 and 2 or with a zero value improves inter-operation with recent releases of Cisco IOS.
All CM/SES/SM setup	Support HTTP 1.1 for connection reuse	Avaya one-X Deskphone SIP shall download information from PPM using HTTP 1.1 so that a single connection is used for all downloads during the phone's login sequence.
All CM/SES/SM setup	Identification of the type of proxy	The phone can identify the type of registered proxy server (e.g., SES/Session Manager/Branch Session Manager/Generic proxies such as Audiocodes, Cisco 2811 ISR, I55).
All CM/SES/SM setup	Added Support for third-party MWI	When used with CM 4.0+, Deskphone SIP shall properly handle a PPM messages allocating one or more aut-msg-wt buttons and, when an MWI notification is received for other than the primary extension, sets the Message Waiting Indicator on the corresponding button.
Feature List Applicable for CM 6.0+ environment		
Requires CM 6.0+	Call pickup alerting.	When used with CM6.0+, Deskphone SIP alerts (ring or flash) when a phone in a provisioned call pickup group alerts.
Requires CM 6.0+	Improved Call Unpark behavior	When used with CM6.0+ and either SES5.2 or SM6.0+, a call unpark button should only appear on phones provisioned on CM or SMGR.
Requires CM 6.0+	Support SIP Direct Media.	Avaya one-X Deskphone SIP R2.6 and later release shall support Direct Media as supported by CM 6.0 and later.
Requires CM 6.0+	Support SDP Capability Negotiation.	Avaya one-X Deskphone SIP2.6 and later shall support SDP capability negotiation as defined by

		draft-ietf-mmusic-sdp-capability-negotiation-10.txt and supported by CM 6.0 and later.
Features Applicable SM 6.0+		
Requires SM 6.0+	Handling of a redirect response to a REGISTER request.	When a user is moved from Session Manager #1 to SM #2, if a phone tries to register the user on SM #1, it responds with a redirect to SM #2. This means that a user will be able to login after the home SM is changed, but before the information is propagated to System Manager.
Requires SM 6.0+	Voice mail destination can be set in System Manager or Communication Manager	The voice mail destination number can be set in System Manager for individual users instead of administering it in the settings file.
Requires SM 6.0+	Improved user-friendly phone number display format	For example, a telephone number such as “5381234” the display portion of PAI or Contact header might be formatted in a more readable fashion as “538-1234” before it is displayed on the phone.
Requires SM 6.0+	Support geo-redundant Session	Ensures that a telephone user can always receive incoming calls even when another call is active and the call controller has changed due to maintenance, network or server failure. Without this change, an incoming call would be dropped if it is received after a controller switch happens while a user is on another call. When the SIPREGPROXYPOLICY parameter is “simultaneous,” the phone accepts an incoming INVITE from any of its Available Controllers, regardless of which one is active.
Requires SM 6.0+	Simultaneous registration with up to 3 proxies.	Avaya one-X Deskphone SIP 2.6 and later shall simultaneously register with the three highest priority registrars specified by any of the supported provisioning mechanisms.
Requires SM 6.0+	New extended call pickup behavior	An extended call pickup button will appear on phones provisioned for extended call pickup on CM.

List of fixes that have gone in Deskphone SIP 2.6.7

Area	Defect ID	Defect Name
Performance	SIP96X0-7293	Phone reboot when access a WML file size bigger than or equal 347 KB.
	SIP96X0-7167	9650 gets stuck and crashes if we try to load the logo file 2-3 times.
	SIP96X0-7131	Phone gets stuck displaying Logging in if one of the controllers responds with 401 (no credential for the user).
	SIP96X0-7365	Propagation of DE950/DE1028 SBM Memory Leaks.
	SIP96X0-375	m_nNumActiveCalls >= 0 exception in Avaya ISPEAC code.
	SIP96X0-6929	96xx SIP VLAN membership traffic violation.
	SIP96X0-7390	Propagation of the CArray* memory leak activity: part 2 (propagation of CR-CLIENTPLATFORM-355 and CR-SIP96X1-1289).
	SIP96X0-7269	High memory fragmentation is seen on the 96x0 set that is getting busy-indicator traffic (3 SBM configured, 1 SBM connected).
	SIP96X0-7429	96x0 sip phone stuck on login screen when user profile is in one of the PROXY in SIP_CONTROLLER_LIST.
	SIP96X0-7425	Phone sometimes reboots when loading WML file bigger than 250K and smaller than 1MBs.
	SIP96X0-7477	Phone hangs up during start-up if secondary SM is available and primary is not
	SIP96x0-7433	Phone reboots when failing over from SM1 with some feature of CM in SBM24.
	SIP96X0-7432	Phone is getting in hung when is idle with WMLIDLETIME value.
	SIP96X0-7431	Phone suspended in Active call after deny services on SM1 and SM2.
	SIP96X0-7486	9630 receives TCP socket reset from SES while trying to subscribe to CM features. The phone will not try again to subscribe and will boot up with no AST.
Configuration	SIP96X0-7299	File Server IP address is not advertised into LLDPDU packets sent to switch by some models of Avaya phones: 9630, 9620L, 9630G, 9640, 9620C.
	SIP96X0-6715	User is not able to enter IP address or DNS name in 'Host to Ping' field in a particular scenario.
	SIP96X0-7418	Invalid (broadcast) IP address advertised by the IP Phone when multiple call servers addresses are configured on the switch.
	SIP96X0-7415	96xx SIP HTTP GET issue with port 81.
	SIP96X0-7414	Add option to remove the HOLD soft key.
Audio	SIP96X0-7266	When SDES Phone or 2.6 Phone with capneg set to zero receives a capneg offer from CM, Phone should send answer with RTP.
	SIP96X0-7370	R2.6.SP4 SIP Unable to send DTMF on the active call using the auto-dial button.
	SIP96X0-7178	Hold un-hold fails if Phone has G.729A configured, CM has DM=No, Shuffling=No IP-codec set has G.729A, G.711A, G.711Mu, G.729AB in order.
	SIP96X0-7411	Two way speech path fails after Hold/Resume if Phone has G.729A configured, CM has DM=No, Shuffling=No IP-codec set has G.729A, G.711A, G.711Mu, G.729AB in order.
	SIP96X0-7347	Need user selectable tuning feature for Hearing Aid Compatibility (HAC).
	SIP96X0-7383	Cannot handle change in IP address and port in the ACK message

