Avaya E129 SIP Deskphone Release 1.25.1.1 Readme

This file is the Readme for the Avaya E129 SIP Deskphone Release 1.25.1.1 software. This file describes the contents of the April 2014 release software distribution package.

This software release for the Avaya E129 SIP Deskphone is used with Avaya Aura® Communication Manager with Avaya Aura® Session Manager. This software will not load or operate on any other phone models.

Support for the Avaya E129 SIP Deskphone with Avaya IP Office is planned to coincide with the release of IP Office 9.0.3. A further update to this readme document identifying support is planned to be published at that time of IP Office 9.0.3 GA.

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.

Avaya Aura® Compatibility

The Avaya E129 SIP Deskphone using 1.25.1.1 software is supported on:

- Avaya Aura® Communication Manager 6.3.2 or later with Avaya Aura® Session Manager 6.3.2 or later and with Avaya Aura® System Manager 6.3.2 or later
- Avaya Aura® Conferencing 7.2 or later (supports MeetMe conference only)
- Avaya Aura® Messaging 6.2 or later
- Avaya Session Border Controller for Enterprise 6.2 is the minimum version to support remote Avaya E129 SIP Deskphone.
- Avaya Aura® Solution for Midsize Enterprise 6.2 or later

The above releases are the minimum requirements to support the Avaya E129 SIP Deskphone

Features in release 1.25.1.1

The 1.25.1.1 Release launches the Avaya E129 SIP Deskphone to the market.

The Avaya E129 SIP Deskphone is ideal for replacing the aging analog and digital phones, with an attractively priced basic telephony solution. The Avaya E129 SIP Deskphone offers entry-level communications capabilities that ideally suit the needs of basic voice communications

The Avaya E129 SIP Deskphone is different from other Avaya SIP phones because it supports only basic, generic SIP implementation.

The essential features supported are the ones that most analog replacements need, such as Transfer, Forward, Mute, Ad-Hoc Conference (except with G.729) and Hold.

The E129 does not support the rich set of SIP enabled telephony features that are a part of Avaya Aura® Advanced SIP Telephony (AST). Customers that desire a phone similar to the E129, that supports Advanced SIP Telephony should consider the Avaya 9601 SIP Deskphone or the other more advanced phones in the 9600 Series.

Key characteristics and features of the phone are:

- Single-line IP phone, with support for two concurrent calls with a Flash key.
- Common features: Transfer, Forward, Mute, 3-Way-Ad-Hoc-Conference and Hold.
- Monochrome 128x40 pixel display (2 ¾" x 1") with three rows that provides calling party information, soft key labels, and status indicators.
- Three context sensitive soft keys.
- Fixed hard buttons for Mute, Transfer, Conference, Flash, Voice-Mail, Speaker, Volume, and Directory.
- Directory with up to 500 contacts, and Call History with up to 200 entries.
- G.722 wideband audio, handset and headset (with wideband headsets).
- Full duplex speakerphone.
- Wall-Mount and Desk-Mount stand.
- Headset port (RJ9 connector).
- Message Waiting Indicator.
- Five Button Navigation Cluster, for easy manipulation.
- Dual Ethernet port operating at 10/100 Mbps speed.
- Class 1 Power over Ethernet.
- Support for optional (separately orderable) international AC power adapters.

The phone supports the following languages*:

- English
- French
- German
- Italian
- Polish
- Brazilian-Portuguese

- Spanish
- Russian
- Simplified Chinese

*The administrator documentation, and administrator web user interface is available in English only.

For a compete overview of feature functionality please refer to the User and Administrator documentation.

Features not supported in release 1.25.1.1

- G726 codec
- 802.1x using EAP-TLS

E129 release 1.25.1.1Package Content

The Avaya E129 SIP Deskphone release 1.25.1.1 package contains all the files necessary to upgrade Avaya E129 SIP Deskphone to the SIP 1.25.1.1 software.

- E129fw.bin – The E129 SIP 1.25.1.1 application, platform and DSP bin file.
- Avaya E129 SIP Deskphone Release 1.25.1.1 Readme, which is this document.
- Cfg.xml - The E129 config file for system specific parameter.

Advisements with E129 1.25.1.1 software

Upgrading the E129 to the latest release.

New out-of-the-box phones must be upgraded to use this release as it includes the most updated firmware with fixes for known issues in previous releases.
Please refer to "*Installing and Maintaining Avaya E129 SIP Deskphone*" for upgrade instructions.

