Avaya 1100 and 1200 Deskphone SIP Release 4.4 Readme

This file is the Readme document for the Avaya 1100 and 1200 Deskphone SIP Release 4.4 software for the 1220, 1230, 1120E, 1140E, 1165E IP Deskphones. This file describes the contents of the Oct 2013 (4.04.10) release software distribution package. Note that 1100 and 1200 Series SIP Software Release 4.4 is on Controlled Introduction for customers considering migrating their existing 1100 / 1200 Series SIP-capable IP Deskphones to Aura. Please follow the directions on the Avaya Support Download page to ensure a successful migration. SIP SW R4.4 is a unrestricted availability for customers using the SIP 4.4 as a maintenance release with CS1000, CS2100, or B5800.

SIP 4.4 software is supported on the 1220, 1230, 1120E, 1140E, 1165E IP Deskphones used with Avaya Aura® Communication Manager with Avaya Aura® Session Manager. The SIP 4.4 software will not load or operate on any other models.

This release supersedes previous Avaya 1100 and 1200 Deskphone SIP Releases 4.3/4.1/4.0 software releases.

The information in this document is accurate as of the issue date and subject to change.

Please refer to the Advisements in this file for important information prior to deploying this software.

SIP 4.4 Compatibility

Avaya recommends an upgrade to this SIP Software Release 4.4 for all applicable IP Deskphones and Call Servers at the earliest convenience. This SIP Software Release 4.4 is compatible with the following Call Server platforms:

- Avaya Aura® 6.2 Feature Pack 2 (1220, 1230, 1120E, 1140E, 1165E)
  - Avaya Aura® Communication Manager 6.3
  - Avaya Aura® Session Manager 6.3 / 6.3.2
  - Avaya Aura® System Manager 6.3.2
  - Avaya G430/G450 Media Gateway 6.3
  - Avaya Aura® Conferencing 7.2
  - Avaya Aura® Presence Services 6.2
  - Avaya Aura® Messaging 6.2
  - Avaya Session Border Controller for Enterprise 6.2
  - Avaya one-X® Communicator 6.1.6
- Avaya Branch Gateway B5800 Release 6.2 (1220, 1230, 1120E, 1140E IP Deskphones only)

- Avaya Communications Server 1000 (CS1000) R7.6 (1220, 1230, 1120E, 1140E, 1165E)
- Avaya Communications Server 2100 (CS2100) Release SE13, SE16, and SE17 (1220, 1230, 1120E, 1140E, 1165E IP Deskphones)

New Features in SIP 4.4

Avaya 1100 and 1200 Deskphone SIP Release 4.4 contains the following new features.

New with this release	Description
Existing 1100/1200 SIP4.3 Features behind Aura®	▸ Local Phone Features ▸ Aura® System Features with 1100/1200 SIP phones

Avaya Aura® Presence Services	▸ SIP4.4 Enhanced for Avaya Aura® connectivity:
	    ▸ Supports Rich Presence Extensions (RPID) per RFC 4480
	    ▸ Supports additional Avaya Aura® required extensions
	▸ Other phones can see 1100/1200 presence state. See Product Advisement on Presence
	▸ Requires using TLS for connection to S1/S2

Directory Services via Personal Profile Manager on Avaya Aura®	Enables some Avaya Aura® PPM support in 1100/1200 phone: ▸ Retrieving Contact List from PPM ▸ Adding/Deleting Contacts ▸ Updating Contact ▸ User Search ▸ Retrieving E911 numbers ▸ PPM reboot mechanism

One-X PC Client Control	▸ One-X Communicator softclient has shared control of 1100/1200 deskphone voice and call control
	▸ Supports the following actions:
	    ▸ Originate call
	    ▸ Answer incoming call
	    ▸ End a call
	    ▸ Ignore a call

TLS Support for Avaya Aura®	▸ SIP4.4 supports TLS and SRTP with Avaya Aura® server
SRTP Support for Avaya Aura®	▸ As above

New with this release	Description
Session Border Control	- SBC enables secure access for remote users - SIP4.4 tested behind the Avaya SBC for Enterprise 6.2 to confirm operation - Phone configured to use TLS - Features and calls successful

Multi User Login	- Enables some multi-user scenarios to be supported
	- One user can log onto up to 10 phones
	- Multiple users (extensions) can log onto one phone
	    - One phone can be logged into more than one system
	    - Not supported on 96x1 phones
	- Does not replace Shared Call Appearance / Bridged Line Appearance, since once the call is answered at one phone, other users cannot see status of that call on the other phones
	- The number of User logins to one extension is controlled by SMGR
	- Dependent on Avaya Aura® FP2 with parallel forking

