sipXcom 14.10

sipXcom 14.10 Release Notes

Overview

This section describes the sipXcom 14.10 software release, including:

- Software Enhancements
- Application, Hardware, and Documentation Requirements
- Installation & Upgrade Instructions
- Two attachments (PDF files) containing Fixes for internally-reported issues and for customer-reported issues

Software Release History

Starting with the April 2014 release, sipXcom will have a regular release cycle for new features and a new versioning schema which incorporates the Year and Month of release.

- December release for 2014 is 14.10 (late due to 14.04.3 critical fixes)
- April release for 2015 is 15.04

Release Level History

- 14.10 - December 25, 2014
- 14.10.1 - March 7, 2015

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)

- **Operating System:**

- **Devices:**
    - **Phones**
        - **Polycom VVX Devices** with firmware 4.1.7 are recommended for new installations
        - Polycom SoundPoint Devices with firmware 4.0.7 are also recommended.
    - **Gateways**
        - **AudioCodes Gateways** are recommended for PSTN connectivity
    - **SBCs**
        - **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 sipXcom - "Use built-in SIP Trunk SBC" found in Gateway Details for use with Trunking or Remote Worker solutions works out to about 50 concurrent sessions. sipXbridge does also not work with all ITSPs.

Documentation

Installation and Upgrade Notes

- **Installation note**

After sipXcom 14.10 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 sipXcom Admin GUI, click *System>Database*.
2. Click the *Add query metadata* button.
3. To verify that the MongoDB replica contains the unique read tags, run in the command line:

mongo

rs.config();

- **Special MongoDB note**

```
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
```

Installing from ISO image

Download sipXcom ISO

- The ISO files are available here: http://download.sipxcom.org/pub/sipXcom/ISO/
- This ISO file name ends in **x86_64.iso**

Install sipXcom

- Boot from the DVD created with the sipXcom ISO image.
- Press **Enter** at the boot screen below to begin the openUC 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 an 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 openUC.

Installing from Repository

sipXcom can be installed using the below 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 sipXcom repo:

```
yum install wget
```

cd /etc/yum.repos.d

wget http://download.sipxcom.org/pub/sipXcom/14.10/sipXcom-14.10.0-centos.repo

```
yum clean all
```

```
yum install sipXcom

sipXcom-setup
```

Upgrade from Previous Versions

Upgrade from sipXcom 4.6

Upgrade from sipXcom 4.6 is based on the next procedure:

- Edit /etc/yum.repos.d/sipXcom.repo with your favorite text editor pointing to the correct download location of the new 14.04 version:

```
[sipXcom]
............
baseurl=http://download.sipxcom.org/pub/sipXcom/14.10/CentOS_$releasever/$basearch
............
```

- Perform a backup of the entire system
- Download backup from the server
- SSH to the server and login as root (if you don't have root access to the server, login with your non-root account and enter sudo -s)
- Run the command yum clean all. When that completes, run the command yum update. As long as your sipXcom instance has access to the Internet, your system should download and upgrade all necessary updates.

Special Note for sipXcom 4.6 versions older than Update 9

The **pymongo** package in our sipXcom repository is outdated and should be replaced with the differently named but newer version **python-pymongo** pack age from the EPEL repository. This allows pymongo to upgrade along with other OS packages from the upstream provider (EPEL) rather than sipXcom repository.

Not all Updates previous to Update 8 have **pymongo** installed. If upon running yum update you encounter a message similar to:

```
Error: Package: pymongo-2.4.2-1.el6.x86_64 (@openuc)
Requires: python-bson = 2.4.2-1.el6
Removing: python-bson-2.4.2-1.el6.x86_64 (@openuc)
python-bson = 2.4.2-1.el6
Updated By: python-bson-2.5.2-3.el6.x86_64 (epel)
python-bson = 2.5.2-3.el6
Available: python-bson-2.1.1-1.el6.x86_64 (epel)
python-bson = 2.1.1-1.el6
You could try using --skip-broken to work around the problem
```

