



Avaya one-X® Deskphone SIP Release 6.2 Readme

This file is the Readme for the Avaya one-X® Deskphone SIP Release 6.2 software for the 9608, 9611G, 9621G, and 9641G IP deskphones. This file describes the contents of the August 2012 release software distribution package.

SIP 6.2 software is supported on the 9608, 9611G, 9621G and 9641G IP deskphones used with Avaya Aura® Communication Manager with Avaya Aura® Session Manager. The SIP 6.2 software will not load or operate on any other models. The SIP 6.2 software is not supported on the 9601 IP Deskphone.

This release supersedes all previous Avaya one-X® Deskphone SIP 6.0 software releases. Avaya recommends that all customers using Avaya one-X® Deskphone SIP 6.0 software upgrade to this version at their earliest convenience.

Avaya Aura® Communication Manager and Avaya Aura® Session Manager Compatibility

The 9608, 9611G, 9621G and 9641G IP Deskphones using one-X® Deskphone SIP Release 6.2 software are supported on Avaya Aura® Communication Manager 6.0.1 SP 6 patch 19350 or later, and Avaya Aura® Session Manager 6.1 SP5 or later.

Avaya Aura® Communication Manager 6.2 or later, and Avaya Aura® Session Manager 6.2 or later is the minimum requirements to support the following features:

- Group Paging
- Hotline
- Team Button

Avaya Aura® Call Center Elite 6.2 is the minimum requirements to support the following feature:

- Call Center Elite features

Avaya Aura® Call Center Elite Compatibility



SIP 6.2 provides enhanced interworking with Avaya Aura® Call Center Elite 6.2. **The use of SIP 6.2 with Avaya Aura® Call Center Elite 6.2 requires prior approval from Avaya.** Please contact sipagent@avaya.com to gain approval to use SIP 6.2 in conjunction with Avaya Aura® Call Center Elite 6.2.

Compatibility Matrix

The 9608, 9611G, 9621G and 9641G IP Deskphones using one-X® Deskphone SIP Release 6.2 software are supported with the following applications.

Service	Minimum Software
Avaya Aura® Conferencing Server 6.0	6.0 with bridge patch 6.0.1.6.1 or bridge patch 6.0.1.7.1
Avaya Aura® Conferencing Server 7.0	7.0
Avaya Aura® Presence Server	6.0, 6.1
Avaya Aura® Messaging	6.1, 6.2
Avaya Aura® one-X Communicator (shared control)	6.1.5
Avaya Secure Router 2330/4134 (for failover and survivability)	
Cisco 2811 ISR (for failover and survivability) (TLS is not supported)	c2800nm-ipbasek9-mz.124-20.YA2.bin
Audio Codes (for failover and survivability)	5.60A.021.004

SIP 6.2 Deskphones General Quality Improvement

Avaya has made significant improvements to our development and testing processes within SIP Release 6.2. Our resources have been engaged in testing and inspecting SIP 6.2 software to deliver a high quality, stable, reliable, and feature-rich solution that you can rely on for your critical business success.

Avaya employs a highly disciplined quality end to end approach; from improved software development practices, extensive code inspection, enhanced product verification and widespread live user trials. Stabilizing the foundation of the product has been the highest priority with special emphasis on quality as the number one goal of the release. This release of software reflects the most demanding and stringent practices and criteria ever used with an endpoints program in Avaya's history.

Avaya increased the focus on real-world solution testing by replicating customer network topologies and use case scenarios spanning the solution to thoroughly exercise how the phones are used by our customers. Extensive live system trials were part of the qualification process. The new enhancements to logging and diagnostics available in this release will enable Avaya to address any issues that may arise more quickly and effectively than with previous releases.

Avaya fully recognizes the responsibility and trust you place in us in achieving your business success.

In addition to adding the new features which are described in the next section, stabilizing the foundation of the product has continued to be our highest priority target also for release 6.2.

New Features in SIP 6.2

Avaya one-X® Deskphone SIP Release 6.2 is a major release than contains the following new features.