Improved UI for Multiple Calls	- Brings a more predictable UI operation to the phone - Now Fixed key presses (e.g. line, hold, release, mute, headset, hookswitch) are applied to the active call - If no active call, action is applied to the highlighted call
Improved Non-consultative Transfer UI	- “Exit” softkey added to non-consultative transfer dialog - Supplements “Quit” hardkey - More obvious option for closing the dialog prior to transfer completion
Increased Address Book Default Size	- MAX_ADDR_BOOK_ENTRIES default increased from 100 to 1000 - Changes apply to Local address book (stored on phone) - Local address book = locally entered + network downloaded entries - New parameter MAX_DOWNLOAD_ADDR_BOOK_ENTRIES - Max number of entries downloaded from network - Range 0 – 1000; default = 1000 - Limited to MAX_ADDR_BOOK_ENTRIES - Local entries added at phone until MAX_ADDR_BOOK_ENTRIES limit is reached

New with this release	Description
Improved prtcfg Command	• Enables faster debugging as config of a customer's phone can be easily converted into a config file for duplicating problems • Output can be directly copied to another phone's cfg file • Displayed names match dev config file parameter names • Output now matches device config file format • Previously missing parameters now listed • Detailed server profile information shown • Passwords replaced by *** • Domain related info grouped and separated by comments
Server Profile Auto Login Parameters	• User can now have different auto-login per server • Auto login parameters processed from Server Profiles

Improved Directory Search	• Search of Directory is now case insensitive
	    • Displayed entries based on user input characters
	    • Applies to Name and 1st Character search methods
	• Name and 1st Character search now on 1200 phones
	    • Added to existing index search
	    • Provides same search options on both 1100 and 1200

Improved Security	• Debug port requires password entry to enable
	    • Reduces unauthorized access to debug port and thus network
	• Port mirroring can be disabled via provisioning
	    • Prevents enabling even when the menu password is known
	    • Reduces unauthorized access to network

BootC HTTPS Support	• Enables secure FW download in BootC • BootC used to download app when available memory exceeded • Download can now use HTTPS in such cases
IPv6 Enhancements	• IPv4/IPv6 Redirect Scenario Support • Improved DHCPv4/DHCPv6 Server Unreachable • Support for DHCPv6 DECLINE message • Applicable to non-Aura systems as Aura currently doesn't support IPv6 endpoints

Removed Features in SIP 4.4

Avaya 1100 and 1200 Deskphone SIP Release 4.4 removes the following functionality.

Removed with this release
NONE

SIP 4.4 Package Content

The SIP Software Release 4.4 package contains all the files necessary to upgrade Avaya new or previously installed Avaya 1120E, 1140E, 1165E, 1220 and 1230 IP Deskphones to the SIP Software Release 4.4.

Language files are available on the Avaya Support portal, and include some translation additions and improvements. Language files are used for localization of IP Deskphones, and need to be updated separate from the SIP Software upgrade. The Language files are downloaded to the IP Deskphone via the Provisioning file. Up to five language files can be downloaded to the IP Deskphones. More details can be found in NN43170-600 SIP Software for Avaya 1100 Series IP Deskphones – Administration Guide, and the NN43170-601 SIP Software for Avaya 1200 Series IP Deskphones. After a language file is downloaded to the IP Deskphone, the end-user can change the language that is displayed on the IP Deskphone by selecting the corresponding language in the Preferences menu.

- Czech
- Danish
- Dutch
- Finnish
- French
- German
- Greek
- Hungarian
- Italian
- Japanese
- Latvian
- Norwegian
- Polish
- Portuguese
- Russian
- Slovenian
- Spanish
- Swedish
- Turkish

System specific parameters should be entered into the DeviceConfig file which is available for separate download at FTP site. New configuration parameters introduced with this release of software are shown in Appendix 1.

Advisements with SIP 4.4 software

SIP 4.4 Resolved Issues

The SIP Software Release 4.4 continues to improve the overall quality of the IP Deskphones software through delivery of ongoing resolution of Avaya identified work items and customer reported issues. The following table includes the list of externally reported field issues resolved with this release of software compared to SIP 4.3 Service Pack 1.