🚫 **DO NOT run yum update --skip-broken to proceed**

As mentioned, the **pymongo** package in our openUC repository is outdated and should be replaced with the differently named but newer version **python-pymongo** package from the EPEL repository. This allows pymongo to upgrade along with other OS packages from the upstream provider (EPEL) rather than eZuce openUC repository.To do so, run the following commands:

Remove Outdated pymongo package

```
rpm -e --nodeps pymongo
```

Install newer python-pymongo package

```
yum install python-pymongo
```

Clear yum cache, update, and reboot

```
yum clean all
yum update -y
reboot
```

- Run service sipXcom status to check if all services started correctly after upgrade
- Login to Admin GUI, click on System -> Servers, Select all servers and click 'Send Profiles' button

Note:

When upgrading sipXcom from 4.6 Update 11 or 14.04.3 to 14.10 follow these steps to ensure the SEC service is correctly running:

- 1. Upgrade from 4.6 Update 11 or 14.04.3 to 14.10.
- 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.

Issues

Enhancements:

JIRA ID	JIRA name	RN Content
UC-1565	sipx-servtest	The script sipx-servtest was removed in version 4.6. This script has been invaluable at sites where remote monitoring and notification are needed. Recommend restoring this script to production code so it can be maintained. This was re-added with the re-add of sipXtools.
UC-2904	Adding personal language option to VM or conference menus	Enhancement to add a personal language option for VoiceMail and Conference: - VoiceMail the new option will be: User / Unified Message / Language. This will allow the user to set it's personal voicemail prompts language. - Conference the new option will be: Conference / Configuration / Language. This option will specify the language used by the conference prompts menu.
UC-2950	Changing user voicemail greeting is not reflected in sipxconfig-audit.log or sipxconfig.log.	Enhancement to have any altered configuration item within the webui annotated at least with a timestamp and username in sipxconfig-audit.log is handled in new Audit feature.
UC-2959	REST: Personal Attendant languages	Enhancement to have a new REST call for the personal attendant settings to contain a list of all server-defined languages. Previously the JSON REST call for personal attendant settings produced the key "language" with the value "null" for default. Request URL: http://openuc.ezuce.com/sipxconfig/rest/my/voicemail/attendant Accept: application/json
UC-298	Voicemail to email template should allow custom url that can direct to a user mailbox	The voicemail to email template now allows the administrator to include the user's inbox extension as part of the URL. For more information, see http://wiki.sipfoundry.org/display/sipXcom/Voicemail-Email+Custom+Notifications

