Avaya E129 SIP Deskphone Release 1.0.2.2 Readme

This file is the Readme for the Avaya E129 SIP Deskphone Release R1.0.2.2 software. This file describes the contents of the August 2017 release software distribution package.

The build number to be seen on the phones with R1.0.2.2 is 1.25.2.52.

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

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 1.0.2.2 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 IP Office 9.0.3 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 1.0.2.2

The 1.0.2.2 Release provides a firmware update to the Avaya E129 SIP Deskphone.

E129 product has reached end of life and therefor is being worked in maintenance mode only.

The Avaya E129 SIP Deskphone is different from other Avaya SIP phones because it supports only a 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 Call, Mute, Transfer, Conference, Flash, Voice-Mail, Speaker, Volume, and Directory.
- Directory with up to 500 contacts, and Call History with up to 500 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.

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

What's new with 1.0.2.2 release

This release enhance the Trusted CA certificates management using the configuration file. The Avaya default CA certificate will be deleted upon upgrade for security requirements. See advisement section for more details.

Introduced with prior releases.
R1.0.2:

- Default settings changes:
  - Default value of Use Random Port is changed and with R1.0.2 upgrade this parameter will override prior value, unless defined explicitly in settings file On prior releases, customers were advised to change this setting in the settings file to use Random port – with this change, the admin do not need to manually set this parameter (Use Random Port - *<P78>1</P78>)*

  Default value of the "G726-32 ITU Payload" has been modified to "Dynamic". So in case of doing factory default, unless entry is specifically set in the setting file or modified by user, the "G726-32 ITU Payload" should be set to "Dynamic"; The value is not authomatically changed with upgrade, user is advised to set it to Dynamic in the cfg.xml file.
  - Default value of "SUBSCRIBE for Registration" will be set to "Yes" during upgrade. After upgrade from build 1.25.2.26, the changes doesn't take effect if previous value was no, need to reboot the phone again to apply properly (or access to WEBUI, change this value to No> save & apply, continue changing to Yes> save & apply).

- With resolution to 210 issue, phone may be now configured to enforce SIPS

R1.0.1:

- Configuration options for idle screen:
  adding speed dial soft keys and changing default display name

Features not supported in release 1.0.2.2

- 802.1x using EAP-TLS
- With Aura 7.0, E129 doesn't support aes-256 and 'SRTCP Encryption'.

E129 release 1.0.2.2 Package Content

The Avaya E129 SIP Deskphone release 1.0.2.2 package contains all the files necessary to upgrade Avaya E129 SIP Deskphone to the SIP 1.0.2.2 software.

- E129-IPT-SIP-R1_25_2_52-080917.zip which includes:
  - e129fw.bin - the E129 SIP 1.25.2.52 application, platform and DSP bin file.

  - Signatures folder – required files to support utility server upload.
- cfg.xml - The E129 config file for system specific parameter.
- idle_screen.xml file.
- Avaya E129 SIP Deskphone Release 1.0.2.2 Readme, which is this document.

Advisements with E129 1.0.2.2 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.

Before upgrading to 1.0.2.2 – When upgrading to 1.0.2.2 the default Avaya CA certificate will be deleted for security requirements. If you are using TLS connection, it is recommeneded to generate your own CA certificate and add it to the cfg.xml so upon upgrade the device will download this new certificate and be able to establish the TLS connection. Avoiding this procedure will cuase the TLS connection to fail after upgrade as there will be no CA certificate on the device

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.

It is recommended to use the polling feature for new firmware that will trigger automatically an upgrade once a new firmware is placed on the HTTP server. This will save the end user from resetting the E129 to trigger an upgrade. In order to do so set parameter P194 in the config file to yes and use P193 to set the polling interval.

Utility server upload

Starting Release 1.0.1 the packaged zip file containing this release, supports regular upload to the utility server through the web interface and release upgrade through Utility server. Please refer to the Admin guide or Utility server documentation for farther details.

E129s are not supported by the DHCP function of the Avaya Aura® Utility Server. The E129 model requires Option 66, which is not supported by the Utility Server.