External ID	Internal ID	Issue Description
SR 1-4262358532	IPCLIENTS-11434	SR 1-4262358532 - SIP Phones are not recovered after a network has been unavailable during 17 hours
SR 1-4314297908	IPCLIENTS-11423	SR 1-4314297908 - 1140E SIP phone rejects an incoming call
SR 1-4262358532	IPCLIENTS-11387	SR 1-4262358532 - Autologin doesn't work after the network was restore
SR 1-4239662318	IPCLIENTS-11332	SR 1-4239662318 - 11x0 SIP Phone locks up and becomes non responsive to key strokes
SR 1-4234598146	IPCLIENTS-11299	SR 1-4234598146 - 1140E SIP phones display garbled characters in caller id for incoming calls
PEA 1-1UE9CLQ	IPCLIENTS-11157	PEA 1-1UE9CLQ Need assistance reading SIP debug logs from a 1140E sip set.

Product Advisements for unresolved issues in SIP 4.4

New Issues in SIP 4.4 (not found in earlier releases)

Contact Center features are not supported on 1100 / 1200 Series SIP-capable IP Deskphones with Avaya Aura®

The 1100 and 1200 Series IP Deskphones with SIP Software R4.4 are not supported as Contact Center devices with Avaya. Customers who need Contact Center features on their Deskphones should use 96x1 IP Deskphones.

Bridge call appearance is not supported with Avaya Aura®

Mutli-user login does offer some mult-user features.

Presence does not support the parallel forking feature in Avaya Aura®.

If end user has logged into more than one phone, the presence state reason information is not updated correctly.
IPCLIENTS-11535

Idle 1100/1200 IP Deskphones appear as "offline" in 96x1 IP Deskphones' presence status.

For customers who have a mix of 1100/1200 IP Deskphones with Avaya 96x1 IP Deskphones, limitations exist for how the 1100/1200 presence status shows up on 96x1 IP Deskphones when the 1100/1200 IP Deskphones are in idle state. Busy, On the Phone and Away activities are correctly displayed.

Making a call as the phone does failback from S2 to S1 may cause a phone reboot.

An intermittent occurance of the phone rebooting if a call is started just when S1 comes up and the phone does failback has been seen.
IPCLIENTS-11615

Existing Issues in SIP 4.3 or previous releases, found in SIP 4.4 testing

Product Advisements for SIP Software Release 4.3 SP1 and earlier

The following is a list of Product Advisements or notes associated with SIP Software Release 4.3 Service Pack 1 or earlier. Some advisements remain from previous releases of software, whereas other advisements reflect new or changed behavior found or introduced with SIP Software Release 4.4.

Feature interaction issue between local phone 3-way conferencing and putting users on hold.

Putting user on hold while in the local phone 3-way conference may cause Music on hold to play.

If using SRTP with SRTP_CIPHER_2 AES_CM_128_HMAC_SHA1_80 the originator can not hold a call.

Workaround: either do not use SRTP or configure the SRTP to use SRTP_CIPHER_2 AES_CM_128_HMAC_SHA1_32. Tracking with CM ticket [defsw131455]

Limitations to User Data memory usage

User data such as custom ring tones, images, etc. must not exceed a total of 500kB. There may be not enough space in the file system for other data (e.g. phone logs, language files, user preferences, address books, Inbox/Outbox, etc.)

Potential for multiple reboots when upgrading earlier vintages of 1120E and 1140E IP Deskphones

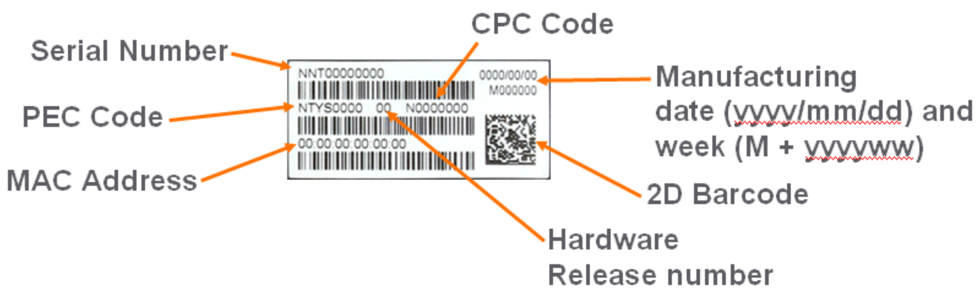
Some earlier vintages of 1120E and 1140E IP Deskphones may go through a number of reboots before the SIP Software Release 4.4 upgrade is installed, due to a different memory configuration on these earlier models. See the chart below for vintages of 1120E and 1140E IP Deskphones that may exhibit this behavior. Not that this has changed from a previous communication where the earlier vintages of the 1120E and 1140E IP Deskphones noted below were said to be not supported at all.

