uniteme 17.08

Summary

eZuce is pleased to announce the General Availability Release of uniteme and reachme 17.08.

We’ve been busy since 17.04! While we’ve been working hard on the next generation ‘Docker-ized’ version of uniteme and reachme we’ve added quite a large collection of new features and enhancements for the 17.08 release. These new features include SIP Proxy congestion management tools, configuration support for some new phones and a new SIP message diagramming tool. Unite Web user portal also continues to get additional enhancements including the ability for users to upload their greetings as MP3 or WAV files.

reachme gets a few new features, one new report and various improvements. There are some significant changes underway for the next generation reach me. The next generation will be running in Docker containers and is on its way to becoming a stand-alone product (i.e., not dependent on a uniteme installation and on its own release schedule). Additionally, the statistics storage and report generation are being moved out of MongoDB and into Elasticsearch which will provide a significant performance advantage.

Big thanks to our Partner IANT for a raft of Polycom and Yealink updates and to João Veríssimo for his work on the Grandstream plugins. An eZuce customer helped to fund and contribute back additional work for Grandstream and Zoiper phone plugins.

Also as always, hats off to the Dev & QA teams at eZuce as they have done excellent work on this release!

In all 73 issues (enhancements / fixes) are addressed for uniteme and reachme in this beta release.

In a change from our recent releases every 4 months, we’re planning on a 17.10 release as our next release. This will be a release focused on the next generation reachme statistics as well as a new uniteme option for anchoring calls in freeswitch.

Highlights

uniteme New Features:

- Zoiper Provisioning Support
- G.729 Codec included
- Optional Retry-After header in Proxy 503 Responses when Overloaded
- Proxy Congestion Management feature
- Grandstream 2130, 2140 and 2160 Phone Templates
- Allow MP3 or WAV User Greeting Upload in Unite Web User Settings
- New SIP Log Diagram tool to display SIP message flows

uniteme Improvements:

- Improvements to Yealink phone configurations
  - AutoProvision service now supports Yealink phones
  - Support for Cisco Discovery Protocol settings
  - Phone power savings setting for firmware 8.x and later.
  - Local DTMF Tone parameter
- Improvements to Polycom phone configurations
  - RealPresence Trio Firmware Support
  - Call Waiting Behaviour
  - Device Base Profile (Ensure Generic for phones that may have shipped as Lync phones)
  - Firmware 5.5.2 support
- Display calling number and caller-id in Unite Web Voicemail page
- Flexible automatic Phone Line label generation for Polycom phones
- Allow SipRedirectorPickUp port to bind to other TCP/IP Ports
- gridfs-voicemail-cli.jar command line tool now allows upload of voicemail greeting
- New REST Calls

reachme New Features:

- Block transfer to queue when there are no eligible agents
- Added ‘agent’ specific skill to remove skill in a recipe action
- On Hold Counter in the Agent Session Manager to show how long a call has been on hold.

reachme Improvements:

- Improved the outbound user experience
- Add Calling Number to Filters in the Reachme Call Recording widget
- Report Improvements:
  - Agent Productivity Reports get ‘Available Time’ column added.
  - Allow Reach reports that use Interval Data to hide intervals with no activity.

Who Should Install?

This release is recommended for all 4.6 and later installations. If you have a patch installed to your system a new patch may be required. Please contact sa@ezuce.com if think you may have a patch applied as that may be replaced during the update.

eZuce's software products continuously progress through an Agile based development methodology that keeps feature functionality comprehensive and up-to-date in response to evolving market and customer requirements.

New software releases are made at a rate of four to six releases a year. Releases are numbered in the <yy>.<mm>.<uu> format where <yy> and <mm> designate the year and the month, respectively, in which a release is made generally available. Where applicable, <uu> corresponds to an update release relative to a general release on which fixes are made available.

In order to ensure service continuity and stability, customers may keep their production environments unchanged for up to a 6-month period during which release updates or patches are made available. After a release is more than 6-months old, eZuce customers would have to upgrade to the latest generally available release - inclusive of all fixes to date and any new patches.

Questions

If you have questions about updating you can email sa@ezuce.com or if you need assistance with the update contact your account manager or email sales@ezuce.com.

Software Release History

We're currently running on a 4-month release cycle.

- April release for 2016 is 16.04
- August release for 2016 is 16.08
- December release for 2016 is 16.12
- April release for 2017 is 17.04
- August release for 2017 is 17.08

Release Level History

- 14.04 - April 30, 2014
- 14.04.1 - June 01, 2014
- 14.04.2 - July 11, 2014
- 14.04.3 - October 24, 2014
- 14.10 - February 5, 2015
- 15.04 - April 29, 2015
- 15.05 - May 27, 2015
- 15.06 - June 30, 2015
- 15.08 - August 31, 2015
- 15.10.1 - December 9, 2015
- 15.12 - January 6, 2015
- 16.02 - March 14, 2016
- 16.04 - May 31, 2016
- 16.08 - October 6, 2016
- 16.12 - January 17, 2016
- 17.04 - April 18, 2017
- 17.08 - September 7, 2017

System Requirements

For a reasonably performing system, we recommend the following configuration.

Minimum hardware requirements

- Pentium 4 or Xeon processor @ 2.0 GHz Core 64bit or higher
- Minimum 4 GB of RAM with sufficient swap space
- 80 GB disk (75 users depending on usage patterns)

Notes:

- **uniteme** supports an unlimited number of voicemail boxes, the total number of hours of recorded messages is determined by the size of the hard-disk. As a rule, for every minute of recorded messages, you will need 1 MB of disk space (About 3 hours per 10 GB of disk space).
- **reachme** requires more memory, processor and disk space. Please consult with eZuce SA team for your specific installation.

Operating System

CentOS/RHEL 6 x86_64 with latest updates is required.

Devices

Phones

- **Polycom VVX Devices** with firmware 5.5.2 (split) are recommended for new installations
- **Polycom SoundPoint IP** Devices should run firmware 4.0.11 (split)

Gateways

- **AudioCodes Gateways** are recommended for PSTN connectivity

SBCs

- **Frafos, Sangoma, AudioCodes, Acme Packet and Ingate** SBC's are recommended for SIP Trunking and Remote Worker connectivity (commonly referred to as sipXbridge and MediaRelay services respectively).
- **NOTE:** The eZuce unite**me** - "Use built-in SIP Trunk SBC" found in Gateway Details for use with Trunking or Remote Worker solutions should be used only for lab purposes. The openUC "Built-In SIP Trunk SBC" (sipXbridge) will not be supported in any production or live environment. Additionally, sipXbridge does not work in an HA environment.

Documentation

Technical Reference Manuals, User Guides, Reach Reference Manuals, and other technical and user information can be found under the following link: Do cumentation Page

Installation and Upgrade Notes

Installation note

After **unite****me** 17.04 is downloaded and installed, the *clusterId* read tag is unique (same as *locationId*). Follow these steps to propagate the new read tags to the MongoDB replica set:

1. In the **unite****me** menu, click *System>Database*.
2. Click the *Add query metadata* button.
3. To verify that the MongoDB replica contains the unique read tags, run from the command line:

```
//mongo
rs.config();//
```

Special MongoDB note

Please be aware of these Mongodbrequirements http://docs.mongodb.org/manual/reference/ulimit/ **Note:** Both the “hard” and the “soft” ulimit affect MongoDB’s performance. The “hard” ulimit refers to the maximum number of processes that a user can have active at any time. This is the ceiling: no non-root process can increase the “hard” ulimit. In contrast, the “soft” ulimit is the limit that is actually enforced for a session or process, but any process can increase it up to “hard” ulimit maximum.Every deployment may have unique requirements and settings; however, the following thresholds and settings are particularly important for mongod and mongos deployments:

```
ulimit –a
-f (file size): unlimited
-t (cpu time): unlimited
-v (virtual memory): unlimited
-n (open files): 64000
-m (memory size): unlimited
-u (processes/threads): 32000
```

Always remember to restart your mongod and mongos instances after changing the ulimit settings to make sure that the settings change takes effect.If you limit virtual or resident memory size on a system running MongoDB the operating system will refuse to honor additional allocation requests. After every install/upgrade please check that "cat /proc/$pid_of_mongo/limits" have the recommended value of 655350. To make this value permanent you need to create this file */etc/security/limits.d/99-mongodb-nproc.conf* and add the following lines:

```
mongodb soft nproc 64000

mongodb hard nproc 64000

mongodb soft nofile 64000

mongodb hard nofile 64000
```

Special Patch Note

If you have a patch installed to your system a new patch may be required. Please contact sa@ezuce.com if think you may have a patch applied as that may be replaced during the update.

Installing from ISO image

Download **uniteme** ISO

Download the ISO image corresponding to your hardware and write the image to a DVD.

- The ISO files are available here: https://download.ezuce.com/openuc/ISO/
- You will need a valid ezuce.com user IDtologin and download.
- We recommend the 64 bit installation in most cases. This ISO file name ends in **x86_64.iso**

Install **uniteme**

- Boot from the DVD created with the **uniteme** ISO image.
- Press **Enter** at the boot screen below to begin the unite**me** installation.
- Select **Manual Configuration** under **Enable IPv4 support** and select **OK**.
- Set a static IPv4 address with the corresponding networking information and click OK.
- In certain situations, a warning of the use of indicated storage devices will be displayed.
- Select the language to be used during the installation.
- Select the keyboard layout to be used.
- Select the timezone to be used.
- Set a root password.
- Login to the system as root with the password you provided earlier and continue on to the Configure of unite**me**.