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.

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.

In locked state, call may be answered by handset only (not speaker or headset) and only when “keypad is locked” popup does not appear on the screen (it will appear upon any key press).

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.

Starting with 1.0.2.1 the Avaya default certificate is not supported anymore. Before using the phone in production environment customer must install a new dedicated certificate for TLS connection. 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).

Release 1.0.2.2 enhance the TLS functionality by:

- Removing the default Avaya certificate upon upgrade to this release.
- Enable the administrator to download Trusted CA certificates

Starting 1.0.2.1 E129 verifies the server's certificates as opposed to prior releases.

• Please note that starting release SM6.3.8 (FP4), SM enhances security by removing demo certificate and that will not exist by default with new installations. The demo certificate will remain on the server in case of an upgrade and is not removed.
In case you have a fresh SM installation of FP4 or above, the initTM script should be run on SM to install the demo certificate again (please view SM documentation for farther details)

Without the correct certificate installed on the server, the E129 will fail TLS registration.

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

The demo certificate is not recommended by Avaya.

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, Google Chrome 32.0.1700.107 m and Safari.

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”

Using Avaya Aura® Communication Manager features with E129

E129 supports call forwarding features locally as described in the user guide (this means that the local activated feature will not follow the extension to another device and will not be available when the extension is logged out). 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.

Please note to not activate conflicting local and FAC features at the same time to avoid unrequired\confusing behavior.

Feature access codes are used to allow the end user to activate\deactivate various PBX features, as being administered by the PBX administrator. In Avaya Aura environment, these features are administered in CM, while on other Avaya endpoints can be administered to provide a visual indication when a call related feature is activated, Since E129 is not an AST phone, not all FACs are supported and no visual indication is available.

Following Feature Access codes were tested and are supported with E129:

Feature	Comments
Unconditional Call Forwarding	
Authorization Codes*	
Call Forward All	
Extension to Cellular (EC500)*	Cannot handoff established calls between phone and mobile

*There is no visual indication on the E129 phone if a feature is activated or deactivated.

All other FACs are not supported.

Using the E129 config file

The config file includes configurations to enhance phone behaviors to Avaya Aura environment. For ease of installation it is recommended to edit this file with the configuration needed and to use it from first installation.

Default values & using config file

Please be advised that the phone is saving the last settings configured. In case you change any settings and then comment it out or remove it from the settings file or direct the phone to a different settings file which does not contain this value – the last configured setting remains and the phone does not go back to the default.

In case you change any parameter and later on wish to revert it back to its default, please do so explicitly by configuring it back to its default value.

Anyways – executing Factory reset on the phone, will make all parameters return to its defaults.

Resolved issues in E129 1.0.2.2

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

ID	Issue Description
1028	E129 phone is responding slowly to TFTP info received in DHCP offer
1024	E129 Does not send IP addr in LLDP Chassis but MAC addr
996/1005	E129 Broken audio in beginning of calls
1026	E129 phone continue to send RTP after call ends
1000	E129 Transfer hard key doesn't do blind transfer when 1 line is in use and the other line has an incoming call.
994	E129 ‘Account name’ settings overwritten with extension
1035	E129 Cannot dial second entry from LDAP *See ticket below - LDAP number could now be dialed by using send hard key, without entering into "edit and dial" page
87/1032	E.129 phone can not make a call by SEND hard key under Phone book -> local group.
795	RTCP DLSR field is showing delay though there is no such delay
1031	For those using 1.0.2.1 – Issues uploading certificate to E129 are resolved.