If your phone has a pre 1.0.5.45 software release installed (To identify the release level go to E129 menu, scroll down to status menu and look for Prog version) perform the following steps after upgrading:

- Restore phone to factory default (connect via SSH remotely and use the reset command or use the phone menu, scroll down to config menu and in this menu scroll down to factory reset option).
- If you opened the WebUI on load prior to 1.0.5.45, clear the browser caching before reconnecting with the latest load otherwise it might result in display errors (wrong values and menus).

Important note: During upgrade procedure the phone must have an accessible HTTP server containing the new firmware. Failing to do so and interrupting the upgrade procedure without an accessible HTTP server requires an additional power reboot to recover.

It is recommended to configure the E129 phones for automatic upgrade that periodically checks for the new firmware on the HTTP server. This will save the need for manual reboot to trigger an upgrade.

Using the E129 config file

The config file includes configurations to enhance phone behaviors to Avaya Aura environment. It is recommended to use this file even if the administrator did not make any changes.

E129 supports only one audio stream with G729 codec

E129 supports only one audio stream using codec G729. Customers that are using solely G.729 codec need to note that they will not be able to make local conference call managed by the E129 phone and will need to use Avaya Aura conference server for making such calls.

SRTP configuration

In order to use secured RTP (SRTP) the Aura Communication Manager or the Avaya SBC (ASBCE) must be configured not to enforce SIPS.

Working in an environment where both RTP and SRTP are used, E129 should be configured for “SRTP Mode” “Enabled But Not Forced”.

Working in an environment strict to SRTP only the crypto suite on Communication Manager should not contain the “None” option and E129 should be configured for “SRTP Mode” “Enabled And Forced”.

The E129 phone does not support the option of AES_CM_128_HMAC_SHA1_32.

Using multiple devices with the same extension

In order to use the E129 as part of multiple devices serving the same extension this extension should be configured on the Aura System Manager as multiple registration group.

Using E129 keypad lock feature

The keypad lock feature can be used to secure the phone while the user is away. It is important to note that it is possible to unlock the phone only when it is in idle state.

Presence is not supported

The E129 doesn’t support the presence features and does not publish its presence status. Do not use this option in the WebUI menus.

Session Manager Firewall – potential for service outage (Applies only to SM 6.3.0 / 6.3.1)

Starting with Avaya Aura® Session Manager 6.3, the SIP firewall function will be enabled by default. The SIP firewall will be turned on for new installations and for upgrades from systems where the SIP firewall had not been previously configured. Due to the default values chosen for firewall rules, SIP endpoint users that engage in more than 10 call related feature invocations within a 5 minute interval will encounter a brief service outage. Evidence of this can be seen in the System Manager Alarm logs. Call origination, hold/un-hold, transfer, conference and park/un-park are some examples of call related feature invocations that count toward this limit. If you have users that fit this traffic profile, it is recommended that the "ASM Default Configuration" be modified. Refer to the Session Manager Release Notes for additional information.

SIP Transport Protocols

TCP or TLS are the recommended transport protocols. UDP transport is not supported with E129.

Using TLS transport with E129 default certificate is only for demo purposes and not recommended to be used in production environment. Before using the phone in production environment customer must install a new dedicated certificate. When providing the private key and associated certificate using the config file please make sure that it does not include line feed (It should have space instead).

Note: If the E129 phone was preloaded from manufacturing with a load prior to 1.25.1.1 please refer to RAPID-767 in the known issues section

E129 WebUI support

The WebUI is designed for the administrator usage only. The main configuration method is the config file and the WebUI should be used when there is a need to change specific phone configuration (like in the case of debugging).

Please note the following information while using the WebUI:

- E129 WebUI interface is supported only for English.
- The date format is always MM/DD/YYYY.
- Configuration parameters that are part of the WebUI, but not part of the config file are not officially supported. For example, SIP Advanced Feature tab is not supported.
- WebUI is supported on Internet Explorer 8, 9, 10 and 11, on Firefox 27.0.1 and Google Chrome 32.0.1700.107 m.

Using the E129 with the Avaya SBC Server

When registering an E129 behind an Avaya Session Border Controller the following configuration should be applied to the SBC:

1. Create topology hiding profile. Provide the domain name for different SIP headers.
2. Apply that topology hiding profile to "Endpoint flows" and "server flows"

When using the TLS connection, the E129 phone needs to be configured to use SIP and not SIPS and Avaya SBC needs to be configured not to enforce SIPS. In order to configure it on the E129 phone, modify the config file parameter of "SIP URI Scheme when using TLS" (P2329) to SIP (value is 0).

Using CM features with E129

E129 supports call forwarding features locally as described in the user guide. Customers can also use the Avaya Aura Communication Manager call features utilizing the Feature Access

Codes (FAC) numbers. It is important to note that E129 will not provide indication on the FAC status.