Installing from Repository

uniteme can be installed using the following procedure

- Download and install CentOS 6.x minimal ISO
- Run the following command:

```
yum update && reboot
```

- Run the following commands to retrieve and run the eZuce **uniteme** installer:

```
curl https://download.ezuce.com/openuc-setup > /usr/bin/openuc-setup
chmod +x /usr/bin/openuc-setup
openuc-setup
```

This utility will guide you through the process of installing unite**me** from the eZuce software repository.

Upgrade from previous versions

Modify the repo file in /etc/yum.repos.d and replace the baseurl= with the location of the repository you'd like to upgrade to.

Identify any existing 'rpmnew' or 'rpmsave' files on the system with:

```
find / -print | egrep "rpmnew$|rpmsave$"
```

As root, execute the following commands:

```
yum clean all
```

```
yum update
```

Note any additional 'rpmnew' or 'rpmsave' files that may have been created by running find command again

```
find / -print | egrep "rpmnew$|rpmsave$"
```

If there are any files that didn't get overwritten by yum, please see 'Modified Files Upgrade Note' information below.

A system reboot after the update has completed is recommended.

SEC Service Upgrade Note

When upgrading unite**me** from openUC 4.6 Update 11 or 14.4.3 to 15.06 follow these steps to ensure the SEC service is correctly running:

- 1. Upgrade from 4.6 Update 11 or 14.4.3 to 15.06.
- 2. After the upgrade is complete, perform the usual restart.
- 3. Once possible, connect via CLI and monitor processes using top. Notice that the SEC process is using a lot of CPU memory.
- 4. Perform another restart OR restart only the Sipxlogwatcher service.

Modified Files Upgrade Note

If you have manually modified any system related files or some files are not as yum would expect them to be, the yum update process may not overwrite them. It will instead create 'rpmnew' or 'rpmsave' files and not overwrite the files. The administrator may have previously modified the files knowingly or as part of a patch supplied by TAC.

To check your upgrade.log and search for *.rpmnew *.rpmsave on your system check the upgrade log:

You will be responsible for merging any changes from the old file to the new or contacting Technical Support if you require assistance.

Support Tips and Contact Information

Please see the Getting Support section for support tips and support contact information

Issues Sorted by Issue Number

	JIRA name	RN Content	Enhancement/Fix/Known Issue	Keywords
SIPX-159	Yealink auto-provisioning	An administrator would like Yealink phones to auto-provision in the same manner as Polycom phones.	Enhancement	Yealink
SIPX-573	Grandstream profile fails if advanced parameter is used	Fixed an issue generating Grandstream phone profiles if an advanced parameter was used. Steps to reproduce: 1. devices -> phones -> create new, grandstream gxp2010 2. Under custom configuration -> show advanced, add a P Value of P2910=1, apply 3. send profiles Expected: phone profile to be created Actual: phone profile fails to be created	Fix	Grandstream
SIPX-604	Upgrading sipxgrandstream doesn't delete all cfg files	Fixed an issue caused when installing sipxgrandstream 17.08 where some config files are left in the config directory.	Fix	Grandstream
SIPX-608	Polycom UseTelUriAsLabel improvement	An administrator would like to allow the use of the Polycom Firmware parameter for UseTelUriAsLabel. This parameter was limited to 4.1.8 template. This parameter is now available in newer Polycom Firmware >= 5.3.1.	Enhancement	Polycom
SIPX-611	Yealink Power Saving	An administrator would like to be able to configure the power saving settings for Yealink phones. This should be available for Firmware 8.x and later.	Enhancement	Yealink
SIPX-612	Yealink CDP Provision	An administrator would like to be able to configure CDP (Cisco Discovery Protocol) parameters in the Yealink Phone Configuration Plugin static.network.cdp.enable (default 0) static.network.cdp.packet_interval (default 60)	Enhancement	Yealink
SIPX-613	Settings evaluation with regular expression	A developer would like to be able to utilize regular expressions in provisioning plugins. This enhancement will make regular expressions available for all settings evaluation and support the old writing with || Current Settings Evaluation utilizes compare and supports optional settings separated by || Some Provisioning Plugins are already using Regular Expressions to do this.	Enhancement	Polycom sipXconfig
SIPX-615	Polycom Firmware 5.x.x	An administrator would like to be able to specify Polycom Firmware up to version 5.5.2.	Enhancement	Polycom
SIPX-616	Yealink CP-Source Parameter	Enhancement to allow a Yealink phone to use the Contact Header to display the calling party source. In Yealink 8x (x.80.250.x) the parameter cp_source has been improved. Now phones can use the Contact Header to display the calling party source. This is useful if you want to show the callee number after pickup or from hunt groups.	Enhancement	Yealink

SIPX-617	Make phone line label configurable	An administrator would like to control a phone or phone group to either have the phone line label left null in the Polycom phone configuration or have the phone configuration populated with the line number. UC-3903 implemented a change for Polycom phone line labels in reaction to Polycom changing the default behavior of their firmware. In 4.x and earlier firmware if the line label was blank, the line extension was used. In 5.x and later if the line label was blank the name associated with the line was used. Our change for UC-3903 looks at the database settings for a phone and if that is null or empty we put the line number into the phone configuration to defeat Polycom's change. This should continue to be the default behavior.	Enhancement	Polycom
SIPX-618	Polycom config parameter for new Polycom phones	Fix for some parameters in the phone provisioning that are model based. Example: Background of VVX 500 phone has specific parameter and for VVX 501 its the same. Current checks ignore the 501's	Fix	Polycom
SIPX-620	Jitsi preferred transport	A fix to correct parameter name in Jitsi provisioning plugin. PREFFERED_TRANSPORT -> PREFERRED_TRANSPORT	Fix	Jitsi
SIPX-622	Add opensource G729 codec to Freeswitch	An administrator would like to be able to enable/disable the use of the belladonna communications open source version of G729 codec in Freeswitch. The g.729 protocol is now unencumbered and customers are free to use this. https://github.com/xadhoom/mod_bcg729 To Enable: 1. Login to Linux box as root and execute: "yum install mod_bcg729" 2. In Media Settings page select the G729 codec driver 3. Send profile to Server, Freeswitch configuration should pickup latest changes	Enhancement	g729
SIPX-624	Yealink Resource List Subscription	Yealink plugin should check if a phone line has BLFs before setting the resource list subscription URI	Enhancement	Yealink
SIPX-625	Polycom Call Waiting	Add parameter to configure call.callWaiting.enable and call.callWaiting.ring for Polycom phones	Enhancement	Polycom
SIPX-626	Polycom RealPresence Trio POE configuration	Add parameter to configure POE for LAN out port and USB charging on Polycom RealPresence Trio poe.pse.enabled usb.charging.enabled poe.pse.class	Enhancement	Polycom
SIPX-627	Polycom RealPresence Trio Firmware	The Polycom RealPresence Trio uses a special firmware revision. This is an Enhancement to add a seperate version tree for Trio to seperate Trio Firmware from other Polycom phone firmware. Example: FW 5.4.0 for VVX/SoundPoint/SoundStation/... FW 5.4.0_Trio for RealPresence Trio Trio starts with Version Number 5.4.0. The latest current release (Mar 31, 2017) is 5.4.6	Enhancement	Polycom
SIPX-628	~~sipXprovision user has to much permission by default	The sipXprovision user which is used for auto provisioning has to much permission. To reduce load to MWI in Polycom provisioning there is an additional check if a user has voicemail permission. If they don't have voicemail permission, the phone isn't provisioned to subscribe to voicemail. This was added for Yealink as well. As the autoprovisioning feature for Yealink was added a developer noticed that the voicemail settings are provisioned. After some research it was determined that the provisioning special user has all default dialing permissions which isn't necessary. All dial permissions for the provisioning user are now disabled.	Fix	sipxconfig
SIPX-638	sipxcom 17.04 social links not in footer	The social links are not displayed in the footer within any tab beneath System -> Servers, or Diagnostics -> CDRs	Fix	sipxconfig
SIPX-639	Users invited to conference rooms use only g711u	In europe g711a-Law is mostly used so some ISPs only accept g711a but not g711u. If a non-encoding SBC is used and a user is invited to a conference room via Web Portal, than the config server produces a Invite with u-law only. Some of the bigger ISPs in Germany refuse such requests with "not acceptable here" (with a cause of no supported codec). Look for the configured codecs of media services and use them only but as workaround it would be sufficient to add a-law with lower priority.	Enhancement	Conference
SIPX-640	Polycom Device Base profile	Add a new config parameter for Polycom devices device.baseProfile Values are: Generic, Lync Why this is necessary: Normaly the default value is "null", so Generic is choosen. If you buy a phone which is marked as "Lync", this is automatically choosen. We need to ensure that "Generic" is configured to get those phones working with SipXcom/UniteMe Parameter available in Firmware 5.3.x and later.	Enhancement	Polycom
SIPX-641	Grandstream GXP 2160 phone profile XML errors	Fix for 2 errors in the xml config file that is created for Grandstream GXP 2160 phones. <!--OpenVPN Client Key --> <#P9904></#P9904> <!--Prefix for dialing password 0 --> <26049></26049> This should be : <!--OpenVPN Client Key --> <P9904></P9904> <!--Prefix for dialing password 0 --> <P26049></P26049>	Fix	Grandstream
SIPX-646	Yealink BLF 7x vs 8x	Fix difference between 7x and 8x for BLF configuration	Fix	Yealink