ID	Issue Description
889\893	In case of network issues or packet loss, users may experience audio path issues with ongoing calls. This may result with one way audio. Customers were previously advised to change default settings and use Random Port instead of using a fixed port -this is no longer require to be done manually <!-- # Use Random Port. 0 - No, 1 - Yes. Default is 0 --> <!-- # Number: 0, 1 --> <!-- # Mandatory --> <P78>1</P78> With this release Default value of Use Random Port is changed and with R1.0.2 upgrade this parameter will override prior value, unless defined explicitly in settings file
917	With this load, in case of doing factory default, we have modified the default value of the "G726-32 ITU Payload" to "Dynamic" So after factory default, unless entry is specifically set in the setting file or modified by user, the "G726-32 ITU Payload" should be set to "Dynamic". The value is not authomatically changed with upgrade, user is advised to set it to Dynamic in the cfg.xml file
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.
210	SIPS is now supported. Please note that when using TCP transport type on E129, even if SIPS is configured, SIP scheme can only be SIP. Also, in case TCP is configured with SRTP mode, E129 will use SRTP, while other Avaya endpoinds will use RTP in this case. Please refrain from setting SIPS or SRTP along with TCP as it is bad practice.
802	In case E-129 configured to subscribe for reg event package (disabled by default), and in case is requested by the SIP controller to logged out (e.g: extension takeover, force unregister from SMGR): E-129 will logged out and then immedialtly logged in again. In the case of extension takeover, the device that tookover the extension will finally be request by the SIP controller to log out as soon as E-129 will log in again, this result in a strange behavior that device trying to login and then immedialy log out.
882	In case of a large phonebook.xml file in use (large amount of contacts) and auto login set and rebooting the phone, user may experience a second reboot and delay with phone start up. Likelihood of seeing this is up to timing and is low.

ID	Issue Description
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. Prior suggested workaround: Configure the user extension on Aura System Manager for multiple registration group or logout the E129 manually.
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).
791	E129 phone while registering receives 301 SIP message(s) and then primary SM goes down, it takes long time for outside phone call
913	Marker bit is 1 in all packets of the DTMF RTP Event
738 927	E129 validates certificate chain for the server's certificate When working against Aura 7.0 and above, after failover – phone may present one incoming call on 2 line appearances simultaneously.

Issues resolved with this load that were previously delivered with E129 1.0.1:

ID	Issue Description
863	In case of fast response from the server, calls may drop.
854	SSLV3 support is removed due to protocol vulnerabilities
772	Changing the phone language to an already installed phone will result in loss of handset functionality. Speaker and headset will continue working fine. Previously provided workaround: logout the user and re-login to recover. An alternative workaround is to reset the device after the change. Issue is now resolved with no workaround needed.
856 / 857	One way audio upon revival of many RTCP packets prior to establishing the call
870	E129 may show slowness in case Subscribe for Registration option is enabled
861 / 869	Issues on SRTP conference calls & AAC conference calls
768	Failures when making an SRTP call using media encryption "AES_CM_128_HMAC_SHA1_32 "
828	Russia time zone alignment – Autumn changed back to GMT +3

ID	Issue Description
807	# sign is not treated correctly on all scenarios
805	Phone can't download phonebook thru TFTP/HTTP/HTTPS if the login is performed by phone screen. In this case, user may upload phonebook thru WEBUI directly.
799	Payload alignment on RTP sessions
788 / 854	On some occasions port 4 was opened.
727	Attendant transfer fails in case that Communication Server is configured for SIP Endpoint Managed Transfer (SEMT). Previously given workaround – Change Communication Manager configuration to disable SEMT Issue is resolved with this release with no required workaround.
741	Network down message is shown on the phone, although phone is fully functional
726	E129 was sometimes waiting for an ACK on an Update request - On specific occasions this has caused calls to dropped 30 seconds+ into the call.
825	Redundant DHCP discover messages sent by E129
845	When failover is configured and using SRTP with SIP transport type, in case the secondary server goes down – issues may be seen on active calls on the main server.
689	Device that is configured for ip address will boot up and prompt to login providing no access to phone menu. Updated instructions when using static IP are available with R1.0.1 Installing and maintaining Avaya E129 SIP Deskphone guide. 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
793	After failover to secondary SM, several seconds delay may be noted on outgoing calls in case
61	E129 will not handle a 301 message.
779	Web interface did not show correct registration server in case of failover. Please note – in case you still face this issue with latest release, please clear the web client cache upon upgrade.
	Enable/Disable local conferencing parameter, which previously could have been added by the admin, is now visible with the published cfg.xml file. Please note that if you would like to change its default value, you first need to uncomment it.