Resolved issues in E129 1.25.1.1

The following table includes resolved issues delivered in this release of software.

External ID	Internal ID	Issue Description and possible workaround if any
		Reboot/Lockup
	Rapid-696	In rare cases after sitting idle for long time and using multiple device on the same extension as E129 the phone might lockup. PC port connectivity will work fine. This issue is now resolved
		User Interface
	Rapid-208	User starts a call in handset mode and move to speaker mode w/o on hooking the handset. If in that state the user will start transferring a call and before completing the transfer action he will on hook the transfer action will be aborted (returned to the original call). This issue is now resolved
	Rapid-114	In specific timing while user is establishing call on first line (ringing phase) and receives incoming call on the second line the visual MWI LED alerting will continue blinking after the call is ended. This issue is now resolved
	Rapid-357	For non-English language boot up strings were corrupted. It is now fixed and will be displayed in English.
	Rapid-117	Phone would play low volume ringtone when volume level is set to zero This issue is now fixed and there would be no audio alert with volume level set to zero
		Contacts
	Rapid-676	E129 will fail to dial LDAP results that contain spaces in the phone number This issue is now resolved.
		Administration
	Rapid-695	E129 doesn't send RTCP report. This issue is now resolved
		Audio

External ID	Internal ID	Issue Description and possible workaround if any
	Rapid-747	Sidetone is too low when using E129 handset. Users speaking with the handset might think that their voice is not transmitted. This issue is now resolved
	Rapid-718	Using Hands free mode in mid to high volume might generate echo on the far end. This issue is now resolved

Unresolved issues in E129 1.25.1.1

The following table includes unresolved issues with this release of software which were known as of the issue date of this document.

External ID	Internal ID	Issue Description and possible workaround if any
Failover		
	Rapid-102	If a failover occurs while a call is in HOLD state, the call cannot be resumed nor dropped. The line will stay in use (so other incoming/outgoing calls will be using the 2nd line). Workaround: Either reset the phone, or wait for the failed SM to recover, or use the 2nd line (as mentioned above).
	Rapid-104	In an incoming call scenario, while ringing, if the connection to the active SM is lost and at the same time the user is trying to answer the call, the phone might get to a state where the call is not established while line-1 will stay busy (so additional calls will be coming on line-2). Workaround: Rebooting the phone will resolve the situation altogether. Another option is to wait until the failed SM recovers, or use the 2nd line (as mentioned above).
	Rapid-110	In case a call is put in a HOLD state shortly after a failover occurred (less than 40-60 seconds after), the ability to resume the call is lost until the failed SM recovers. Workaround: Wait for the SM to recover, or reset the phone.

External ID	Internal ID	Issue Description and possible workaround if any
	Rapid-111	When the primary SM is put in "Deny Service" mode (for maintenance purposes, etc.), E129 phones will not detect that as a reason to initiate fail over to the secondary SM, and hence outgoing calls made by the E129 phones will fail. Incoming calls will be correctly treated (by the secondary SM), and since E129 phones do register also to the secondary SM, these calls will be properly routed to them as well, so incoming calls will continue to work for E129 phones. *Workaround: block port 5060 (for TCP) or 5061 (for TLS) on the SM in "Deny Service" mode. After 40-60 seconds, all E129 phones should failover to the secondary SM.*
		User Interface
	Rapid-772	Changing the phone language to an already installed phone will result in loss of handset functionality. Speaker and headset will continue working fine. Workaround: logout the user and re-login to recover. An alternative workaround is to reset the device after the change.
	Rapid-744	Secured RTP icon on phone display will be set to off after holding and resuming a SRTP call. This is even though the resumed call is using SRTP.
	Rapid-743	User starts a call in headset mode and move to speaker mode w/o on hooking the handset. If in that state the user will start transferring a call and before completing the transfer action he will on hook the transfer action will be aborted (returned to the original call). Workaround: resume the original held call and start the transfer again w/o changing the audio source
		Administration
	Rapid-767	E129 that was loaded with an earlier release prior to upgrading to load 1.25.1.1 will fail to receive calls when using TLS in demo mode. Workaround: Download your own generated private key and certificate to the device as recommended. If you wish to work with the default private key and certificate either restore the device to factory reset or connect via SSH, enter config mode and use the "*unset 280*" command.
	Rapid-705	Downloading multiple custom ringtone will download only the first ringtone (ring1.bin) Workaround: Download only one ringtone at a time. Replace the ring tone on the server with the next one after downloading the previous one.