		and this leads to one way talk path.
	SIP96X0-7453	Side-tone and gain levels adjustments for Hearing Aid Compatibility (HAC) and 6-vent TIA tuning for WB codecs.
	SIP96X0-7449	Endpoint not able to negotiate codecs after hold/un-hold.
Failover\Failback	SIP96X0-7473	Phone does not do failback to primary SM in a scenario mentioned in description.
	SIP96X0-7485	Phone can’t dial after failover from SM1 to SM2.
WML	SIP96X0-7157	“br” tag added at the start of any tag is not working.
	SIP96X0-6092	WML browser sometimes runs out of memory for sockets.
PPM	SIP96X0-6963	Invalid value in PPM update Contact request when setting speed dial.
	SIP96X0-6697	PPM number formatting is not applied correctly.
	SIP96X0-7456	PPM download continues even if user logs out while PPM download is happening.
SBM24	SIP96X0-7364	Simplify messaging in button modules.
	SIP96X0-7397	Autodial feature doesn’t function when SBM is attached to the phone and at times when SBM is not attached.
	SIP96X0-7435	SIP2.6 phone with 3 BM works unstable on CM 6.2.
	SIP96X0-7488	Busy indicators are not displayed properly on BM12.
Call Log	SIP96X0-7355	Wrong outgoing name in Call Log for call pickup.
User Interface	SIP96X0-7384	SIP 2.6.7: Full phone picture is not shown on 102 version.
	SIP96X0-7387	SAC soft key is not updated immediately when sac is activated.
	SIP96X0-7398	Icon of Favorite contact should be changed according to its own Trace Presence setting.
	SIP96X0-7450	No way to return to Home screen if press More soft-keys on Track Presence line.
	SIP96X0-7489	BCAs fail to update on transferred calls.
The Avaya one-X® Communicator Shared Control	SIP96X0-6817	No Response to Shared Control Dialog INVITE after "RESET VALUES".
Signaling	SIP96X0-7455	User can remain logged in on two phones.
	SIP96X0-7494	Phone is sending 491 request pending response for UPDATE with SDP request causing the call to be dropped.
	SIP96X0-6970	Phone does not send SDP in 200 OK response to UPDATE with SDP.
	SIP96X0-7410	Unregister a valid extension because previous registered extension. On that phone had BSM as survivability and new extension does not have survivability.
Emergency	SIP96X0-7140	Implement support for PHNMOREEMERGNUMS parameter in settings file for multiple emergency numbers.

Known Limitations of Release 2.6 Service Pack 7

❖ 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
B5800 Branch Gateway	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
- Always provision 'None' encryption (setting 9) as one of the encryption selections in the settings file and on CM

❖ 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 Service Pack 7

Area	Issue	Workaround
Certificates	When the phone has installed 3rd party certificate and expired Avaya root certificate, it is not possible to login the phone.	Need to remove expired Avaya root certificate.
	EAP-TLS authentication with Microsoft SCEP server failed.	Do not use Microsoft SCEP server.
Failover\Failback	If the user makes changes in the options and settings when the phone has not yet detected that the proxy is unreachable and then the phone detects failover and registers with the secondary server and the user logs out and logs in as a different user, the changes are displayed to the new user.	This issue can be avoided by not logging in with a different extension after failover.
	When the phone is in Failover and not connected to a controller, the Emergency Soft key is displayed even though the phone is not able to connect calls.	Wait for 20 seconds.
	The phone requires up to 20 seconds to detect that the active controller has become unreachable. If the user goes off-hook during this interval, the phone will not play dial tone during this interval.	Wait for failover to occur.
Upgrade\Downgrade	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	A phone may retain a HTTPS file server setting set by DHCP even if the DHCP parameter is later set to “No” in CRAFT menu.	
	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.
	Change in Craft during phone boot-up will returns to Login screen.	Go to CRAFT menu and reboot the phone manually, after boot up the new parameters will be applied.
	Phone doesn’t restart after Phone IP address is changed followed by changes in SIP proxy settings, gets yellow triangle.	Wait extra 2 minutes for reboot or reboot it manually by unplugging\plugging power Ethernet cable.
	Invalid address advertised into LLDP packet by the IP Phone when multiple signaling servers addresses or File Server	Configure one File Server IP and one signaling server IP on the switch.