Feature	Description
Advanced Conferencing with Avaya Aura Conference 7.0	Supports displaying a roster of conference participants while providing the conference moderator with the ability to selectively drop or mute participants.
Avaya Audio Branding	The SIP Deskphone plays the Avaya audio brand when the user logs into the IP Deskphone or when the user unlocks the SIP Deskphone.
Call Center Elite features.	Provides enhanced interworking with Avaya Aura® Call Center Elite. The supported Call Center Elite features are: • Agent login/logout • After call work • Auxiliary work • Auto and Manual in • CC-Info agent event package • Third party MWI • Stroke counts • Call work codes • Display active VDN name • VuStats • Accept and display ASAI UUI information • Forced logout override • Supervisor assist • QStats • Interruptible AUX work
Call Pickup Alerting (CPA)	When a member of the pickup group is called, both visible and audible indications will be played on SIP Deskphones in the same pickup group.
DHCP Option 43	Provides support for selected configuration parameters via DHCP Option 43.
Dialing E.164 Numbers	Allow a user to dial an E.164 number, including the leading “+”
Enhanced Call Forwarding	Allows individual control of up to six different Call Forwarding types and destinations.
Group Paging	Supports SIP Deskphones as a member of a paging group, can make and/or receive pages
Hotline	Allows a user to place a call to a preconfigured destination by simply going off-hook or pressing a Call Appearance without the need to enter any dialing information.
Instant Messaging	Allows instant messages to be sent/received on the 9621G/9641G IP Deskphone when used with Avaya Aura® Presence Server.
Key Repeat Delay	Controls how long a physical key must be depressed before it begins to “auto-repeat”.
Large Text	The SIP Deskphone can display Latin based characters used in English in a bigger size font for better readability.

Feature	Description
Microsoft® Exchange 2007 contact integration	Allows the user to connect to a Microsoft® Exchange 2007 server via the user's Exchange ID and password to download up to 1000 contacts.
Microsoft® OCS Early Media	Supports interoperability with Microsoft® OCS.
Per Line Ring Control	Allows the user to assign one of the eight supported ringtones to any number on their contact list.
Phone Lock Idle Timer	Extends range of this feature
Presence with Avaya Aura® Presence Server	Allows the user to see the presence status of enterprise internal contacts stored in their contact list as well as to advertise their own presence status.
Privacy Feature	When Privacy header is present, displays the localized string "Restricted" for any Caller ID received in the From or Contact headers.
Enhanced Push API support	Allows trusted applications to send unsolicited content to the SIP Deskphone.
Shared control with Avaya one-X® Communicator.	Shared control with Avaya one-X® Communicator is supported.
SLA Mon Software Agent	The SLA Mon software agent allows interoperability with the Avaya Operations Intelligence Suite (OIS).
SSH support	Supports the Secure Shell (SSH) protocol to enable remote diagnostics of the SIP Deskphones by Avaya support resources.
Team Button	The Team Button feature allows a user to keep track of another member's presence, or enables the user to receive a call on behalf of the other (monitored) user. The team button also acts as a speed dial.
Ten Emergency numbers	Supports up to ten emergency numbers. The emergency numbers can be dialed even while the Deskphone is locked or unregistered
Visual Indication of Encrypted Call	Provides a visual indication to the user if a secure media connection (via SRTP) is being used by the SIP Deskphone. This indication does not imply that SRTP is used between the SIP Deskphone and the remote endpoint.
WML Browser	The SIP Deskphone supports a WML browser for Web applications. Please refer to the documentation for limitations.
WML Push	The WML Push feature enables the SIP Deskphone to display any Push messages.

For additional details on these features, the installation/maintenance/administration of these IP Deskphones, and User Guides, please refer to the following documents which can be accessed at <http://support.avaya.com>:

- Avaya one-X® Deskphone SIP 9608, 9611G, 9641G Administrator Guide Release 6.2 – (16-601944)
- Avaya one-X® Deskphone SIP Installation and Maintenance Guide Release 6.2 for 9608, 9611G, 9621G and 9641G IP Deskphones - (16-603504)
- Avaya one-X® Deskphone SIP for 9608/9611G IP Telephone User Guide – (16-603595)
- Avaya one-X® Deskphone SIP 9621G/9641G User Guide for 9600 Series IP Telephones – (16-603596)

- Avaya one-X® Deskphone SIP Call Center Agent 9608 and 9611G User Guide – (16-604108)
- Avaya one-X® Deskphone SIP Call Center Agent 9621G and 9641G User Guide – (16-603759)
- Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Application Programmer Interface (API) Guide – (16-600888)

SIP 6.2 Package Content

The SIP 6.2 package (96x1-IPT-SIP-R6_2_0-082012.zip) contains all the files necessary to upgrade Avaya new or previously installed Avaya 9608/9611G/9621G/9641G IP Deskphones to the SIP 6.2 software.

The following files are included in this package:

- S96x1_SALBR6_2_0r72_V4r70.tar – The SIP 6.2 application upgrade tar file
- S96x1_SALBR6_2_0r72_V4r70.orig.tar – The SIP 6.2 application tar file
- S96x1_UKR_V12r2083_V12r2083.tar - The SIP 6.2 platform system tar file
- 96x1Supgrade.txt – This file is downloaded by the IP Deskphones and instructs the deskphone on how to upgrade.