SIPX-648	Yealink Play Local DTMF Tone	Enhancement to add a parameter to enable/disable local playback of DTMF tones for Yealink phones. Parameter is: features.play_local_dtmf_tone_enable Currently there is parameter with this description. This is not correct. From current property file: features.AUDIO.features.partition_tone.label=Play Local DTMF Tone	Enhancement	Yealink
SIPX-72	Add support for Grandstream GXP 2130, 2140 and 2160	Enhancement to add support for Grandstream GXP 2130, 2140 and 2160 phones.	Enhancement	Grandstream
UC-2575	Extend gridfs-voicemail-cli.jar script to allow copying a prompt into a user's mailbox prompt	An administrator would like to be able to copy a pre-recorded greeting into a user's mailbox. The existing script allows moving files but not copying a new prompt into the user folder.	Enhancement	voicemail
UC-3199	Block transfer to queue when 0 eligible agents	This feature is intended to allow a Reach administrator to configure a queue so that it will NOT accept transfer/conference calls to enter it if there are 0 eligible agents for the defined skill combination. Config Side work: add a field that is a checkbox. Label it "Block Transfer with 0 Eligible Agents" this new field is a checkbox that is not checked by default Reach Side work: When an agent is in process of transferring or conferencing via Queue, they can currently select skills via check boxes and the there is an eligible agent count provided as this active occurs. For this new feature, we should disable the transfer and conference buttons when the eligible agent count = 0. This should ONLY be done for a queue that has the "Block Transfer with 0 Eligible Agents" checkbox selected. If the Block Transfer with 0 Eligible Agents check box is not checked, there should be no call processing changes.	Enhancement	Reachme
UC-3669	Feature request - Allow greetings to be uploaded from user portal	A users would like to have the ability to upload pre-recorded greetings in WAV or MP3 via the unite web interface.	Enhancement	UniteWeb voicemail
UC-3829	Reachme On Hold Counter	Add a counter to the Session Manager that shows how long a call has been on hold. Add this below the Wait time indicator. This counter will not be displayed if the call is not on hold. It will be displayed and starts when a call is put on hold. It will be no longer displayed when a call is brought back off of hold. If the call is placed on hold again, display the hold counter again but restart it at 0.	Enhancement	Reachme
UC-3865	Add Calling number to filters in the Reachme Call recording widget	Add the calling number as search criteria in the reach call recording widget. Currently you can't search for recordings based on a calling number. This feature request is intended to allow for that. Add the called number search under the Search for Call ID text box. It should also be a text box where the user can put in a free form phone number. As is done with the "search for call id" filter, include some dim text in the text box of 'Search for Caller Number' indicating that the field is to be used to search for call recordings that are for calls from a particular caller. Ensure that the number that is being searched/matched here is the phone number as it appears in the reach cdr (Caller ANI field) rather than some form of caller ID that perhaps includes other data.	Enhancement	Reachme
UC-3909	Agent gets redirected to "undefined" or bad logout link	Fixed an issue where if a client loses its network connection to the Reachme server, then in about 17 seconds (it will take longer if you have widgets) the dashboard will prompt "Reconnecting" and stay like that. When the connection comes back up, it was observed that (rarely) the dashboard will either continue showing the same "Reconnecting" message for a period of time or most often what will happen is the agent will be prompted by an "Unable to establish a connection to server" message. Once OC is clicked in this message the agent will (most of the time) be redirected to the dashboard login page or shown a blank page with link fqdn/reach/portal /undefined or sometimes fqdn/reach/portal/logout=w...something. This seems to be very consistent with redirection (the redirect issue might be an older issue, not reported by customers, and could probably be caused by non-trusted certificates?) Does not replicate consistently.	Fix	Reachme
UC-3916	Sort Reachme CDRs by date and time by default	Fixed an issue with CDR Reports so that they get sorted by default by date and time as would be expected.	Fix	Reachme reports
UC-4185	Only part of the table header visible when scrolling down in CDR report, table header overlaps with input controls section when scrolling right	Fixed an issue with CDR reports where if the report has enough entries that when a user scrolled through it, the row with column names is sticky, it scrolls along when you scroll down, however, only part of it is visible. Not a huge issue, just looked odd.	Fix	Reachme reports
UC-4197	CDR Brief, CDR compact vs CDR inconsistencies and typos	Fixed some issues with inconsistencies between CDR and CDR Brief/Compact reports. 1. Column named Originating Caller ID vs Caller ID in CDR brief report 2. Time stamps having both Date and Time in the CDR brief report 3. "Time in Queue/Preca" instead of Time in Queue/Precall in the CDR brief report 4. There is also a "Totall" with double L there in CDR Brief 5. CDR compact has columns Endpoint ANI/Caller ID instead of Endpoint ANI/ Originating Caller ID and Date/Time instead of Date/Offered to Reach 6. It's the only report which is missing a capital letter :D "CDR compact".	Fix	Reachme reports
UC-4203	All "detail" reports can't handle From date bigger than To date	Fixed an issue caused when the user entered a later From date for a report than the To date. Only the "detail" reports (outbound,line,client,queue) will error out when you accidentally want to generate them with incorrect dates. Incorrect means start date is after the end date. Other reports won't display errors and will handle this mistake fine.	Fix	Reachme reports
UC-4289	Reach skill removal does not take effect if executed while ringing an agent	Fixed an issue where Reachme skill removal would not take effect if it was executed while a call was ringing an agent. If a skill removal occurs during the period of time that the call is being offered to an agent, the skill is not taken off the call. At least not permanently. Example: Call enters the system into a queue that has a recipe step to remove a skill at 10 seconds. From 5 to 17 seconds, the call is ringing an agent and the agent does not answer the call. Once the call stops ringing the agent you will see that the skill is still on the call.	Fix	Reachme

UC-4320	Re-queue mechanism in case of network interruption does not work as expected	Fixed an issue where a call should be routed to an agent, and his phone should start ringing after a netwok interruption to the Agent Portal. The agent's phone however would start ringing for less than 1 second and stop. Steps to reproduce : Prerequisites: - 2 Agents available (200, 201) ready to receive calls. 1. Customer calls in queue 2. Agent 200 answers the incoming call alert, there is 2-way voice path. 2. Agent 200 PC network connection is interrupted for aproximately 15 seconds. (Agent 201's network connection is unafected). Call is disconnected at the end of those 15 seconds. 3. Agent 200 dashboard will display "Reconnecting" and after a few seconds "The server has terminated your session". 4. Call is offered to remaining available clients - Agent 201. Expected result : call should be routed to Agent 201, his phone should start ringing. Actual result : Agent 201's phone will start ringing for less than 1 second and stop. Agent 201'dashboard will not show any change.	Fix	Reachme
UC-4332	"This Month" queue statistics incorrectly computed	Fixed an issue with Reachme statistics where they showed that "This Month" got populated with statistics based on calls that a user just placed on/since Tuesdays whereas it would be expected that statistics should show the number of calls placed from the beginning of the month. 1.On Monday placed 10 calls to populate Queue statistics. 2. Verified on Tuesday - the number of calls placed on Monday for "This Month" statistics - 10 calls. 3. Placed 10 more calls on Tuesday and verified "This Month " statistics again: Issue : statistics showed that "This Month" got populated with statistics based on calls that i've just placed on Tuesday: 10 calls. Expected: statistics to show the number of calls placed from the beginning of the month.	Fix	Reachme reports
UC-4344	Improve the Reach outbound user experience	Currently, when a reach agent is placing an outbound call, they can select a hard coded option of "No Customer". This option is not a real client in the system. Specifically, it's not built by the admin as a client. In some environments, call center managers would like to block the capability of allowing an agent to place an outbound call that is not associated with a specific client. Furthermore, there are use cases where the call center management does not care about a client specification for agent outbound calls. Therefore there is a need to allow more flexibility around this client selection for reach outbound calls. In addition, it would be a simpler user experience if the agent could type in the number that they want to call, then click the send button. Once they click send, the system would call their phone, wait for an answer, then immediately call the other party. Currently, they select a client from the drop down and that triggers the call to their phone. They then have to answer the call before they are shown a text box that will allow them to put in the number that they want to call. This feature request includes the ability to conditionally remove the "no customer" option and allow a default client for outbound calls. These changes will be driven by the agent group that the agent belongs to. Specific changes required for this part of the jira are as follows: At the bottom of the agent group page, add 2 things: Include "No Customer" Client Option During Outbound (checkbox) Default Client During Outbound (single select drop down of clients in the system) Include "No Customer" Client Option During Outbound will default to checked. Default Client During Outbound will default to no client being selected. The admin will be able to edit these settings by checking/unchecking or by selecting a single client from the drop down. The single select client drop down box will have all of the clients that are built in the system PLUS a special entry of "No Customer". The admin will select none of the entries or one of the entries. The Agent UI will be changed such that the text box allowing a user to enter the number that they want to dial will be shown at all times if they have the outbound calling permission. The "send button will also be visible at all times but only enabled when a number is in the text box AND a client is selected in the client drop down box. Reach will abide by these new settings in the following ways. If the Include "No Customer" Client Option During Outbound is checked; No Customer option will not be in the drop down option for an agent that belongs to this agent group during the outbound call process. If checked, the "No Customer" option will be in the drop down of the agent UI. If there is a client configured in the Default Client During Outbound drop down for the agent group, the agent will NOT have to select a client. Instead, the default client will be populated in the drop down automatically and they will be able to simply place a phone number into the text box to where they place the number that they want to dial and click the send call button. The system will immediately call their phone, wait for it to be answered, then place the call to the other party. The agent will be able to override the default client by using the client drop down and selecting another option. Again, to reiterate, once a client is selected (manual or by default setting) AND a phone number is placed into the text box, the send button will be enabled. During the entire process of beginning an outbound call, the X button to end the process/call will not be displayed. This X button will act as it does today, meaning that it will only be displayed and enabled once the call to the agent phone is placed. In short, the system will now allow an agent to simply place a number in that they want to dial and click send. This would be enough to place an outbound call assuming there is a default client configured in the system.	Enhancement	Reachme
UC-4360	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to change the port that SIP Redirector is bound to in sipXconfig. This is config part of the work for UC-4361: Extend SIP Registrar configuration for pickup redirector plugin bind port with port 5085 as the default. In registrar config file (/etc/sipxpbx/registrar-config) it should look: SIP_REDIRECT.100-PICKUP.BIND_PORT : 5085	Enhancement	SIP sipxconfig
UC-4361	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to configure the port that SIP Redirector binds to for Call Pickup. This is registrar part: Get value of SIP_REDIRECT.100-PICKUP.BIND_PORT and use it during initialisation of SipRedirectorPickUp. If no value present in config file use current PORT_DEFAULT value and system will select port on it's own.	Enhancement	SIP
UC-4380	Improve the Reach outbound user experience - UI work	Make changes to UI for Reach outbound calling improvements as identified in ticket UC-4344. This ticket is intended to be JUST the UI changes required. Initial UI side work will use "No Customer" as the default client. This will be addressed again later once UC-4344 is addressed and the config and Reach back end pieces are in place to send the default client to the UI.	Enhancement	Reachme
UC-4383	Active CDR Page broken when umlauts are used	Fixed an issue with the Active CDR page in Admin GUI where when a User Agent has umlauts in the name, sipXconfig would not display the call detail properly. The problem is on sipXconfig side. If the UA encodes special characters the problem doesn't reproduce. Polycom seems it is not encoding and SIPXCDR API such that config calls to retrieve active calls works as expected. There seems to be a problem on sipxconfig side which may not be handling them properly.	Fix	CDR sipxconfig