	addresses are configured on the switch.	
Audio	During shuffling 9620 does not transmits/receives RTP for approximate 0.6 seconds.	
UI and application	Long labels on the SBM button module may be truncated in the middle of a displayed character.	shorten button labels on the SBM.
	If the web screen saver (setting WMLIDLEURI) uses the same image file as the custom logo, the Web screen saver page is not displayed properly.	Do not use the customized logo image on the WMLIDLEURL page.
	Phone is displaying long form number in redial list even if user dials the short form number.	Dial with short form number if dialing with Long form is not allowed.
	When H323 calls PSTN then blind transfers to SIP phone. In the SIP phone history log shows caller's name H323 endpoint name and transferred-to (PSTN) number.	
	Phone displays two extension of call forward while only configure one extension on PPM.	Perform Logout and then Login.
Features	Assigning number to unlabeled autodial configured feature gives error beep and not allowed message on the top line and hence not able to assign the number when phone is registered with Session manager.	
	The MIB should support checking "NO_DIGITS_TIMEOUT" value and "INTER_DIGIT_TIMEOUT" value	
The Avaya one-X® Communicator Deskphone Mode	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.
Signaling	When IPv4 SIP phone gets a INVITE/re-INVITE with only IPv6 SDP then it responds 200OK.	Do not use IPv6, since phone does not support it.
Network	Phone and switch show different network settings when network parameters are not auto negotiated.	Use auto negotiation instead.

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 SP7 (SIP 2.6.7.0) 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 SP6. The following files are included in each package

- `hb96xxua3_00.bin` – The 9600 R2.6 SIP boot application
- `SIP96xx_2_6_7_0.bin` – The 9600 R2.6 SP7 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 SP7
  - **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 R2.6SP3 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 SP7:

- 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.

One provisioning example may be helpful. Assumptions:

- 96xx SIP telephone IP address: 10.10.10.20
- SIP telephone network mask: 255.255.255.0
- SIP telephone gateway: 10.10.10.1
- HTTP file server: 10.10.20.200
- SES IP address (primary controller): 10.10.20.192
  - TLS transport protocol, default so not explicitly specified in setting
- AudioCodes/Cisco IP address(secondary controller): 10.10.20.193
  - TCP transport protocol, explicitly set in setting file
- SNTP Server IP address: 10.10.30.147
- The SIP domain: mynetwork.com
- Time zone where IP telephone is provisioned: Eastern Daylight Savings Time (following the current rules where Daylight Savings Time begins on the first Sunday of April and ends on the last Sunday of October – for more information on provisioning DST see the 4600 Series IP Telephone LAN Administration Guide)

The following are the parameters that must be defined in your 46xxsettings.txt file to make the 96xx telephone operational:

SET SNTPSRVR	10.10.30.147
SET GMTOFFST	-5:00
SET DSTOFFSET	1
SET DSTSTART	1SunApr2L
SET DSTSTOP	LSunOct2L
SET SIP_CONTROLLER_LIST	10.10.20.192:5061;transport=tls,10.10.20.193:5060;transport=tcp
SET SIP_MODE	0
SET SIPDOMAIN	mynetwork.com

If the 96xx SIP telephone is not acquiring its address from a DHCP server, it must be provisioned on the phone, itself, through the use of the phone administration menu. Once in the “Admin Procedures” menu, the “ADDR” submenu will allow the administrator to provision the following fields:

Phone:	10.10.10.20
Router:	10.10.10.1
Mask:	255.255.255.0
HTTP File Server:	10.10.20.200

- Reset your Avaya 9600 IP Telephones.

License Agreements

This release includes 3rd-party commercial software:

HTTP Server functionality is provided by the GoAhead WebServer Copyright (c) 2009 GoAhead Software, Inc. All Rights Reserved.

This release Includes open source software. The full list, including licensing information, can be obtained by going at this link:

http://support.avaya.com/copyright

2012 Avaya Inc. All Rights Reserved.

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's web site at: http://support.avaya.com/ThirdPartyLicense/

Avaya fraud intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. Suspected security vulnerabilities with Avaya Products should be reported to Avaya by sending mailto: securityalerts@avaya.com.

For additional support telephone numbers, see the Avaya Web site:

http://www.avaya.com/support

Trademarks

Avaya, AUDIX, DEFINITY, and 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. Document ordering information:

For the most current versions of documentation, go to the Avaya support Web site: http://www.avaya.com/support

Avaya support

Avaya provides a telephone number for you to use to report problems or to ask questions about your contact center. The support telephone number is 1 800 242 2121 in the United States. For additional support telephone numbers, see the Avaya Web site: http://www.avaya.com/support

~~END OF FILE~~