UC-3008	SAA - create SIP application and integration with config	Administrators can now enable the Shared Appearance Agent (SAA) service on one node. This service requires at least one network queue in the cluster. To enable the SAA services follow these steps: 1. From the main menu, click System>Servers. 2. Under Servers and Roles, click Telephony services. 3. Select the check box corresponding to the Shared Appearance Agent service. 4. If Network Queue and Network Queue DB weren't previously enabled, this will be indicated as a dependency with a red highlight on the service. Enable them by selecting their corresponding check boxes.
UW-101	Group 'My Profile' fields in a more logical way	Enhancement to group My Profile fields in a more logical fashion.
UW-124	Groups should remain as they were left	Enhancement to have user groups in contacts remain open if left open and closed if left closed upon navigating away from Contacts.
UW-131	Unable to search a contact by phone number	Enhancement to allow a user to search for a user based on their extension. In Unite Search bar enter an extension now to lookup a user.
UW-135	Missing Personal Attendant settings	Enhancement to add the following missing functionality from the old user portal: 1. Override language - Check this box to override the default Auto-Attendant language chosen by the administrator. 2. Change language - The language used by the Auto-Attendant if the Override field is selected above. 3. Operator - Extension or SIP URI the caller is transferred to after pressing 0. 4. Play default voicemail option - If enabled default prompt 'When you are finished, press 1 for more options. To reach the operator. dial 0 at any time.' is played.
UW-143	Roster does not get dynamically refreshed after an user add/remove	Enhancement to allow roster to be dynamically refreshed in Unite and User Portal.
UW-148	Ability to move speed-dial order from UW like in old portal	Enhancement to allow the user to move speed dials up and down in GUI.
UW-151	Move bottom bar functionality to top bar	Enhancement to move the bottom bar (menu, search, dialpad) to the top bar (profile bar) to be more inline with other mobile and desktop applications.
UW-158	Mobile Support for Unite Web	Enhancement to allow use of Unite Web on Mobile Devices. Tested on Android 4.4, Chrome & Native, Android Tablet, Chrome & Native, iOS 8 Smartphone: Safari, Tablet: Chrome & Safari
UW-99	Add Editable Subject for a Conference Record	Enhancement to include a subject to a conference record; With Old User Portal the user is able to edit the Conference Subject (this might give the option to add some info about the conference / participants - currently the user does not receive any info about the conference,this way, at least, he could add them) With Unite / Unite Lite this is not possible.
XX-10035	XX-10035 Voicemail to email template should allow custom url that can direct to a user mailbox	Enhancement to add PortalURL in the custom file,and its value was then automatically used to compose the links from VM to email. See this for how-to : http://wiki.sipfoundry.org/display/sipXcom/Voicemail-Email+Custom+Notifications
XX-10097	sipXevent (WebSocket) should listen for voicemail events	Enhancement so that voicemail events are now generated by the IVR component, in order to avoid system pooling that clients have to perform in order to get real-time information . Now, the messages are sent through Wbsockets using the JSON standard.
XX-10099	sipXevent (WebSocket) should listen for conference events	Enhancement to allow sipXevent (WebSocket) to listen for conference events.

X X -1 0 7 79	Allow sipXcallqueue calls to agents to follow call forwarding	Enhancement for a new feature for CallQueue that allows a call rining an agent to follow the user's call forwarding settings. When enabled, administrators can enable the feature by selecting the new Follow Call Forwarding option. 1. Click System>Servers in the main menu. 2. Under the Servers and Roles tab click Telephony Services. 3. Check the check box corresponding to the Call Queue (beta) feature and click Apply. 4. Click Features>Call Queue. Under the Call Queue tab, click Agents. 5. Click Add Agent. 6. Select the Follow Call Forwarding and other required details and click Apply.
X X -1 0 9 14	Port merge-logs and sipx-trace to 14.10	Enhancement to bring back Merge-logs, sipx-trace and associated tools to 14.10.
X X -1 1 0 83	System Alarms for DNS Lookup Problems	Enhancement to create system alarms for DNS Lookup Problems. Alarm information should contain: Service Trying to do Lookup Date / Time Type of failure (DNS server unavailable, no host / srv found, etc.) What was trying to be looked up.
X X -1 1 0 84	Be able to customize DNS Views from UI	Enhancement to make DNS more configurable. DNS Views were added in update 8, but you could not customize the values w/o java programming .
X X -1 1 1 15	Update Freeswitch to current stable release	The Freeswitch server has been upgraded to version 1.4.6. Previously, the supported version was 1.0.7 plus some additional patches
X X -1 1 1 16	Upgrade openfire and hazelcast	The Openfire server has been upgraded to version 3.9.3. Previously, the supported version was 1.2.0. Also, Hazelcast was upgraded to version 3.1.5. Previously, the supported version was 2.5.0.
X X -1 1 1 17	Move openfire configuration to mongodb	Enhancement to the Openfire Mongo Sync plugin, until now an eZuce proprietary feature, has been moved to the open source project sipXcom. This plugin allows Openfire to collect data directly from the underlying MongoDB database.
X X -1 1 1 18	Spring framework update	The Spring Framework has been upgraded as follows: -upgraded spring core, beans, context, AOP, expression to latest release 4.0.3, -upgrad dspring security to version 3.2, -added spring boot dependencies, -upgraded Jackson to 2.3.3, -upgraded Velocity to 1.7, -added websocket dependencies, -u pgraded Jetty for websocket support, Notes: -spring orm and jdbc remain at 3.1.4 due to major API change in version 4, -spring web and javax.servlet remains at 3.1.4 due to Jetty 5 limitation to servlet spec 2.x
X X -1 1 1 46	Upgrade Java to 1.7	Java has been updated to version 7 and servlet to version 3. The new versions allows you to create modern client web modules for sipXcom. Previously, the system used Java 6. Note that the Jetty dep 5.1.4 and servlet 2.0 have been kept for other projects that use Jetty container.