ID	Issue Description
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. Previous 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.

Issues resolved with this load that were previously delivered with 1.25.2.1 patch -

ID	Issue Description
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.
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).
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.
357	For non-English language boot up strings were corrupted. It is now fixed and will be displayed in English.
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
676	E129 will fail to dial LDAP results that contain spaces in the phone number
695	E129 doesn’t send RTCP report.
747	Side tone is too low when using E129 handset. Users speaking with the handset might think that their voice is not transmitted.
718	Using Hands free mode in mid to high volume might generate echo on the far end.
796	SIP socket isn’t closed properly on receivable of new IP address
786 / 871	Issues such as no dial tone or reboot upon certain SIP configuration and empty header for user details
789	Extension in dial screen does not show more than 10 digits. New release will present up to 12-13 characters (depends on call duration)
866	AAC meet-me / AAM mailbox - in some occasions, in case the code is being dialed very quickly, the user may get an error attempting to join due to invalid code. Workaround: Please repeat entering the code digits in a normal pace (slower)

Unresolved issues in E129 1.0.2.2

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

ID	Issue Description and possible workaround if any
1047	E129 is not able to resume the call using Resume SK if headset SK is pressed twice on hold call
1043	E129 does not pair contact LDAP name on outgoing call after far end answers the call. In order to have pairing done correctly, user should set under Account >SIP >Basic Setting >Caller ID Display from "Auto" to "Header"
1039	When making a conference call on E129, user will be presented with End Call softkey, but that will not be functional and will not end the call. User may end the call using the in use transducer – headset, handset or speaker instead.
941	Starting R1.0.2 release, E129 TLS security was tightened. E129 will now validate certificate chain for the server's certificate using the Avaya demo built in trust certificate. The scenario of the problem is as follows: 1) Phone version 1.0.2 or above 2) Phone configured to use TLS Transport Protocol 3) Phone will succeed with registration only if the Avaya demo certificate is used on the Session manager. E129 now verifies the server's certificates as opposed to prior releases. • Certificate verification is done using the Avaya demo built in trust certificate • Built in trust certificated cannot be replaced/removed/updated. • Please note that starting release SM6.3.8 (FP4), SM enhances security by removing demo certificate and that will not exist by default with new installations. The demo certificate will remain on the server in case of an upgrade and is not removed. In case you have a fresh SM installation of FP4 or above, the initTM script should be run on SM to install the demo certificate again (please view SM documentation for farther details) Without the correct certificate installed on the server, the E129 will fail TLS registration. *** When using TLS transport type, make sure the Avaya demo certificate is installed on the SM

ID	Issue Description and possible workaround if any
2	When failover occurs during an ongoing call, the call will be dropped once the call reaches Session expiration timer. Session expiration timer may be configured through - Web UI or in the config file; Account -> Sip settings -> Session timers -> Session expiration
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).
949	In case of changing Media security mode in IPO (IPO > System > VOIP Security > Media security: change from "Best Effort" to "Disabled"), and confirming IPO reboot on the "Save Configuration" popup in IPO, the reboot will trigger E129 reboot as well. Workaround: In case you would like to avoid E129 reboot, please trigger the IPO reboot from IPO > File > Advance > Reboot
950	Issues initiating RTP calls from E129 when configuring IPO Media security as "Best Effort" Workaround: Please enable SRTP on the phone, or in case RTP is desired, configure IPO media security to Disable.
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.
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.
901	In order to apply value sip/sips of "SIP URI Scheme When Using TLS" - a reboot is required. Pleased note the reboot may be done remotely.
926	On rare occasions, in case of attempting to download a music ring tone when applying new configuration and the download speed is slow, phone may show the loading ring tone screen even after the upload is complete. In such scenario, user will need to reboot the phone in order to get full functionality back. In such scenario, please check your network environment to avoid future instances.