IP Phone 1120E
NTYS03AC N0132697 IP Phone 1120E Graphite w/ Icon Keys w/o PS
NTYS03ACE6 N0132699 IP Phone 1120E Graphite w/ Icon Keys w/o PS (RoHS)
NTYS03BC N0132698 IP Phone 1120E Graphite w/ Eng Keys w/o PS
NTYS03BCE6 N0132700 IP Phone 1120E Graphite w/ Eng Keys w/o PS (RoHS)
NTYS03BCGSE6 N0132701 IP Phone 1120E Graphite w/ Eng Keys w/o PS (GSA)
NTYS03CAE6 N0142351 IP Phone 1120E (SIP) Graphite w/ Icon Keys w/o PS (RoHS)
NTYS03DAE6 N0142352 IP Phone 1120E (SIP) Graphite w/ Eng Keys w/o PS (RoHS)

IP Phone 1140E
NTYS05AC N0132702 IP Phone 1140E Graphite w/ Icon Keys w/o PS
NTYS05ACE6 N0132704 IP Phone 1140E Graphite w/ Icon Keys w/o PS (RoHS)
NTYS05BC N0132703 IP Phone 1140E Graphite w/ Eng Keys w/o PS
NTYS05BCE6 N0132705 IP Phone 1140E Graphite w/ Eng Keys w/o PS (RoHS)
NTYS05BCGSE6 N0132706 IP Phone 1140E Graphite w/ Eng Keys w/o PS (GSA)
NTYS05CAE6 N0142353 IP Phone 1140E (SIP) Graphite w/ Icon Keys w/o PS (RoHS)
NTYS05DAE6 N0142354 IP Phone 1140E (SIP) Graphite w/ Eng Keys w/o PS (RoHS)

1120E and 1140E IP Deskphones with PEC codes later than those in the table above will not show this behavior. That is, 1120E with PEC Codes with 'D' in them (NTYS03ADE6, NTYS03BDE6) or higher (where this character in the PEC Code is 'D' 'E', 'F' or beyond) will not show this behavior. Similarly, 1140E PEC Codes NTYS05AEE6, NTYS05BEE6, NTYS05BEGS or above will not show this issue.

The Figure below provides an explanation of where to identify the PEC and Hardware



Release Number on the white product label located on the back of the IP Deskphone.

The following Product Advisements are related to known product behavior related to network conditions, hardware and accessories.

IP Deskphone's performance will be diminished during broadcast storms (applies to all the IP Deskphones)

By default, network traffic to the IP Deskphone will be accepted based on the packet's destination MAC address. The phone will therefore accept, in addition to all unicast packets sent to the phones MAC address, all broadcast and multicast packets as well. If the network environment results in a high amount of broadcast or multicast traffic, the IP Deskphone's performance may be impacted.

If "Voice 802.1Q" is enabled on the phone, the phone can then be provisioned to filter some or all of the broadcast or multicast traffic. If "VLAN Filter" is enabled, packets will be accepted by the phone based on the packet's destination MAC address as well as the packet's VLAN tag. Untagged packets and packets with a VLAN tag different from the Voice VLAN ID will be prevented from reaching the phone. This will protect the voice application from excessive traffic sent to the broadcast address or to the multicast addresses. But please be aware, <u>if VLAN filtering is enabled on the phone, one must ensure that voice packets are tagged with the appropriate VLAN ID as they exit the network switch, or else the packets will be dropped by the filter</u>.

Throughput may be slow for large file transfers on conversions from GigE to 100Mbit (applies to the 1120E, 1140E, and 1165E IP Deskphones)

In networks in which a PC is connected to the IP Deskphone's PC port and the PC's NIC speed is 100Mbit but the network speed is at GigE, large file transfers to the PC can take quite a long time. This is an issue with large file transfers only. Due to the speed mismatch between the phone's two ports the buffers in the phone can overflow resulting in retransmissions.

Although the IP Deskphones support Ethernet flow control (802.3x), the support is only implemented on the phone's PC port, not on the phone's network port. Ethernet flow control is a mechanism were the IP Deskphone can request a brief "pause" from the transmitting Ethernet device if the IP Deskphone buffers are about to overflow.

Ethernet flow control cannot be implemented on the phone's network port, since it impacts the phone's voice quality. As a result, in environments were the network is GigE but the PC NIC is only 100Mbit, large file transfers from the network to the PC can take quite a long time. On the other hand, since Ethernet flow control is implemented on the phone's PC port, in environments were the PC NIC is GigE but the network is only 100Mbits, large file transfers should be well managed by the phone's Ethernet flow control mechanism.