UC-4385	Idle agent not being able to receive a call after websocket disconnect and call requeueing is set to false	Fixed an issue with idle Reachme agents where they were not able to receive a call after they lost their browser websocket connection and if call requeueing is set to false. Steps to reproduce: -Requeue Call on UI Failure: false -While agent is oncall with customer, a network disruption occurrs for more than 15 seconds, his websocket connection dies, since Requeue Call on UI Failure is set to false, call will remain up. -As soon as the timeout on the client side is detected, Agent is being presented with the following message " Network disruption when agent is oncall, you will be redirected to login page". -Upon redirection to login page and while the call is still up, Agent will be presented with another message if he tries to log in: "Do you want to terminate your previous session? There is a call in that session which has not been handled yet. That call will be terminated if you choose to terminate the session. " -Agent forcefully terminate his session by pressing "Yes" on the prompt, at which point call is requeued - should not happen. -Agent goes to IDLE state, a new call enters the queue, but the idle agent is not able to receive it.	Fix	Reachme
UC-4386	Idle agent is not assigned to available vqueues after websocket disconnect	Fixed an issue with an edge case scenario involving an issue identified during testing for UC-4359/UC-4316: - Sending events to the client fails with an exception. - Sometime later the agent logins and tries to register with all vqueues. - There is one vqueue but the agent is notified that the aren't any.	Fix	openacd
UC-4397	Group firmware gets reset after upgrade	Fixed an issue with Group Firmware being reset to default when upgrading a system to 16.12. After the upgrade, in the Device > Phone Groups > any group > any phone, the Group Firmware version was reset to default.	Fix	sipXconfig
UC-4398	VVX 1500 profile voice codec typo	Fixed an issue with the Polycom VVX 1500 configuration plugin. There was a typo in the voice codec settings.	Fix	Polycom
UC-4400	Sometimes not getting redirected to dashboard after clicking the Continue button	Fixed an issue with the Reachme console where the user is not redirected to the dashboard properly after clicking on the Continue button. This occurred if the user had a slow network connection. Symptoms: Instead of the dashboard the user sees: {"redirect_location":"/reach/portal/dashboard","token":"JaWmoYK4tmARZ04....} From Browser Console: Request URL:https://one.ctradu-ng.ezuce.ro/reach/portal/login/advanced Request Method:POST Status Code:200 OK Remote Address:10.5.0.210:443 Referrer Policy:no-referrer-when-downgrade Response: {"redirect_location":"/reach/portal/dashboard","token":"JaWmoYK4t... Console: Resource interpreted as Document but transferred with MIME type application/json: "https://one.ctradu-ng.ezuce.ro/reach/portal/login/advanced". I am assuming this is because I tried to Login via https but can someone please confirm? I attached relevant part of reach log.	Enhancement	Reachme
UC-4406	Allow Reachme reports using Interval Data Source to hide intervals with no activity	A Reachme user would like to be able to hide intervals with no activity in the reports that utilize Interval Data Source. Changes need to be made in these reports: - Queue Traffic Detail - Client Traffic Detail - Line Traffic Detail - Voicemail Detail - Outbound Detail by Client - perhaps in Agent Availability report ?? The solution will be based on checkbox input control allowing to hide intervals with no activity	Enhancement	Reachme reports
UC-4421	Default behavior for blind transfer on Polycom FW > 5.3.0	Due to changes in Polycom firmware, the old functionality for Blind transfer feature performed by hitting Transfer and then the Blind softkey is now gone. Users must now press and hold the Transfer softkey button, push down, push select "Blind Transfer" and then input their destination. If "Blind Transfer" is used more often than "Consultative Transfer" this can be set as the default option for transfer type in Preferences > Additional preferences > Default Transfer Type page in the Polycom configuration page. Also, the procedure used in the pre 5.3 versions to create a softkey with a macro(efk) to do one-button blind transfers seems to no longer work.	Enhancement	Polycom
UC-4422	In session inbound not aligned properly in Firefox, default zoom	QA would like so fix a small esthetic when running Firefox with 100% zoom on two different resolution 1366x768 & 1920x1080, the In Session Inbound is misaligned in the agent widget.	Fix	Reachme
UC-4424	Disable ouc_agent_report_job module in production code	A developer would like to disable some code that aggregated data from rstats to agent_report collection. This code isis not used anymore but the developer would like keep the code in-tact for debugging purposes. ouc_agent_report_job erlang module aggregates data from rstats collection to agent_report collection, which is not used anymore and contains inaccurate data. We should disable this module in production version, but still keep its code in project for debugging purposes. More details in this doc: https://docs.google.com/a/ezuce.com/document/d/1S6gq9QGae8jZVVhx1hrHFwtPeDWp_R7O62InEv1C4cc/edit?usp=sharing	Fix	Reachme reports statistics
UC-4431	Call recording playback issues when Reach is not running on primary	Fixed an issue with call recording playback that caused playback to not work when Reachme was not running on a primary cluster server. Steps to reproduce: Cluster with 3 nodes, primary config on node 1 and reach running on node 3. Make a recorded call and try to listen that call from supervisor : --> it will display "audio not available" message Problem was narrow it down to a redirect to localhost. Customer found an workaround to re redirect towards to the primary configuration node FQDN	Fix	Reachme
UC-4432	Correct action-on-timeout feature in reach_ouc/src/reach_vqueue.erl	Fixed an issue with the action-on-timeout feature as introduced by UC-4432. As mentioned in UC-4385, there is a potential problem with scheduling an action upon timeout set in the process' init function: If an integer time-out value is provided, a time-out occurs unless an event or a message is received within Timeout milliseconds. There are two issues: Timeout = int()>0 | infinity (not allowed to be 0), and the API doesn't guarantee that the timeout has to occur. For the record, this problematic change has been introduced with fix UC-2840 in: https://github.com/ezuce/reach-app/commit/2c630609e1b62ce4bad5cb99737a8f82b09824bd A similar change has been added to apps/reach_ouc/src/reach_vqueue.erl which will also need to be corrected in case this fix works as expected. Fix for UC-4385 has been verified and this issue is to track a similar change in apps/reach_ouc/src/reach_vqueue.erl.	Fix	Reachme