ID	Issue Description and possible workaround if any
902	E129 phone do not support Auto negotiation from the phone side on direct media. This would mean, for example - in case of attempting a call from E129 using TLS transport with sips to E129 using TCP transport, the call will failed. Workaround: In such scenario you may change Aura configuration to “DM (direct media) “OFF”” or may configure both endpoints to use the same transport type.
852	CM configuration clarification: On "locations" form, please make sure "Proxy Rte" is set for all locations If for some reason the field is empty, that is misconfiguration which will cause issues such as call forward failure after failover. Please put route pattern 1 for example or any other desired value under the "Proxy Rte" field on "locations"
944	When deny service is performed on the primary server while E129 is in ringing state, that specific call may still be established
929	When ERS5000 v6.3.5.024 & ERS400 v5.6.2.027 are used and user disconnects the network cable, "NETWORK DOWN" status information is showed for 2 minutes and then disappears from the screen. User may still look into the IP address field and view it is blank. After plugging in the network cable, the phone will be fully functional and will show correct status. The problem is not seen with ERS500 version 5.6.5.005 & ERS400 v5.6.2.027 releases for example.
937	When configuring to use shuffled calls on CM, there may be some audio clipping for a couple of seconds during reshuffling of the call (switching RTP to be direct). Customer may use indirect calls or consider this behavior at the beginning of each call
936	When E129 is logged in and connection with the sip controller has lost, E129 will still play dial tone if the user picks up the handset. To avoid confusion, please pay attention to the register icon \ Network Down message on the phone as indication of its connectivity and do not rely on the dial tone
919	Along with the resolution provided with R1.0.2, value of “SUBSCRIBE for Registration” changed to “Yes” on WEBUI, however still functions as if was set to No. Workaround: Reboot the phone again, phone will send SUBSCRIBE even: reg properly Or reconfigure the value under WEBUI. (Access to WEBUI, change this value to No and save & apply, continue changing to Yes> save & apply).

ID	Issue Description and possible workaround if any
904	On a very rare use case, exactly as detailed in the scenario below, mute button is not functional easily. At all-time mute indication on the phone reflects the current mute state correctly. a) Establish a call from A to B. b) Press "Hold" soft-key on A. c) Press "Mute" hard button on A. d) Press SPEAK hard button on A twice. (Or make & drop a call on line 2) e) Press "Mute" hard button on A to un-mute. After these steps, Phone A will not be able to unmute the call easily Workaround: Pressing resume & pressing Mute hard button would un-mute the call
930	In case of doing extension take over, playing with the call history submenu of the phone that logged out in the below way and then re-login with the same extension on that phone, user may lose call log information. Detailed scenario: a) A user takes over an E129 extension from a different phone. b) On the E129 – Press more soft-key twice and then press menu soft-key c) Press Call History submenu (even though it's with no information) d) Phone displays "Loading user data…" e) Re-login with same user. As a result the call log history may be lost. Workaround: In order to avoid this, in occasions in which call history submenu is available once the phone was logged out, do not press that submenu.
922	Hold and Endcall soft keys may be confusing when "Auto-Attended Transfer" was set as "Yes". Be aware: In case the call was transferred, and on the phone the call was transferred to, hold soft key is pressed – the call will not be held. If in the same state Endcall soft key is pressed, the call will be transferred back instead of ended.
921	As NAT Traversal is not supported please ignore NAT Traversal and Options selections (E.g. STUN) on the E129 web UI and cfg.xml file Do not use following setting - E129 Web GUI - Settings --> General Settings --> "Use NAT IP" and "STUN Server" are available options
925	E129 supports only 802.1X MD5. Please ignore EAP-TLS and EAP-PEAPv0/MSCHAPv2 on phone menu, WEB UI and cfg.xml Please view administration guide for clarifications.
924	As UDP is not a supported transport type, please ignore such option in the configuration, it will be removed on future releases.