External ID	Internal ID	Issue Description and possible workaround if any
	Rapid-701	SSH status in webUI is displayed incorrectly when changing the default value in the config file. Changes to enable/disable SSH in webUI will take affect after reset. In the case that the config file is changed from the default value the webUI will display this value after the reset even if the user configured SSH on the webUI differently Workaround: Use the config file default value of SSH disabled and changes on the webUI will work fine. In case config file is changed, connect to device using SSH to understand the correct status
	Rapid-689	Device that is configured for ip address will boot up and prompt to login providing no access to phone menu. Workaround: in case the ip address is valid and administrator would like to make changes to specific phone he should access the webUI. In case that that the ip address is not valid the admin will have to connect it to a network where this address is reachable and use the webUI. When in login screen press the conference hard key to get the phone ip address
	Rapid-680	User that accessed the webUI for device with release prior to 1.25.1.52 and upgraded to the latest release will have display issues with the webUI. Such issues are wrong values of parameters or corrupted text. Workaround: Clear the browser cache before accessing the webUI after the upgrade.
	Rapid-106	E129 will unregister and immediately register again when additional phone will register with the same extension in a non-multiple registration group configuration. The new phone that will try to register will unregister or will not get calls. Workaround: Configure the user extension on Aura System Manager for multiple registration group.
	Rapid-61	E129 will not handle a 301 message. Workaround: configure the E129 SIP server as the ip address of the Session Manager that it was configured on
Call features		
	Rapid-727	Attendant transfer fails in case that Communication Server is configured for SIP Endpoint Managed Transfer (SEMT). Workaround – Change Communication Manager configuration to disable SEMT

External ID	Internal ID	Issue Description and possible workaround if any
	Rapid-712	In hot desking scenario where multiple users use the same E129 phone, the call forward features enabled by first user will be applied to the next user if not turned off before logging out by the first user. Workaround: User should disable call forward feature before logging out in a hot desking environment. New user that is logging in and call forward icon is on should disable the call forwarding
	Rapid-708	Local call features are not active while user is logged out. During this time the call forward icon is still on though functionality is not working. Workaround: Use the keypad lock instead of logging out at the end of the day
	Rapid-698	Enabling call features for the first time after login will cause the phone to prompt the user for login again. Additional changes to the call forwarding features will not require logging in again. Workaround: Use the keypad lock instead of logging out at the end of the day.
	Rapid-109	Call forward on busy will not work in case that the phone is hosting a local conference call and the extension configured on Communication Manager with "Restrict Last appearance" as yes. Workaround: Configure the extension on Communication Manager with "Restrict Last appearance" as No.
	Rapid-47	After conference or transfer call the redial functionality will dial the first number and not the last number used to establish the conference or make the transfer.
		Call Log
	Rapid-766	E129 with Do Not Disturb feature enabled on the phone will display an incoming call as 2 missed calls instead of 1 missed call
	Rapid-742	If call log has more than 50 entries the 'Delete All Entries" option in the last page will not work Workaround: use the "Delete All Entries" option from any other page of the call log
	Rapid-26	When Call Waiting feature is disabled and E129 has an incoming call while already on a call, this incoming call will be rejected but will be logged as a missed call. In this scenario the caller of this incoming call will hear ringing for few seconds and will go either to cover path or to error tone

External ID	Internal ID	Issue Description and possible workaround if any
	Rapid-4	When calling from E129 and using call features that make use of the 2 lines (like conference and transfer) the first dialed number will be logged as the last dialed number and not the later number dialed for completing the call scenario.
Data Networking		
	Rapid-218	When using Avaya 4548GT-PWR switch and changing port configuration from 10Mbps to 100Mbps on the port that the E129 is connected the phone network will stay down. Workaround: If such change is required change first to Auto and then to 100Mbps.
	Rapid-100	Phone that is configured to static ip and detects a duplicate ip will not abstain from using it
Interworking		
	Rapid-410	When connecting to the Avaya SBC over TLS connection the SBC must be configured as not to enforce SIPS. Failing to do so will cause failure to establish calls. In phone configuration set SIP URI Scheme to SIP When Using TLS
	Rapid-685	802.1x configured for EAP-MD5 will not work with the Avaya 5650TD-PWR or 4850GTS-PWR switch. It will work with the Avaya C360 switch and Extreme switch.
Audio		
	Rapid-220	In rare cases while using headset, resuming the previous call will play audio using the speaker. This will happen if first call was established using handset and during that call the user pressed the headset w/o placing the handset back in the cradle. If the user will now answer the second call still using the headset and then will place the handset back to the cradle then if he will resume the first call the audio will go to speaker and not the headset Workaround: Place the handset back in cradle during the first call to avoid this situation or if already happened press the headset soft key to resume the audio path to headset.

errors. Changes and corrections to the information in this document may be incorporated in future releases.

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