X X -1 1 1 48	Upgrade Jetty to latest (9.1.5)	The Jetty server has been upgraded to version 9.1.5. In support of this upgrade, the following changes have been performed: -upgraded jasper-compiler and the jasper-runtime to 5.5.23 -upgraded slf4j to 1.7.7 -updated sipxconfig and cmcprov to use the new Jetty version -upgraded servlet 2.x spec to servlet 3.1 -update sipxconfig-jetty according to new jetty -update sipxconfig startup script -fix precommit Note that the existing Jetty dep and servlet 2.0 have been kept for other projects that use Jetty container. Previously, the system used Jetty 5.1.4.
X X -1 1 1 66	Port Phones management API to REST	See the API Guide for more info
X X -1 1 1 71	Expose Phone Model REST API	See the API Guide for more info
X X -1 1 1 74	Expose Server, Bundle, Feature & Jobs management via REST API	See the API Guide for more info
X X -1 1 1 75	Expose REST API for Registration management	See the API Guide for more info
X X -1 1 1 76	Expose IVR Service REST API management	See the API Guide for more info
X X -1 1 1 93	Expose Park Orbits REST API	See the API Guide for more info
X X -1 1 1 99	Expose REST API for Page groups management	See the API Guide for more info
X X -1 1 2 02	Expose CDR, MyBuddy, Proxy, Registrar, SAA, REST server REST API services management	See the API Guide for more info
X X -1 1 2 04	Expose DNS and MWI management REST API	See the API Guide for more info

XX-11205	Upgrade hibernate from 3.2.7 to 3.6.10	The Hibernate FLush strategy has been upgraded from 3.2.7 to 3.6.10. This upgrade has been performed to expose postFlush event handling mechanism (org.hibernate.EmptyInterceptor).
XX-11206	Expose MoH management REST API	{{For info see the API Guide}}
XX-11218	Integrate Yealink plugin	Add Yealink plugin from work created by siplabs.ru. Adjust plugin for e911 work that may cause issues with it. There are still a number of issues with plugin identified by QA. See linked issues in Jira
XX-11221	Be able to map LDAP item to Employee ID field.	Administrators can now add information from LDAP to the existing Employee ID field, under the User>identification tab and LDAP / AD Configuration section.
XX-11222	Create 3 new user record fields	Three new fields named Custom1, Custom2 and Custom3 have been added to allow administrators to add any additional data about the user. The values are mapped to the LDAP fields and are kept in the userProfile MongoDB. Also, they can be exported and imported in the CSV format. To view the fields, follow these steps: - Click Users>Users on the main menu. Click on a user name. The new fields are under the Identification section. or -Click System>LDAP AP in the main menu. Under the Configuration section, the fields are displayed when a LDAP configuration is being set up.
XX-11228	Expose Phone Group via REST API	See the API Guide for more info
XX-11231	Expose Gateways via REST API	See the API Guide for more info
XX-11250	Admin REST Api for Schedule (General, User, Group)	See the API Guide for more info
XX-11255	DialPlan REST AP	See the API Guide for more info
XX-11265	Proxy should normalize GRUU uri locally	For higher performance, the sipXproxy now normalizes the Globally Routable User Agent URI (GRUU) requests by querying the Registration DB in the proxy. Previously, the sipXproxy evaluated forwarding rules which in turn routed all GRUU URI through the registrar.
XX-11294	DNS advisor should know about Regions	A new parameter named Region has been for the DNS Advisor added to allow administrators to validate and view the DNS for any specific region. Note that the field is not displayed if the environment does not have any regions.