ID	Issue Description and possible workaround if any
899	E129 is not supporting XcaaS. XCaaS-1.3: E129 registered via ASBCE is not failing over after Primary SM is put on "Deny New Service"
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
767	E129 that was loaded with an earlier release prior to upgrading to load 1.25.2.1 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. * Note that this does not occur on later on releases.
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.
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
859	Login the phone through the web UI is not supported (even though this option may be available). Following fields should also not be used: in sip basic setting: SIP Transport Local SIP Port In general setting: SIP User ID Authenticate ID Authenticate Password * Please review update R1.0.1 Administration guide for specific details and explanation

ID	Issue Description and possible workaround if any
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.
945	Specific issues with AAC, that may be avoided by configuration change, seen when Security Policy is set as Best Effort on AAC server Following are the issues you may experience and required configuration: E129 phone using SRTP mode "Enabled And Forced" can't resume the call after holding the call in AAC meet-me; When E129 phone using SRTP mode "Enabled But Not Forced" and Mode on SRTP is "2-srtp-aescm128-hmac32" joined conference, it may introduce audio issues to the meetme conference; E129 phone using SRTP mode "Optional" and all other modes "Enabled And Forced", E129 may have issues joining meet-me conference. (If other modes are "Enabled But Not Forced" & "No", no issue will be seen). To avoid issues joining meet-me conference: E129 phones using SRTP mode as "Enabled And Forced" or "Enabled But Not Forced" should select security policy as "SECURITY ENFORCED" on AAC. E129 phones using SRTP mode as "Optional" or "No" should select Security Policy as "SECURITY DISABLED" on AAC.
779	Web interface did not show correct registration server in case of failover. Please note – with release R1.0.1 and above, this issue is resolved - in case you still face this, please clear the web client cache upon upgrade.
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
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.
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.
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.
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

ID	Issue Description and possible workaround if any
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
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
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.
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.
100	Phone that is configured to static ip and detects a duplicate ip will not abstain from using it
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
685	802.1x configured for EAP-MD5 will not work with the Avaya 5650TD-PWR or 4850GTS-PWR switch running an old release. It will work with the Avaya C360 switch and Extreme switch. Workaround: Please upgrade the switch release to one of the following versions or above – 5.6.6 / 5.7.1 / 5.8.1 / 6.3.5 / 5.1.3 / 5.2.1 / (and future 6.6.2 once released)
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.
880	In case of pressing hold\unhold several times rapidly (3-6 times), you may need to press resume several times in order to resume the call. If this error appears - You will note that resume button is still available on the screen and the call was not resumed. Workaround: Please press release button again.
879	When the config file parameter is un-commented incorrectly (wrong syntax) - Phone will come up properly, but a core will be generated. The parameter will not take effect. Workaround: User should correct the config settings per defined syntax to be valid

ID	Issue Description and possible workaround if any
878	When user is not logged out, "log out" option still appears in the menu list. Nothing will occur in case pressed. Workaround: Please ignore it
874	In case the phone was never rebooted through the submenu or Web UI, there may be occurrences in which unplugging phone cable will cause the phone log to be lost. This is expected only on a fresh installation with no prior phone history of logging in and out.
868	In case the DHCP server is down, the E129 will lose its IP after T1 time instead of keeping previous IP.
865	E129 currently do support renaming the bin \ config files to add a pre or post fix, per customer choice. Detail configuration is mentioned below for fw upgrade using prefix and Postfix parameters through config file with drop 1.25.1.1 name as an example (please change it per the relevant release). You can also do through web page for single phone. The purpose for this is that now, ALL of the firmware with different version can be stored in one single directory, and they can be differentiated by using prefix or postfix, i.e., all files with a postfix of “1.25.1.1” belong to the firmware version 1.25.1.1 and add prefix like "Release" Now change the file name as "Release_E129FW.bin_1.25.1.1" in the stored in the file server. Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. Parameter P232 and P233 are for Prefix and Postfix for Firmware, respectively in config.xml file.. In addition, when Parameter P238 (Check New Firmware only when F/W pre/suffix changes) is set to 1, the device will only issue the Firmware Upgrade request if there are changes in the firmware Prefix or Postfix. In DHCP server in option 66 give the file server path. http://<xxx.xxx.xxx.xxx>/ In config file : 238>1</P238> - Check New Firmware only when F/W pre/suffix changes --> 12>1</P212> :- HTTP Upgrade P192>172.16.10.155</P192> :<!-- # Firmware Server Path --> P237>172.16.10.155</P237> : <!-- # Config Server Path --> <P232>Release_</P232> :<!- # Firmware File Prefix --> <P233>_1.25.2.23</P233> : <!-- # Firmware File Postfix -->
851	In case on WEB UI or cfg.xml file, primary “SIP Server” is configured as blank - upon login attempt, the phone will present “Server Not Available”, even if a secondary server is configured correctly. (This will not occur if the primary SIP server configured is invalid but not empty). Workaround: Users should not configure blanks as SIP server. Please avoid such configuration