NOTE – This package does not contain platform/application files for the 9601 IP Deskphone which uses SIP 6.1 software. The included 96x1Supgrade.txt file references the SIP 6.1.4 platform/application files for the 9601 IP Deskphone (not included). Refer to the SIP 6.1.4 Readme for specific instructions on upgrading a 9601 IP Deskphone from previous versions of SIP 6.1 software and modify this 96x1Supgrade.txt file as required.

- Fifteen predefined language files for phone display:
 - o Mlf_Arabic.xml
 - o Mlf_BrazilianPortuguese.xml
 - o Mlf_CanadianFrench.xml
 - o Mlf_CastilianSpanish.xml
 - o Mlf_Chinese.xml
 - o Mlf_Dutch.xml
 - o Mlf_Enlarge.xml
 - o Mlf_English.xml
 - o Mlf_German.xml
 - o Mlf_Hebrew.xml
 - o Mlf_Italian.xml
 - o Mlf_Japanese.xml
 - o Mlf_Korean.xml
 - o Mlf_LatinAmericanSpanish.xml
 - o Mlf_ParisianFrench.xml
 - o Mlf_Russian.xml
- Eight extended Korean ring tone files:
 - o KoreanRT1.xml
 - o KoreanRT2.xml
 - o KoreanRT3.xml
 - o KoreanRT4.xml
 - o KoreanRT5.xml
 - o KoreanRT6.xml
 - o KoreanRT7.xml
 - o KoreanRT8.xml
- Three certificate files:
 - o av_csca_pem_2032.txt – Avaya Call Server CA certificate with an expiration date of 2032.
 - o av_prca_pem_2033.txt – Avaya Product Root CA certificate with an expiration date of 2033.
 - o av_sipca_pem_2027.txt – Avaya SIP Root CA certificate with an expiration date of 2027.

- 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.x Utility Server is the only file server that currently supports this.

System specific parameters, including new parameters introduced with this version of software, should be entered into the 46xxsettings.txt file which is available for separate download at <http://support.avaya.com>

Unresolved issues in SIP 6.2

The following table includes unresolved issues with this release of software:

External ID	Internal ID	Issue Description
	SIP96X1-4443	Key Repeat delay menu item should not visible to user when PROVIDE_KEY_REPEAT_DELAY value is set from 5 to 9 in settings file.
	SIP96X1-6347	Conference DAC (CCElite Direct Agent Calling) cannot drop DCP (Digital Phone) endpoint on Hold out of conference
	SIP96X1-6431	The team button is in ringing state on the BM12. If the user moves to the second page and comes back to the first page on the BM12, then the team icon disappears.
	SIP96X1-6448	Phone is not generating the DHCP-701 and DHCP-501 syslog event record.
	SIP96X1-6473	If you have both the team button and ECPA (Enhanced Call Pick-up Alerting) button configured simultaneously and one has abbreviated ringing, the other won't ring if first one is active.
	SIP96X1-6489	Adding valid user name with calendar integration does not immediately bring up the session. Five mins later you get a false error (bad exchange server address) and then the calendar button shows up
	SIP96X1-6584	AAC7.0-P3S3- Ignore button does NOT work on invited 96x1 clients instead the phone continues to ring
	SIP96X1-6613	In the case of displaying of a Call Forwarded call, if the user's name is too long to fit in the line, it is shortened and the forwarding party's name is not fully displayed after answer.
	SIP96X1-6647	While the phone is in conference, the user presses "ADD" and "DROP" softkey simultaneously and then press "Cancel" softkey. There is no resume softkey and there is no voice path.
	SIP96X1-6658	Intermittent "Invalid Password" displayed for exchange calendar is displayed on the phone.
	SIP96X1-6882	Session line of an established call from a BCA line appearance on SM1 disappears after fail over to SM2
	SIP96X1-6973	Intermittently the "blue arrow" icon: Outlook contacts missing with Exchange Contact integration
	SIP96X1-6980	Phone does not display number of H323/Digital/Analog contacts which are added from SMGR.
	SIP96X1-7032	Phone failed to go to Mute CRAFT when SET PROCPSWD to "0".