X X -1 1 3 21	Under Type drop-down menu options are not sorted alphabetically	The Type drop down under the History section of the System Audit page has been improved and now displays the options in alphabetical order, to help users improve productivity. To review the changes, click Diagnosis>System Audit>History and click the arrow next to the Type drop down.
X X -1 1 3 29	Remove Encrypted Communication service	Enhancement to remove the Encrypted Communication service has become obsolete and has been removed from the user interface. Starting with 14.10, the Secure Shell (SSH) cryptographic network protocol is used to communicate between nodes.
X X -1 1 3 47	Features that should be enabled by default on Yealink phones	Enhancement to make the MWI (Meassage Wainting Indicator) and Call Waiting features are available by default on Yealink phone models.
X X -1 1 4 34	New 14.10 alarms needs to be moved in dedicated alarm log files	New alarms are now logged in the /var/log/sipxpbx/sipxbridge-alarms.log. This ensures that new alarms are displayed int the log history, snmptrapd.log or sent through email.
X X -1 1 5 11	Add support for Polycom firmware 4.1.7, 4.1.8, 5.1.1, 5.1.2, 5.1.3, 5.2.0	Add in device files, group firmware selection, device firmware selection and default firmware for auto provisioning.
X X -4 8 30	RedirectServer should not add Contact for voicemail to non-INVITE requests	Made an improvement to not add contact for voicemail to non-INVITE requests. This caused the system to generate a 501 Not implemented error message when a SIP request of the non-invite type was sent to a user that was not logged in the system. Now, the system correctly sends the 404 user not found error message. For example, sending an IM to a non-registered user would have been redirected to the IVR.
X X -8 4 73	FreeSWITCH parking	The Parking service has been reimplemented in order to be integrated with FreeSwitch. All the previous features are still available. You can now configure the Parking service on multiple nodes.
X X -9 5 35	Request to track config changes	Enhancement to allow the ability to track config changes on the system.Following info should be stored by the system & made available on a page with filtering capabilities: - Date & time - User - IP Address - Type of Process - Process - Value Before Process - Value After Process Process type can be - Login / logout - User config - Device config - Features config - System config - Diagnostics config - License config Filtering based on start & end dates, user, user group, type of process, process & IP address should be possible on this page. This page will replace the Login History page already available in the system.