UC-4435	Add Retry-After in 503 proxy response	An Administrator would like to be able to have UA's sending requests to proxy be able to back off if the proxy is overloaded. As determined by RFC 3261 - https://tools.ietf.org/html/rfc3261#page-176 and RFC 5390 - https://tools.ietf.org/html/rfc5390, in the 503 reply, we have added an optional Retry-After header such that phones that support this RFC will know when to retry. We currently don't do that, this is how the message looks at the moment: OUTGOING.. ...nSIP/2.0 503 Queue Size Is Too High From: sipxtest <sip:sipxtest@10.2.0.174>;tag=23289SIPpTag001683 To: 1238 <sip:1238@cluster.ezuce.ro>;tag=fzD_qG\r\n Call-Id: 1683-23289@10.2.0.174\r\n Cseq: 1 INVITE\r\nVia: SIP/2.0/TCP 10.2.0.174:8100;branch=z9hG4bK-23289-1683-1\r\n Date: Wed, 03 May 2017 13:39:44 GMT\r\n Content-Length: 0\r\n\r\n --------------------END	Enhancement	sipxproxy
UC-4436	Decrease default number of backups to retain	Currently the default number of backups is set to 50. If an administrator doesn't pay attention to that field they can quickly run out of space. Decrease this number to 5 by default.	Enhancement	backup sipxconfig
UC-4437	Phonebook REST improvements	An administrator would like to add entry-id endpoint to the my/phonebook rest call. This should return all entryId's so you should not be forced to get them from psql queries /my/phonebook/entry/{entryId}	Enhancement	REST
UC-4439	Add config support for Zoiper soft phone	A customer would like to introduce a soft phone client into their environment that is usable on a myriad of different operating systems. It has been determined that the most appropriate softphone for their purposes is Zoiper since it will work on linux, MAC, Windows, Android, etc. The customer worked with Zoiper to identify changes that will be required on their side and eZuce met with Zoiper to better understand how to best implement a provisioning model for their softphones. The result of that meeting was that an XML configuration file would be generated with all of the configuration variables in it. This XML file will then be downloaded and applied by the Zoiper client based on a configuration server value being set to our Uniteme cluster. The contents of this XML file have already been discussed and there is a sample XML file already available for review.	Enhancement	Zoiper
UC-4440	Improve Agent Productivity Reports with additional column showing available time	A customer would like to have available an additional column on Agent Productivity reports (Agent Group Productivity and Agent Productivity by Groups) called "Available" and fill it with data that is "logged in time - released time".	Enhancement	Reachme reports
UC-4450	Implement proxy congestion protection - Config Work	Add new sipxproxy configuration options for the Administrator to be able to control which congestion policy is used and what the SIP Proxy Message Queue size will be. The administrator will be able to select a congestion policy: SIPX_PROXY_CONGESTION_POLICY : with default value : "SERVICE_UNAVAILABLE". Another possible value is "IGNORE" Also, add another option for the sipxproxy queue size (this is not a new setting but was not extended to web interface): SIPX_PROXY_QUEUE_SIZE:default 1024 Options to be placed under System -> SIP Proxy -> Advanced in the area of Reject calls on filled queue options.	Enhancement	sipxconfig sipxproxy
UC-4451	Add Retry-After in 503 proxy response - config work	An administrator would like to be able to allow the SIP Proxy to insert a retry-after header in replies when the SIP Proxy service becomes overloaded. This is config side work for UC-4435. The new setting is: SIPX_PROXY_RETRY_AFTER: default to 60. The new setting is located in System -> SIP Proxy -> Advanced. The purpose of this new option is to give the Administrator another method to help avoid proxy congestion. UA's that understand RFC 5390 (https://tools.ietf.org/html/rfc5390) will back off and retry after the set amount of time.	Enhancement	sipxconfig sipxproxy
UC-4452	SipDiagram Builder	Implement a sip diagram builder in Uniteme. This tool is for evaluation in the 17.08 release and not directly accessible from Admin GUI. Requirements: 1. Diagram tool shall work with elasticsearch as storage or with local sipXproxy.log 2. After installation all required scripts will be setup in web-server /var/www/html/siplogview folder 3. Generated diagrams should be in /var/www/html/siplogview/tempdata 4. Tempdata folder should be cleaned by cron task. All files which older than 1 day can be removed. 5. For building diagram "http://HOST/siplogviewer/logview.php?call_id=CALLID&es=ELASTICHOST should be called. HOST is uniteme host, CALLID - call id for required call, ELASTICHOST - url of elasticsearch as storage. If es=ELASTICHOST will not be provided, local sipXproxy will be used as call data source. 6. Ensure logging for SIP Proxy is set to INFO.	Enhancement	Diagnostics
UC-4463	Negative values in "Unanswered calls" column of Agent Answer Performance by Group report	Fixed an issue with the Agent Answer Performance by Group report where it sometimes showed negative values in "Unanswered calls" column. The issue was due to ring data inconsisitency in the call segment facts records. "Offered calls" number is computed using "value.f_ring" flag and it can be sometimes 0 even if there is valid "value.first_ring_ts" timestamp. The fix will use both values - "value.f_ring" and "value.first_ring_ts" - to check if call was offered to an agent.	Fix	Reachme reports
UC-4472	Proxy Segfault caused by malformed request URI	Fixed a problem caused by malformed SUBSCRIBE message that caused sipxproxy to crash. Core file analysis shows that reason for the crash was first character of a SUBSCRIBE message was a space. This was being caused by an ALG at a remote site adding a space into SUBSCRIBE messages. This remote site had a cable modem where the ALG could not be disabled. The fix will ignore these malformed messages.	Fix	sipxproxy
UC-4481	Greeting REST API which return filenames of different greetings.	An administrator would like a new rest API call which returns the filename for the standard, outofoffice and extendedabs greeting types.	Enhancement	REST
UC-4498		JasperServer configuration failed during system upgrade from 14.04	Fix	Reachme
UC-4516	A second voicemail left on queue defined with recipe SEND TO VM, will not alert an idle agent anymore	VoiceMail left in queue will not alert idle agents	Fix	Reachme
UW-297	Voicemail download does not work in Chrome with Unite Lite	Fixed an issue when trying to download a VM when logged in Unite Lite via Chrome on any OS. The download simply would not work. Valid only for Chrome and Unite Lite.	Fix	UniteWeb
UW-380	Upload MoH files in Unite Web	Fixed an issue with an error in the browser when up MoH files from Firefox and Chrome. Also addressed the issue of uploading either WAV or MP3. Both are allowed by MoH service.	Enhancement	UniteWeb
UW-389	Grey out or remove edit features when user group speed dials are selected	Fixed an issue where if a user goes to their speed dials and the "Use group speed dials" box enabled, the speed dials were not grayed out. The user might also think that he can add more speed dials and edit the ones he sees. If the user were to do that however, as soon as he would save the changes, they would not get saved and will instead refresh to show you the inherited group speed dials. While this behavior is correct, the user might not understand why he can't edit/add speed dials, so a solution might be to see the speed dials in "read-only" mode.	Fix	UniteWeb

UW-390	Sometimes during an IM conversation, the chat window does not automatically scroll to the new messages	An issue occurs when a user is having a chat session with someone which is lengthy enough you will eventually hit the bottom of the chat window with your new messages. Sometimes when new messages come from the party you are chatting with, the user will have to scroll down to see the new messages.	Fix	UniteWeb
UW-392	Display calling number and caller-id in voicemail page.	A user would like to be able to see the caller-id (name) and also the phone number that left the message. This number should also be clickable for click to call.	Enhancement	UniteWeb
UW-396	Make warning about closing UniteWeb configureable	An administrator would like the user to be able to have the warning when UniteWeb is closing optional on a per-user basis. The default value should be enabled.	Enhancement	UniteWeb

Issues Sorted by Keyword

	JIRA name	RN Content	Enhancement/Fix/Known Issue	Keywords
UC-4436	Decrease default number of backups to retain	Currently the default number of backups is set to 50. If an administrator doesn't pay attention to that field they can quickly run out of space. Decrease this number to 5 by default.	Enhancement	backup sipxconfig
UC-4383	Active CDR Page broken when umlauts are used	Fixed an issue with the Active CDR page in Admin GUI where when a User Agent has umlauts in the name, sipXconfig would not display the call detail properly. The problem is on sipXconfig side. If the UA encodes special characters the problem doesn't reproduce. Polycom seems it is not encoding and SIPXCDR API such that config calls to retrieve active calls works as expected. There seems to be a problem on sipxconfig side which may not be handling them properly.	Fix	CDR sipxconfig
SIPX-639	Users invited to conference rooms use only g711u	In europe g711a-Law is mostly used so some ISPs only accept g711a but not g711u. If a non-encoding SBC is used and a user is invited to a conference room via Web Portal, than the config server produces a Invite with u-law only. Some of the bigger ISPs in Germany refuse such requests with "not acceptable here" (with a cause of no supported codec). Look for the configured codecs of media services and use them only but as workaround it would be sufficient to add a-law with lower priority.	Enhancement	Conference
UC-4452	SipDiagram Builder	Implement a sip diagram builder in Uniteme. This tool is for evaluation in the 17.08 release and not directly accessible from Admin GUI. Requirements: 1. Diagram tool shall work with elasticsearch as storage or with local sipXproxy.log 2. After installation all required scripts will be setup in web-server /var/www/html/siplogview folder 3. Generated diagrams should be in /var/www/html/siplogview/tempdata 4. Tempdata folder should be cleaned by cron task. All files which older than 1 day can be removed. 5. For building diagram "http://HOST/siplogviewer/logview.php?call_id=CALLID&es=ELASTICHOST should be called. HOST is uniteme host, CALLID - call id for required call, ELASTICHOST - url of elasticsearch as storage. If es=ELASTICHOST will not be provided, local sipXproxy will be used as call data source. 6. Ensure logging for SIP Proxy is set to INFO.	Enhancement	Diagnostics
SIPX-622	Add opensource G729 codec to Freeswitch	An administrator would like to be able to enable/disable the use of the belladonna communications open source version of G729 codec in Freeswitch. The g.729 protocol is now unencumbered and customers are free to use this. https://github.com/xadhoom/mod_bcg729 To Enable: 1. Login to Linux box as root and execute: "yum install mod_bcg729" 2. In Media Settings page select the G729 codec driver 3. Send profile to Server, Freeswitch configuration should pickup latest changes	Enhancement	g729
SIPX-573	Grandstream profile fails if advanced parameter is used	Fixed an issue generating Grandstream phone profiles if an advanced parameter was used. Steps to reproduce: 1. devices -> phones -> create new, grandstream gxp2010 2. Under custom configuration -> show advanced, add a P Value of P2910=1, apply 3. send profiles Expected: phone profile to be created Actual: phone profile fails to be created	Fix	Grandstream
SIPX-604	Upgrading sipxgrandstream doesn't delete all cfg files	Fixed an issue caused when installing sipxgrandstream 17.08 where some config files are left in the config directory.	Fix	Grandstream
SIPX-641	Grandstream GXP 2160 phone profile XML errors	Fix for 2 errors in the xml config file that is created for Grandstream GXP 2160 phones. <!--OpenVPN Client Key --> <#P9904></#P9904> <!--Prefix for dialing password 0 --> <26049></26049> This should be : <!--OpenVPN Client Key --> <P9904></P9904> <!--Prefix for dialing password 0 --> <P26049></P26049>	Fix	Grandstream
SIPX-72	Add support for Grandstream GXP 2130, 2140 and 2160	Enhancement to add support for Grandstream GXP 2130, 2140 and 2160 phones.	Enhancement	Grandstream
SIPX-620	Jitsi preferred transport	A fix to correct parameter name in Jitsi provisioning plugin. PREFFERED_TRANSPORT -> PREFERRED_TRANSPORT	Fix	Jitsi