External ID	Internal ID	Issue Description
	SIP96X1-7065	Phone will not return to old firmware instead of coming up with default binary when the binary mentioned in upgrade file was not found or file name unknown during cross upgrade when cross grading from SIP to H.323.
	SIP96X1-7176	Changing the enable/disable status of Exchange Contact Integration only takes effect after a reboot or logout/login
	SIP96X1-7262	EAP-TLS works if 802.1x supplicant is enabled when the phone boots but is not correctly started if it is turned on after the phone boots.
	SIP96X1-7267	Feature invocation for Auto callback fails
	SIP96X1-7279	Phone displays wrong status line if invalid router address is configured.
	SIP96X1-7306	No Image is shown via WML push
	SIP96x1-7320	Core dump for AAC 7 participant (when added into a meetme conference) when hold call on participant side
	SIP96X1-7325	While Call Forward-Busy is activated, the call is not getting forwarded even if all the CAs are busy in 3 rd party environments.
	SIP96X1-7344	PPM returns searchUserResponse results, but the phone doesn't display them. The work around for this issue is to configure two SIP handles; one with E.164 and one with AVEXT
	SIP96X1-7376	After every clear phone displays "Limited phone service" with failover icon on top line for 1 minute.
	SIP96X1-7390	Using SIP TLS port 5062 impacts phone operation. The workaround is to use port 5061 for TLS.
	SIP96X1-7443	Garbled Unicode string is displayed for a principal call destination of a redirected call

Advisements with SIP 6.2 software

Restriction on Busy Indicators on Button Modules:

- 3 button modules can be connected
- The maximum combined number of busy indicators, team buttons and bridged appearances that can be configured on button modules is **48**.
- All button module lines can be configured as long as the restriction above is met.

Debug mode

As a general guide, it should be noted that response times could be impacted when debug or syslog is enabled

Instant messaging

Instant Messaging with Microsoft® Office Communicator is not supported

Server setting precedence

When upgrading a 9608/9611G/9621G/9641G SIP Deskphone from H.323 to SIP software please note that any setting that is set via the CRAFT menu (e.g. IP address) will take precedence over the information provided by DHCP when the phone is in SIP mode. For example if the HTTP server is set via that CRAFT menu in H.323 and then during the upgrade to SIP, if the phone requests new DHCP data, the DHCP data will not overwrite the data provided by the CRAFT menu.

Settings file

Please ensure the following is set in the 46xxsettings.txt file

```
SET CONTROLLER_SEARCH_INTERVAL 16
SET LOCALLY_ENFORCE_PRIVACY_HEADER 1
HEADSYS 0
```

SRTP

If SRTP is enabled, the following three parameters need to set in the 46xxSettings.txt file. Additionally, Communication Manager requires similar settings and also requires that the default value for ENFORCE_SIPS_URI be set.

```
SET ENFORCE_SIPS_URI 1
SET SDPCAPNEG 1
SET MEDIAENCRYPTION 1,9 or SET MEDIAENCRYPTION 1
```

EAP TLS

When EAP-TLS is enabled using the CRAFT menu, the phone should be rebooted to allow for proper EAP-TLS authentication.

TLS

When using TLS, for SIP signaling, only port 5061 can be used.

AAC 7 conferencing

Do not configure AAC7 conference focus URI so that ad hoc conference uses existing AST1 ad hoc conference.

Miscellaneous message

If you are manually programming IP addresses the error message "Subnet conflict" will appear if the router address entered is incorrect

Arabic language support

Arabic language is not supported on the 9608 model of phone.

SIP 6.2 Resolved Issues

The SIP 6.2 release provides continued improvement and optimization to key audio functions as well as stability, robustness and reliability.

The following table includes issues which are resolved with this release of software. This is not the complete list of all issues resolved between 6.0.4 and 6.2.

External ID	Internal ID	Issue Description
1-3442250372	SIP96x1-5854	Speaker shuts down when handset is placed on hook, but the next call when answered comes through the speaker again.
	SIP96x1-6320	Transfer and conference no speech path.
1-3392249291	SIP96x1-6103	96x1 phones display incorrect number while making an outgoing call.
1-3388664343	SIP96x1-4683	On mixed-user taker over, user remains logged in on two phones.
1-1MBZKUI	SIP96x1-6258	Button on phone display does not update after ACB activation.
1-3469178052	SIP96x1-6036	SIP hard phone issue with EC500.
	SIP96x1-6319	Fails to login during repeated login/logout.
1-3509567059	SIP96x1-6426	BCA phone is ringing forever.
1-2342387291	SIP96x1-4471	Phone does not send DTMF, have to reboot the phone to fix the issue.
1-3455757531	SIP96X1-6202	ACB Button on phone display does not update after ACB activation.
1-1N52AUR	SIP96X1-6435	Phone should not performs getDeviceData along with getAllEndpointConfiguration while "Reload Config" done in system manager.
1-1N12QQ3	SIP96X1-6438	Remove clearing of contacts cache in the phone in case of 500 server errors - users will be able to see their contacts
	SIP96X1-4503	Private Contact added via 1XC shows e-mail id instead of primary number on 96X1 device

License Agreements

License agreements are available at <https://support.avaya.com/Copyright>. Please select 96x1 SIP.

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