Fixes

UC-1563	Proxy should discard short non-SIP messages	Fixed an issue that caused the proxy to attempt to respond to non SIP-messages. Now, these non SIP messages sent by some devices as NAT keep-alive messages are ignored. Previously, the following error and error messages were generated by the proxy: 2013-09-04T17:54:27.346293Z:3034745:INCOMING:INFO:sipx01.rdu2.redhat.com:SipClientUdp-10:7fb2c8ffe700: SipXProxy:Read SIP message: ----Local Host:18.11.100.245---- Port: 5060---- ----Remote Host:18.36.44.232---- Port: 5060---- jaK 2013-09-04T17:54:27.346551Z:3034746:SIP:WARNING:sipx01.rdu2.client.com:SipClientUdp-10:7fb2c8ffe700: SipXProxy:SipMessage::getResponseSendAddress No VIA address, using From address 2013-09-04T17:54:27.346598Z:3034747:SIP:ERR:sipx01.rdu2.client.com:SipClientUdp-10:7fb2c8ffe700:SipXProxy: SipMessage::convertProtocolStringToEnum unrecognized protocol: 2013-09-04T17:54:27.346624Z:3034748:SIP:ERR:sipx01.rdu2.client.com:SipClientUdp-10:7fb2c8ffe700:SipXProxy: SipTransaction::doFirstSend protocol not explicitly set - using UDP 2013-09-04T17:54:27.346734Z:3034749:OUTGOING:INFO:sipx01.rdu2.client.com:SipClientUdp-10:7fb2c8ffe700: SipXProxy:No send address SIP/2.0 484 Address Incomplete From: To: ;tag=VWYG_S Call-Id: Warning: 399 10.11.634.245 \"Missing one or more of To, From, or request URI\" Content-Type: message/sipfrag Server: sipXcom/4.6.0 sipXcom/sipXproxy (Linux) Content-Length: 64
UC-1743	Investigate DST updates in schedules, voicemails, cdrs, etc.	AA/IVR schedules have not updated with the DST change. Phone and server times seem to be correct.
UC-1753	MyBuddy not working for newly created groups	Fixed an issue that caused My Buddy to be non-responsive for newly created IM groups. Now, when users type "Help" in the chat window with My Buddy, it correctly replies with the message "Commands: call, find, missed, listen, pickup, conference, who, mute, unmute, disconnect, lock, unlock or any short forms"
UC-1957	Repeated add/remove of an user to the same group, requires an openfire restart to work	Fixed an issue with the Openfire system that prevented users that were repeatedly added and removed from a group in OpenUC to see other users in the user group, and also prevented other users from viewing the user. Now, the user can not just log in his IM client, but he can correctly see other users in the group and they can view the user.
UC-2007	DNS Advisor reports incorrect results	Fixed an issue that caused the DNS Advisor to generate incorrect results.
UC-2097	Send profiles required for user changes	Fixed an issue that caused a send profiles to be performed to pick up user changes. Problem setup: Created schedule Weekend from 9:00 AM to 6:00 PM, replicated OK in mongo (mind the sipx-ValidTime) <sip:4455@ezuce.ro;sipx-noroute=Voicemail?expires=30>;q=0.9;sipx-ValidTime="2364:2580:1a4:3c0" What was occuring: Changed schedule, it didn't trigger user regeneration in mongo so sipx-ValidTime in contact remained the same. Workaround is to go to call forwarding page and hit apply in case schedule changed (no need for sending profiles) <sip:4455@ezuce.ro;sipx-noroute=Voicemail?expires=30>;q=0.9;sipx-ValidTime="2634:2670:474:4b0"
UC-2306	Delay when trying to place calls after server reboot/shutdown in a partitioned region	Fixed an issue that caused the system to place calls with a 5-10 seconds delay. This occurred in a HA environment, and using a partition that has been previously restarted. Now, the delay decreased from almost 10 to maximum 2-3 seconds for the first call and no delay for the rest of the calls.
UC-2323	VMs received from other regions wont be updated in a partitioned region	Documenting a known limitation with VMs received when a region is partitioned. For a configured region voicemails are left in local database and then synced to global replica set. When retrieving voicemail for an user inside a region both local and global replica set are queried. In case region gets separated voicemails from global replica set can be read (retrieved from local node within global replica set) but they cannot be deleted / updated (since primary node within global replica set cannot be contacted). For the user experience this means he will be able to listen all voicemails but won't be able to delete / mark as read voicemails that were read from global replica set