UC-4386	Idle agent is not assigned to available vqueues after websocket disconnect	Fixed an issue with an edge case scenario involving an issue identified during testing for UC-4359/UC-4316: - Sending events to the client fails with an exception. - Sometime later the agent logins and tries to register with all vqueues. - There is one vqueue but the agent is notified that the aren't any.	Fix	openacd
SIPX-608	Polycom UseTelUriAsLabel improvement	An administrator would like to allow the use of the Polycom Firmware parameter for UseTelUriAsLabel. This parameter was limited to 4.1.8 template. This parameter is now available in newer Polycom Firmware >= 5.3.1.	Enhancement	Polycom
SIPX-615	Polycom Firmware 5.x.x	An administrator would like to be able to specify Polycom Firmware up to version 5.5.2.	Enhancement	Polycom
SIPX-617	Make phone line label configurable	An administrator would like to control a phone or phone group to either have the phone line label left null in the Polycom phone configuration or have the phone configuration populated with the line number. UC-3903 implemented a change for Polycom phone line labels in reaction to Polycom changing the default behavior of their firmware. In 4.x and earlier firmware if the line label was blank, the line extension was used. In 5.x and later if the line label was blank the name associated with the line was used. Our change for UC-3903 looks at the database settings for a phone and if that is null or empty we put the line number into the phone configuration to defeat Polycom's change. This should continue to be the default behavior.	Enhancement	Polycom
SIPX-618	Polycom config parameter for new Polycom phones	Fix for some parameters in the phone provisioning that are model based. Example: Background of VVX 500 phone has specific parameter and for VVX 501 its the same. Current checks ignore the 501's	Fix	Polycom
SIPX-625	Polycom Call Waiting	Add parameter to configure call.callWaiting.enable and call.callWaiting.ring for Polycom phones	Enhancement	Polycom
SIPX-626	Polycom RealPresence Trio POE configuration	Add parameter to configure POE for LAN out port and USB charging on Polycom RealPresence Trio poe.pse.enabled usb.charging.enabled poe.pse.class	Enhancement	Polycom
SIPX-627	Polycom RealPresence Trio Firmware	The Polycom RealPresence Trio uses a special firmware revision. This is an Enhancement to add a seperate version tree for Trio to seperate Trio Firmware from other Polycom phone firmware. Example: FW 5.4.0 for VVX/SoundPoint/SoundStation/... FW 5.4.0_Trio for RealPresence Trio Trio starts with Version Number 5.4.0. The latest current release (Mar 31, 2017) is 5.4.6	Enhancement	Polycom
SIPX-640	Polycom Device Base profile	Add a new config parameter for Polycom devices device.baseProfile Values are: Generic, Lync Why this is necessary: Normaly the default value is "null", so Generic is choosen. If you buy a phone which is marked as "Lync", this is automatically choosen. We need to ensure that "Generic" is configured to get those phones working with SipXcom/UniteMe Parameter available in Firmware 5.3.x and later.	Enhancement	Polycom
UC-4398	VVX 1500 profile voice codec typo	Fixed an issue with the Polycom VVX 1500 configuration plugin. There was a typo in the voice codec settings.	Fix	Polycom
UC-4421	Default behavior for blind transfer on Polycom FW > 5.3.0	Due to changes in Polycom firmware, the old functionality for Blind transfer feature performed by hitting Transfer and then the Blind softkey is now gone. Users must now press and hold the Transfer softkey button, push down, push select "Blind Transfer" and then input their destination. If "Blind Transfer" is used more often than "Consultative Transfer" this can be set as the default option for transfer type in Preferences > Additional preferences > Default Transfer Type page in the Polycom configuration page. Also, the procedure used in the pre 5.3 versions to create a softkey with a macro(efk) to do one-button blind transfers seems to no longer work.	Enhancement	Polycom
SIPX-613	Settings evaluation with regular expression	A developer would like to be able to utilize regular expressions in provisioning plugins. This enhancement will make regular expressions available for all settings evaluation and support the old writing with || Current Settings Evaluation utilizes compare and supports optional settings separated by || Some Provisioning Plugins are already using Regular Expressions to do this.	Enhancement	Polycom sipXconfig
UC-3199	Block transfer to queue when 0 eligible agents	This feature is intended to allow a Reach administrator to configure a queue so that it will NOT accept transfer/conference calls to enter it if there are 0 eligible agents for the defined skill combination. Config Side work: add a field that is a checkbox. Label it "Block Transfer with 0 Eligible Agents" this new field is a checkbox that is not checked by default Reach Side work: When an agent is in process of transferring or conferencing via Queue, they can currently select skills via check boxes and the there is an eligible agent count provided as this active occurs. For this new feature, we should disable the transfer and conference buttons when the eligible agent count = 0. This should ONLY be done for a queue that has the "Block Transfer with 0 Eligible Agents" checkbox selected. If the Block Transfer with 0 Eligible Agents check box is not checked, there should be no call processing changes.	Enhancement	Reachme
UC-3829	Reachme On Hold Counter	Add a counter to the Session Manager that shows how long a call has been on hold. Add this below the Wait time indicator. This counter will not be displayed if the call is not on hold. It will be displayed and starts when a call is put on hold. It will be no longer displayed when a call is brought back off of hold. If the call is placed on hold again, display the hold counter again but restart it at 0.	Enhancement	Reachme
UC-3865	Add Calling number to filters in the Reachme Call recording widget	Add the calling number as search criteria in the reach call recording widget. Currently you can't search for recordings based on a calling number. This feature request is intended to allow for that. Add the called number search under the Search for Call ID text box. It should also be a text box where the user can put in a free form phone number. As is done with the "search for call id" filter, include some dim text in the text box of 'Search for Caller Number' indicating that the field is to be used to search for call recordings that are for calls from a particular caller. Ensure that the number that is being searched/matched here is the phone number as it appears in the reach cdr (Caller ANI field) rather than some form of caller ID that perhaps includes other data.	Enhancement	Reachme

UC-3909	Agent gets redirected to "undefined" or bad logout link	Fixed an issue where if a client loses its network connection to the Reachme server, then in about 17 seconds (it will take longer if you have widgets) the dashboard will prompt "Reconnecting" and stay like that. When the connection comes back up, it was observed that (rarely) the dashboard will either continue showing the same "Reconnecting" message for a period of time or most often what will happen is the agent will be prompted by an "Unable to establish a connection to server" message. Once OC is clicked in this message the agent will (most of the time) be redirected to the dashboard login page or shown a blank page with link fqdn/reach/portal /undefined or sometimes fqdn/reach/portal/logout=w...something. This seems to be very consistent with redirection (the redirect issue might be an older issue, not reported by customers, and could probably be caused by non-trusted certificates?) Does not replicate consistently.	Fix	Reachme
UC-4289	Reach skill removal does not take effect if executed while ringing an agent	Fixed an issue where Reachme skill removal would not take effect if it was executed while a call was ringing an agent. If a skill removal occurs during the period of time that the call is being offered to an agent, the skill is not taken off the call. At least not permanently. Example: Call enters the system into a queue that has a recipe step to remove a skill at 10 seconds. From 5 to 17 seconds, the call is ringing an agent and the agent does not answer the call. Once the call stops ringing the agent you will see that the skill is still on the call.	Fix	Reachme
UC-4320	Re-queue mechanism in case of network interruption does not work as expected	Fixed an issue where a call should be routed to an agent, and his phone should start ringing after a netwok interruption to the Agent Portal. The agent's phone however would start ringing for less than 1 second and stop. Steps to reproduce : Prerequisites: - 2 Agents available (200, 201) ready to receive calls. 1. Customer calls in queue 2. Agent 200 answers the incoming call alert, there is 2-way voice path. 2. Agent 200 PC network connection is interrupted for aproximately 15 seconds. (Agent 201's network connection is unnafected). Call is disconnected at the end of those 15 seconds. 3. Agent 200 dashboard will display "Reconnecting" and after a few seconds "The server has terminated your session". 4. Call is offered to remaining available clients - Agent 201. Expected result : call should be routed to Agent 201, his phone should start ringing. Actual result : Agent 201's phone will start ringing for less than 1 second and stop. Agent 201'dashboard will not show any change.	Fix	Reachme
UC-4344	Improve the Reach outbound user experience	Currently, when a reach agent is placing an outbound call, they can select a hard coded option of "No Customer". This option is not a real client in the system. Specifically, it's not built by the admin as a client. In some environments, call center managers would like to block the capability of allowing an agent to place an outbound call that is not associated with a specific client. Furthermore, there are use cases where the call center management does not care about a client specification for agent outbound calls. Therefore there is a need to allow more flexibility around this client selection for reach outbound calls. In addition, it would be a simpler user experience if the agent could type in the number that they want to call, then click the send button. Once they click send, the system would call their phone, wait for an answer, then immediately call the other party. Currently, they select a client from the drop down and that triggers the call to their phone. They then have to answer the call before they are shown a text box that will allow them to put in the number that they want to call. This feature request includes the ability to conditionally remove the "no customer" option and allow a default client for outbound calls. These changes will be driven by the agent group that the agent belongs to. Specific changes required for this part of the jira are as follows: At the bottom of the agent group page, add 2 things: Include "No Customer" Client Option During Outbound (checkbox) Default Client During Outbound (single select drop down of clients in the system) Include "No Customer" Client Option During Outbound will default to checked. Default Client During Outbound will default to no client being selected. The admin will be able to edit these settings by checking/unchecking or by selecting a single client from the drop down. The single select client drop down box will have all of the clients that are built in the system PLUS a special entry of "No Customer". The admin will select none of the entries or one of the entries. The Agent UI will be changed such that the text box allowing a user to enter the number that they want to dial will be shown at all times if they have the outbound calling permission. The "send button will also be visible at all times but only enabled when a number is in the text box AND a client is selected in the client drop down box. Reach will abide by these new settings in the following ways. If the Include "No Customer" Client Option During Outbound is checked; No Customer option will not be in the drop down option for an agent that belongs to this agent group during the outbound call process. If checked, the "No Customer" option will be in the drop down of the agent UI. If there is a client configured in the Default Client During Outbound drop down for the agent group, the agent will NOT have to select a client. Instead, the default client will be populated in the drop down automatically and they will be able to simply place a phone number into the text box to where they place the number that they want to dial and click the send call button. The system will immediately call their phone, wait for it to be answered, then place the call to the other party. The agent will be able to override the default client by using the client drop down and selecting another option. Again, to reiterate, once a client is selected (manual or by default setting) AND a phone number is placed into the text box, the send button will be enabled. During the entire process of beginning an outbound call, the X button to end the process/call will not be displayed. This X button will act as it does today, meaning that it will only be displayed and enabled once the call to the agent phone is placed. In short, the system will now allow an agent to simply place a number in that they want to dial and click send. This would be enough to place an outbound call assuming there is a default client configured in the system.	Enhancement	Reachme
UC-4380	Improve the Reach outbound user experience - UI work	Make changes to UI for Reach outbound calling improvements as identified in ticket UC-4344. This ticket is intended to be JUST the UI changes required. Initial UI side work will use "No Customer" as the default client. This will be addressed again later once UC-4344 is addressed and the config and Reach back end pieces are in place to send the default client to the UI.	Enhancement	Reachme