Some models of Plantronics Bluetooth headset may unexpectedly become unpaired (applies to the 1140E and 1165E IP Deskphones)

An issue was uncovered with certain Plantronics Bluetooth headsets (including the formerly validated Plantronics Voyager 510/510S) in which the headset may unexpectedly become unpaired. If the unpair occurs during an active call, all audio will be lost to and from the headset. In such a situation the call will remain active and the user is recommended to switch to handset or handsfree. Due to the severity of this issue, Avaya does not recommend the use of the Plantronics Voyager 510/510S headset. The 1165E IP Deskphone is not supported with IP Office.

Backlight Interaction with USB devices (applies to the 1120E and 1140E IP Deskphones)

Some USB devices (i.e. Mice or Keyboards) send regular coordinate update messages to the phone even when the device is not being used. This can cause the sleep mode for the backlight to not be properly invoked.

Power disruption during software upgrade will corrupt the upgrade (applies to all the IP Deskphones)

During a software upgrade, if a power disruption is experienced by the phone, the software upgrade will fail. In some instances a power disruption during an upgrade may also corrupt the existing software on the phone. If this corruption should occur, the phone will fail over into its boot code known as “BootC”. BootC will automatically try to restore the phone’s software from the image on a call server. But for the 1100 Series and the 1200 Series IP Deskphones, if the phone’s software was obtained from a TFTP server instead, in order to restore or upgrade the software from BootC, a manual TFTP download from BootC must be performed. The Manual TFTP Download from BootC Procedure is documented in the IP Phones Fundamentals NN43001-368. **NOTE: Caution should be exercised to avoid power disruptions during software upgrades.**

Appendix 1 – New/Changed DeviceConfig parameters

The following two parameters are new with 1100/1200 IP Deskphones SIP Software Release 4.4, and are related to these phones when registered with Avaya Aura® .

DeviceConfig.dat:

```
# AVAYA_AURA_MODE_ENABLE [YES | NO]
#  This parameter is a command that specifies if Avaya Aura® specific features
#  are active on the IP Deskphone or not.
#   - YES – Avaya Aura-specific features are active.
#   - NO – Avaya Aura-specific features are not active.
AVAYA_AURA_MODE_ENABLE YES

# USE_DEFAULT_DEV_CERT [YES | NO]
#  This parameter controls the use of the default device certificate for
#  HTTPS/TLS connections.
#    - YES – The default device certificate is used for HTTPS/TLS connections
#          when a customer device certificate is not installed.
#     - NO – The default device certificate is not used.
USE_DEFAULT_DEV_CERT YES
```

The following four parameters existed previously. Note the following settings shown are required when the phones are used with Aura.

DeviceConfig.dat:

```
# ENABLE_SERVICE_PACKAGE [YES | NO | PPM ]
#  This parameter toggles the subscription to the call server service package.
#  When the IP Deskphone connects to a call server that does not recognize the
#  service package, the subscription for the service package fails. If this
#  happens, ad hoc conferencing is not available, even if the call server
#  supports ad hoc conferencing. You can configure values for ad hoc
#  conferencing when the service package is not retrieved. The IP Deskphone
#  retrieves the service package based on a configurable Boolean value.
#    - YES – the IP Deskphone downloads the service package. Used with
#          AS5300 and CS2k call serveres.
#    - PPM - it should be set to PPM if Personal Profile Manager with
#          Avaya Aura SM/CM is used
#     - NO – the IP Deskphone does not download the service package.
#          Used with most other call servers.
ENABLE_SERVICE_PACKAGE PPM
```

```
# ADDR_BOOK_MODE [NETWORK |LOCAL | BOTH]
#  This parameter selects the address book that is used to search for other
#  users. The default setting is NETWORK.
#    - NETWORK – downloads the user's address book from the network.
#              New address book entries are uploaded to the network.
#    - LOCAL –   creates a user address book and stores it locally on the IP
#              Deskphone. ( used with Aura in support of the Gobal Search )
#    - BOTH –    attempts to download a network address book and keep a copy
#              on the IP Deskphone. If a network address book is available,
#              the IP Deskphone functions as if NETWORK mode has been
#              selected.
```

ADDR_BOOK_MODE LOCAL

```
# DISABLE_PRIVACY_UI [YES | NO]
#  This parameter disables the privacy setting in UI menus. Disabling the
#  privacy setting in UI menus disables the user's ability to configure
#  privacy options (incoming and outgoing Caller ID).
#    - YES – disables the privacy setting in the UI menus. (used with Aura)
#    -  NO – enables the privacy setting in the UI menus. NO is the default.
```

DISABLE_PRIVACY_UI YES

```
# MKI_ENABLE [YES | NO]
#  This parameter indicates whether to use the Master Key Identifier (MKI)
#  or not
#    - YES – MKI is configured
#    - NO – MIK is not configured (default )
```

MKI_ENABLE NO

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