UC-2355	500 Error returned ,with no other explanation, when configuring databases (intermitent)	Fixed an issue that was displayed while a database operation was in progress that showed a 500 Error is returned with no other explanation of what went wrong. This was happening when removing databases, clicking Add to Config, or Start buttons.
UC-2362	Calls downtime in a partitioned region after a 30 minutes server idle time	Fixed an issue where after a region was partitioned from the global system for 30 minutes calls between phones in the partitioned region would be delayed for 5 minutes. Reverted a mongo driver change to resolve the problem. There is still a delay of 10 to 13 seconds while the driver tries to contact the global database. Recommend waiting on use of Regions until Mongo and Mongo drivers are updated to Mongo 4.6.
UC-2404	Music on Hold rule present twice in mappingrules.xml	Fixed an issue where Music on Hold rule was present twice in /etc/sipxpbx/mappingrules.xml
UC-2484	Stunnel service prevents server profile push	Fixed an issue where if an Administrator enabled Encrypted Communication (System -> Servers -> Core Services -> Encrypted Communication) it prevented the successful server profile push to other nodes in the cluster. Removed Encrypted Communication service as its use is no longer required.
UC-2755	MWI service losses Mongo master resulting in apparent message queue overflow	Fixed an issue where the MWI service claimed that there was "no master found" for MongoDB, the MWI service crashed, refused connections from the proxy and never attempted to reconnect to MongoDB. The proxy would then continue to pile up SUBSCRIBE messages destined for the MWI service. sipXproxy's internal queues would eventually overflow and sipXproxy started taking longer than 1 minute to process any inbound or outbound calls. Restarting the servers resolved the issue. Resolution is for MWI service to more gracefully handle MongoDB outages by issuing a 500 Internal Server Error when there is a MongoDB issue, and try to reconnect in the event of being disconnected from MongoDB
UC-2845	proxy crashes on invite using tel:scheme	Fixed an issue with sipXproxy so that if it receives an INVITE using tel:scheme it won't crash. The fix will: 1. On receipt of a new request, the transport layer detects whether any of the R-URI, From-URI and To-URI is in tel-uri format. This now gets converted to a sip-uri assuming that the local domain is known. 2. Transport layer will add a new request property to the request with tel-uri headers. FROM-TEL-URI and TO-TEL-URI respectively. 3. When responses are received, Transaction::handleOutgoing will check if the transaction request has any of these parameters and rewrite the sip-uri to its tel-uri form.
UC-2876	After user deletion,his folder in mediaserver /data is not removed impacting new users with the same extension	Fixed an issue where if a user is successfully removed from GUI his folder from mediaserver/data/mailstore did not get deleted.
UC-2920	REST: Voicemail change subject does not persist	Fixed an issue where if you issued a PUT on: sipxconfig/rest/my/redirect/mailbox/USERID/message/VMMSGID/subject with body containing a String results in status code 200, meaning that the PUT has been successful. But when doing GET on a folder to see the changes that were made you will receive empty subjects for all vm messages that have been changed.
UC-2921	With regions, phones lose registrations for a short time intermittently	Fixed an issue with regional server where the phones would lose registration for a very short time at random times. The Polycom phone display will show the line is not registered, but the line actually just looses the registration for a very short time. It just takes the Polycom longer to refresh. Note: Recommend setting Polycom phones to TCP signaling and shorten registration interval to 300 seconds to shorten the Polycom refresh.
UC-2999	SFTP purge old backups is not executed	Fixed an issue where SFTP backups were not purged as new backups were added (according to the number of backups to be retained).
UC-3049	Dial-By-URI can bypass authentication rules	Fixed an issue where a phone with its outbound proxy pointed at sipXcom could bypass authentication rules when dialing with a SIP URI.
UC-3066	Linksys ATA: e911 /location setting not found	Fixed an issue that caused replication to fail and sipxconfig would not start if a system had Linksys ATA phones defined.