UC-4385	Idle agent not being able to receive a call after websocket disconnect and call requeueing is set to false	Fixed an issue with idle Reachme agents where they were not able to receive a call after they lost their browser websocket connection and if call requeueing is set to false. Steps to reproduce: -Requeue Call on UI Failure: false -While agent is oncall with customer, a network disruption occurrs for more than 15 seconds, his websocket connection dies, since Requeue Call on UI Failure is set to false, call will remain up. -As soon as the timeout on the client side is detected, Agent is being presented with the following message " Network disruption when agent is oncall, you will be redirected to login page". -Upon redirection to login page and while the call is still up, Agent will be presented with another message if he tries to log in: "Do you want to terminate your previous session? There is a call in that session which has not been handled yet. That call will be terminated if you choose to terminate the session. " -Agent forcefully terminate his session by pressing "Yes" on the prompt, at which point call is requeued - should not happen. -Agent goes to IDLE state, a new call enters the queue, but the idle agent is not able to receive it.	Fix	Reachme
UC-4400	Sometimes not getting redirected to dashboard after clicking the Continue button	Fixed an issue with the Reachme console where the user is not redirected to the dashboard properly after clicking on the Continue button. This occurred if the user had a slow network connection. Symptoms: Instead of the dashboard the user sees: {"redirect_location":"/reach/portal/dashboard","token":"JaWmoYK4tmARZ04....} From Browser Console: Request URL:https://one.ctradu-ng.ezuce.ro/reach/portal/login/advanced Request Method:POST Status Code:200 OK Remote Address:10.5.0.210:443 Referrer Policy:no-referrer-when-downgrade Response: {"redirect_location":"/reach/portal/dashboard","token":"JaWmoYK4t... Console: Resource interpreted as Document but transferred with MIME type application/json: "https://one.ctradu-ng.ezuce.ro/reach/portal/login/advanced". I am assuming this is because I tried to Login via https but can someone please confirm? I attached relevant part of reach log.	Enhancement	Reachme
UC-4422	In session inbound not aligned properly in Firefox, default zoom	QA would like so fix a small esthetic when running Firefox with 100% zoom on two different resolution 1366x768 & 1920x1080, the In Session Inbound is misaligned in the agent widget.	Fix	Reachme
UC-4431	Call recording playback issues when Reach is not running on primary	Fixed an issue with call recording playback that caused playback to not work when Reachme was not running on a primary cluster server. Steps to reproduce: Cluster with 3 nodes, primary config on node 1 and reach running on node 3. Make a recorded call and try to listen that call from supervisor : --> it will display "audio not available" message Problem was narrow it down to a redirect to localhost. Customer found an workaround to re redirect towards to the primary configuration node FQDN	Fix	Reachme
UC-4432	Correct action-on-timeout feature in reach_ouc/src/reach_vqueue.erl	Fixed an issue with the action-on-timeout feature as introduced by UC-4432. As mentioned in UC-4385, there is a potential problem with scheduling an action upon timeout set in the process' init function: If an integer time-out value is provided, a time-out occurs unless an event or a message is received within Timeout milliseconds. There are two issues: Timeout = int()>0 | infinity (not allowed to be 0), and the API doesn't guarantee that the timeout has to occur. For the record, this problematic change has been introduced with fix UC-2840 in: https://github.com/ezuce/reach-app/commit/2c630609e1b62ce4bad5cb99737a8f82b09824bd A similar change has been added to apps/reach_ouc/src/reach_vqueue.erl which will also need to be corrected in case this fix works as expected. Fix for UC-4385 has been verified and this issue is to track a similar change in apps/reach_ouc/src/reach_vqueue.erl.	Fix	Reachme
UC-4498		JasperServer configuration failed during system upgrade from 14.04	Fix	Reachme
UC-4516	A second voicemail left on queue defined with recipe SEND TO VM, will not alert an idle agent anymore	VoiceMail left in queue will not alert idle agents	Fix	Reachme
UC-3916	Sort Reachme CDRs by date and time by default	Fixed an issue with CDR Reports so that they get sorted by default by date and time as would be expected.	Fix	Reachme reports
UC-4185	Only part of the table header visible when scrolling down in CDR report, table header overlaps with input controls section when scrolling right	Fixed an issue with CDR reports where if the report has enough entries that when a user scrolled through it, the row with column names is sticky, it scrolls along when you scroll down, however, only part of it is visible. Not a huge issue, just looked odd.	Fix	Reachme reports
UC-4197	CDR Brief, CDR compact vs CDR inconsistencies and typos	Fixed some issues with inconsistencies between CDR and CDR Brief/Compact reports. 1. Column named Originating Caller ID vs Caller ID in CDR brief report 2. Time stamps having both Date and Time in the CDR brief report 3. "Time in Queue/Preca" instead of Time in Queue/Precall in the CDR brief report 4. There is also a "Totall" with double L there in CDR Brief 5. CDR compact has columns Endpoint ANI/Caller ID instead of Endpoint ANI/ Originating Caller ID and Date/Time instead of Date/Offered to Reach 6. It's the only report which is missing a capital letter :D "CDR compact".	Fix	Reachme reports
UC-4203	All "detail" reports can't handle From date bigger than To date	Fixed an issue caused when the user entered a later From date for a report than the To date. Only the "detail" reports (outbound,line,client,queue) will error out when you accidentally want to generate them with incorrect dates. Incorrect means start date is after the end date. Other reports won't display errors and will handle this mistake fine.	Fix	Reachme reports
UC-4332	"This Month" queue statistics incorrectly computed	Fixed an issue with Reachme statistics where they showed that "This Month" got populated with statistics based on calls that a user just placed on/since Tuesdays whereas it would be expected that statistics should show the number of calls placed from the beginning of the month. 1.On Monday placed 10 calls to populate Queue statistics. 2. Verified on Tuesday - the number of calls placed on Monday for "This Month" statistics - 10 calls. 3. Placed 10 more calls on Tuesday and verified "This Month " statistics again: Issue : statistics showed that "This Month" got populated with statistics based on calls that i've just placed on Tuesday: 10 calls. Expected: statistics to show the number of calls placed from the beginning of the month.	Fix	Reachme reports