ID	Issue Description and possible workaround if any
846	IPO: when performing blind transfer, the transfer functions correctly, but the transferor UI goes back to idle screen with no clear indication if the transfer succeeded or not. Workaround: If you would like to get a clear indication, please perform the blind transfer in the following way - Go to WEBUI > Settings > Call Settings > "Auto Attended Transfer": Select Yes. Executing blind transfer by pressing TRANSFER, dialing to target transfer, and then pressing "TRANSFER" "FLASH" button.
841	When using Hebrew language, User cannot add/remove a contact to/from a Group (such as Family, Friends, and Works) due to inaccessibility of the check box used in this specific action Workaround: user may change the language and add to the group or recreate the group.
839	When changing transport type between TLS and TCP, contact and call log entries may be lost upon next login
829	AAC8.0 issue: When DNS mode is set to default value (A-Record), the user can't call the operator. Workaround: IF DNS mode is set to SRV, this function is functional. Please refer to AAC documentation and resolutions to check resolution of this.
809	IPO: can't dial an extension followed by #
785	After upgrade, few fields in the web page under Accounts->Accounts1->Audio Settings may be displayed as XXXXX.
781	On a specific scenario, When P2329 parameter is changed to 0 (and P130 set to 2), the phone ignores P2329 parameter and it is not taking effect - phone will use TLS and SIPs Workaround: Factory reset phones: P2329 set to 0 in config file or through webUI. If the phone is already up with P2329 is set to 1 (TCP/TLS): Logout the phone - change the settings of P2329 to 0 (either through config file or webUI) - Login again (TCP/TLS)
775	In case of choosing 12 hour format and having a call longer than 12 hours, the call timer on call history will display incorrectly Workaround: when configuring 24H format, the issue will not occur
746	On IPO, when a phone does unsupervised transfer, the phone transferred to will have the call logged as a missed call - even though he answered the call.
885	When using CM version prior to 7.0 - E129 phone can't resume the call in SRTP "enabled but not forced" if far-end answers the call by using bridge line Workaround: Update CM to the latest version. As a temporary workaround until the upgrade is complete - In case bridge line is expected to be used, User may choose to set SRTP to a different configuration per his need – setting SRTP as "Enabled And Forced" (RTP won't be used) or "Optional" \ "No" (initiating outgoing call by RTP, can receive an incoming call by RTP & SRTP) the issue will not occur. This is resolved with CM vcm-017-00.0.441.0 Patch 00.0.441.0-22477 and above.

ID	Issue Description and possible workaround if any
886	In case any issues are seen on WEBUI when trying to view contacts using Internet Explorer browser, please open the WEBUI through a different browser. Workaround: You may use chrome \ Firefox \ Safari to view WebUI properly and perform actions.
884	In case you are enabling syslog and using the following settings on CM: change ip-codec-set 1 Media Encryption 1: 1-srtp-aescm128-hmac80 2: none change signaling-group 1 Enforce SIPS URI for SRTP? N When on an SRTP conference, on some rare occasions, if the conference chair hold the conversation, issues may be seen when he attempts to resume the call. Workaround: In case you encounter such issue, please turn off syslog or change one of the CM parameters.

Notice

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