UC-4406	Allow Reachme reports using Interval Data Source to hide intervals with no activity	A Reachme user would like to be able to hide intervals with no activity in the reports that utilize Interval Data Source. Changes need to be made in these reports: - Queue Traffic Detail - Client Traffic Detail - Line Traffic Detail - Voicemail Detail - Outbound Detail by Client - perhaps in Agent Availability report ?? The solution will be based on checkbox input control allowing to hide intervals with no activity	Enhancement	Reachme reports
UC-4440	Improve Agent Productivity Reports with additional column showing available time	A customer would like to have available an additional column on Agent Productivity reports (Agent Group Productivity and Agent Productivity by Groups) called "Available" and fill it with data that is "logged in time - released time".	Enhancement	Reachme reports
UC-4463	Negative values in "Unanswered calls" column of Agent Answer Performance by Group report	Fixed an issue with the Agent Answer Performance by Group report where it sometimes showed negative values in "Unanswered calls" column. The issue was due to ring data inconsisitency in the call segment facts records. "Offered calls" number is computed using "value.f_ring" flag and it can be sometimes 0 even if there is valid "value.first_ring_ts" timestamp. The fix will use both values - "value.f_ring" and "value.first_ring_ts" - to check if call was offered to an agent.	Fix	Reachme reports
UC-4424	Disable ouc_agent_report_job module in production code	A developer would like to disable some code that aggregated data from rstats to agent_report collection. This code isis not used anymore but the developer would like keep the code in-tact for debugging purposes. ouc_agent_report_job erlang module aggregates data from rstats collection to agent_report collection, which is not used anymore and contains inaccurate data. We should disable this module in production version, but still keep its code in project for debugging purposes. More details in this doc: https://docs.google.com/a/ezuce.com/document/d/1S6gq9QGae8jZVVhx1hrHFwtPeDWp_R7O62InEv1C4cc/edit?usp=sharing	Fix	Reachme reports statistics
UC-4437	Phonebook REST improvements	An administrator would like to add entry-id endpoint to the my/phonebook rest call. This should return all entryId's so you should not be forced to get them from psql queries /my/phonebook/entry/{entryId}	Enhancement	REST
UC-4481	Greeting REST API which return filenames of different greetings.	An administrator would like a new rest API call which returns the filename for the standard, outofoffice and extendedabs greeting types.	Enhancement	REST
UC-4361	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to configure the port that SIP Redirector binds to for Call Pickup. This is registrar part: Get value of SIP_REDIRECT.100-PICKUP.BIND_PORT and use it during initialisation of SipRedirectorPickUp. If no value present in config file use current PORT_DEFAULT value and system will select port on it's own.	Enhancement	SIP
UC-4360	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to change the port that SIP Redirector is bound to in sipXconfig. This is config part of the work for UC-4361: Extend SIP Registrar configuration for pickup redirector plugin bind port with port 5085 as the default. In registrar config file (/etc/sipxpbx/registrar-config) it should look: SIP_REDIRECT.100-PICKUP.BIND_PORT : 5085	Enhancement	SIP sipxconfig
SIPX-628	~~sipXprovision user has to much permission by default	The sipXprovision user which is used for auto provisioning has to much permission. To reduce load to MWI in Polycom provisioning there is an additional check if a user has voicemail permission. If they don't have voicemail permission, the phone isn't provisioned to subscribe to voicemail. This was added for Yealink as well. As the autoprovisioning feature for Yealink was added a developer noticed that the voicemail settings are provisioned. After some research it was determined that the provisioning special user has all default dialing permissions which isn't necessary. All dial permissions for the provisioning user are now disabled.	Fix	sipxconfig
SIPX-638	sipxcom 17.04 social links not in footer	The social links are not displayed in the footer within any tab beneath System -> Servers, or Diagnostics -> CDRs	Fix	sipxconfig
UC-4397	Group firmware gets reset after upgrade	Fixed an issue with Group Firmware being reset to default when upgrading a system to 16.12. After the upgrade, in the Device > Phone Groups > any group > any phone, the Group Firmware version was reset to default.	Fix	sipXconfig
UC-4450	Implement proxy congestion protection - Config Work	Add new sipxproxy configuration options for the Administrator to be able to control which congestion policy is used and what the SIP Proxy Message Queue size will be. The administrator will be able to select a congestion policy: SIPX_PROXY_CONGESTION_POLICY : with default value : “SERVICE_UNAVAILABLE”. Another possible value is “IGNORE” Also, add another option for the sipxproxy queue size (this is not a new setting but was not extended to web interface): SIPX_PROXY_QUEUE_SIZE:default 1024 Options to be placed under System -> SIP Proxy -> Advanced in the area of Reject calls on filled queue options.	Enhancement	sipxconfig sipxproxy
UC-4451	Add Retry-After in 503 proxy response - config work	An administrator would like to be able to allow the SIP Proxy to insert a retry-after header in replies when the SIP Proxy service becomes overloaded. This is config side work for UC-4435. The new setting is: SIPX_PROXY_RETRY_AFTER: default to 60. The new setting is located in System -> SIP Proxy -> Advanced. The purpose of this new option is to give the Administrator another method to help avoid proxy congestion. UA's that understand RFC 5390 (https://tools.ietf.org/html/rfc5390) will back off and retry after the set amount of time.	Enhancement	sipxconfig sipxproxy
UC-4435	Add Retry-After in 503 proxy response	An Administrator would like to be able to have UA's sending requests to proxy be able to back off if the proxy is overloaded. As determined by RFC 3261 - https://tools.ietf.org/html/rfc3261#page-176 and RFC 5390 - https://tools.ietf.org/html/rfc5390, in the 503 reply, we have added an optional Retry-After header such that phones that support this RFC will know when to retry. We currently don't do that, this is how the message looks at the moment: OUTGOING.. ...nSIP/2.0 503 Queue Size Is Too High From: sipxtest <sip:sipxtest@10.2.0.174>;tag=23289SIPpTag001683 To: 1238 <sip:1238@cluster.ezuce.ro>;tag=fzD_qG\r\n Call-Id: 1683-23289@10.2.0.174\r\n Cseq: 1 INVITE\r\nVia: SIP/2.0/TCP 10.2.0.174:8100;branch=z9hG4bK-23289-1683-1\r\n Date: Wed, 03 May 2017 13:39:44 GMT\r\n Content-Length: 0\r\n\r\n --------------------END	Enhancement	sipxproxy

UC-4472	Proxy Segfault caused by malformed request URI	Fixed a problem caused by malformed SUBSCRIBE message that caused sipxproxy to crash. Core file analysis shows that reason for the crash was first character of a SUBSCRIBE message was a space. This was being caused by an ALG at a remote site adding a space into SUBSCRIBE messages. This remote site had a cable modem where the ALG could not be disabled. The fix will ignore these malformed messages.	Fix	sipxproxy
UW-297	Voicemail download does not work in Chrome with Unite Lite	Fixed an issue when trying to download a VM when logged in Unite Lite via Chrome on any OS. The download simply would not work. Valid only for Chrome and Unite Lite.	Fix	UniteWeb
UW-380	Upload MoH files in Unite Web	Fixed an issue with an error in the browser when up MoH files from Firefox and Chrome. Also addressed the issue of uploading either WAV or MP3. Both are allowed by MoH service.	Enhancement	UniteWeb
UW-389	Grey out or remove edit features when user group speed dials are selected	Fixed an issue where if a user goes to their speed dials and the "Use group speed dials" box enabled, the speed dials were not grayed out. The user might also think that he can add more speed dials and edit the ones he sees. If the user were to do that however, as soon as he would save the changes, they would not get saved and will instead refresh to show you the inherited group speed dials. While this behavior is correct, the user might not understand why he can't edit/add speed dials, so a solution might be to see the speed dials in "read-only" mode.	Fix	UniteWeb
UW-390	Sometimes during an IM conversation, the chat window does not automatically scroll to the new messages	An issue occurs when a user is having a chat session with someone which is lengthy enough you will eventually hit the bottom of the chat window with your new messages. Sometimes when new messages come from the party you are chatting with, the user will have to scroll down to see the new messages.	Fix	UniteWeb
UW-392	Display calling number and caller-id in voicemail page.	A user would like to be able to see the caller-id (name) and also the phone number that left the message. This number should also be clickable for click to call.	Enhancement	UniteWeb
UW-396	Make warning about closing UniteWeb configureable	An administrator would like the user to be able to have the warning when UniteWeb is closing optional on a per-user basis. The default value should be enabled.	Enhancement	UniteWeb
UC-3669	Feature request - Allow greetings to be uploaded from user portal	A users would like to have the ability to upload pre-recorded greetings in WAV or MP3 via the unite web interface.	Enhancement	UniteWeb voicemail
UC-2575	Extend gridfs-voicemail-cli.jar script to allow copying a prompt into a user's mailbox prompt	An administrator would like to be able to copy a pre-recorded greeting into a user's mailbox. The existing script allows moving files but not copying a new prompt into the user folder.	Enhancement	voicemail
SIPX-159	Yealink auto-provisioning	An administrator would like Yealink phones to auto-provision in the same manner as Polycom phones.	Enhancement	Yealink
SIPX-611	Yealink Power Saving	An administrator would like to be able to configure the power saving settings for Yealink phones. This should be available for Firmware 8.x and later.	Enhancement	Yealink
SIPX-612	Yealink CDP Provision	An administrator would like to be able to configure CDP (Cisco Discovery Protocol) parameters in the Yealink Phone Configuration Plugin static.network.cdp.enable (default 0) static.network.cdp.packet_interval (default 60)	Enhancement	Yealink
SIPX-616	Yealink CP-Source Parameter	Enhancement to allow a Yealink phone to use the Contact Header to display the calling party source. In Yealink 8x (x.80.250.x) the parameter cp_source has been improved. Now phones can use the Contact Header to display the calling party source. This is useful if you want to show the callee number after pickup or from hunt groups.	Enhancement	Yealink
SIPX-624	Yealink Resource List Subscription	Yealink plugin should check if a phone line has BLFs before setting the resource list subscription URI	Enhancement	Yealink
SIPX-646	Yealink BLF 7x vs 8x	Fix difference between 7x and 8x for BLF configuration	Fix	Yealink
SIPX-648	Yealink Play Local DTMF Tone	Enhancement to add a parameter to enable/disable local playback of DTMF tones for Yealink phones. Parameter is: features.play_local_dtmf_tone_enable Currently there is parameter with this description. This is not correct. From current property file: features.AUDIO.features.partition_tone.label=Play Local DTMF Tone	Enhancement	Yealink
UC-4439	Add config support for Zoiper soft phone	A customer would like to introduce a soft phone client into their environment that is usable on a myriad of different operating systems. It has been determined that the most appropriate softphone for their purposes is Zoiper since it will work on linux, MAC, Windows, Android, etc. The customer worked with Zoiper to identify changes that will be required on their side and eZuce met with Zoiper to better understand how to best implement a provisioning model for their softphones. The result of that meeting was that an XML configuration file would be generated with all of the configuration variables in it. This XML file will then be downloaded and applied by the Zoiper client based on a configuration server value being set to our Uniteme cluster. The contents of this XML file have already been discussed and there is a sample XML file already available for review.	Enhancement	Zoiper