UC-317	Forwarded voicemails cannot be listen in old portal	Fix an issue where a forwarded voicemail with comment the old user portal would play only the comment and not the entire message. This was due to the content length that is not correctly calculated as sum of comment length + original voicemail length.
UC-377	Dial plan for handling blind transfers from FS incorrect	Fix to freeswitch dial plan to properly handle blind transfers coming from freeswitch. Related to XX-10900
UC-378	Wrongly raised alarm in nodes without mongo	Fixed an issue that caused a MONGO_MEMBER_DOWN alarm if a snapshot was taken from a node that does not have postgres.
UC-253	Automatically save lowercase no matter if in LDAP is with upercase letters or not	Fixed an issue caused by customers using mixed case in Active Directory for usernames. For auth purposes in openUC we should only store auth account name as lower case.
UC-300	sipX can get into an infinite keep-alive loop	Fix for sipXtackLib sends back a PONG if any length form of keep-alive packets inluding sending a PONG when it receives a PONG. This would incorrectly result to an infinite PONG-PONG hand shake. RFC 5626 specification below corrects this behavior by differentiating between a PING (crlf/crlf) and a PONG (crlf) 3.5.1. CRLF Keep-Alive Technique This approach can only be used with connection-oriented transports such as TCP or SCTP. The client periodically sends a double-CRLF (the "ping") then waits to receive a single CRLF (the "pong"). If the client does not receive a "pong" within an appropriate amount of time, it considers the flow failed. Note: Sending a CRLF over a connection-oriented transport is backwards compatible (because of requirements in Section 7.5 of [RFC3261]), but only implementations which support this specification will respond to a "ping" with a "pong".
UC-311	4.4 standard greeting migrated twice, out of office greeting not migrated	Fixed an issue when migrating from 4.4 to later versions the migrate logic uploaded standard.wav twice and out of office not at all.
UC-312	SSLv3 POODLE vulnerability mitigation	Address all instances of where SSLv3 was utilized and disable SSLv3 to mitigate POODLE vulnerability.
UC-324	When named is down, incoming trasactions are stuck in 100 trying	Fixed an issue caused by shutting down named on a node, then placing call through that proxy. The phone would get back 100 Trying from proxy then proxy cannot locate a registrar to send invite to. This would cause the transaction to hang there in Proxy queue. Proxy should not send back 100 Trying in such case but a 408 Request Timeout error so phone would know to try another proxy in HA (in 14.10 there are alarms for such failed lookups) Upon failure of DNS lookups, recurseDnsChildren should start a timer that fire immediately to mimic a transaction timeout.
UC-432	Remove port number from sample Bria provisioning URL	In the admin screen for a user account, Registrations tab contains this note : "Use http://servername:12000/cmcprov/login as a provisioning server for Bria 3.x softphone." This issue is to simply remove the reference to :1200 on this screen
UW-161	Voicemail filenames with preceding zeros are truncated	Fixed an issue with voicemails migrated from a 4.4 system that have preceding zeros in the filenames. These were getting truncated by the Unite Web application. This results in a file not found error.

UW-162	Audio bar is not refreshed when deleting a VM	Fixed an issue that was caused by deleting a message while it is being played, the next message takes the place of the deleted one, but the audio bar is still playing Steps to reproduce: Click to play a voice message. While the message is still playing, press the delete button.
UW-176	User Password Change does not allow for all characters allowed by openUC	Fix to allow user password change in Settings to allow user to utilize symbols in password. Input field should match what is allowed by openUC.
UW-26	User should be able to use Unite & Unite Lite on a mobile device (tablet, smartphone, etc.)	Fixes to minimize differences in interaction using the webchat between desktop, mobile and tablet browsers.
XX-10811	openUC in a bad state after primary node reboot	DHCP scenario: 1. openUC Primary reboots 2. /etc/resolv.conf is rewritten by DHCP 3. Mongo cannot resolve peers and cannot establish connections to other hosts in the cluster 4. Time passes and the configuration cannot be generated as Mongo is read only 5. CFEngine rerwrites /etc/resolv.conf with the openUC DNS servers 6. Mongo resolves peers and connects ok * Screenshots attached with the results of this scenario Static IP scenario 1. openUC Primary reboots 2. /etc/resolv.conf is rewritten by the network scripts with the DNS from /etc/sysconfig/network-scripts/ifcfg-eth0 (the DNS in ifcfg-eth0 is the one used to setup the box so that openUC can be installed) … * This has a lower chance of happening than the first one These scenarios happened to both Customers, QA and Dev and they were a pain to debug. Both scenarios are the result of the network interface config not being managed by openUC. The proposed fixes were: 1. Add PEERDNS=no in /etc/sysconfig/network-scripts/ifcfg-eth0, which will prevent rewriting /etc/resolv.conf by the network scripts 2. Manage DNS servers in /etc/sysconfig/network-scripts/ifcfg-eth0 For the purpose of this description, /etc/sysconfig/network-scripts/ifcfg-eth0 = network interface where openUC listens and can be different from setup to setup. Another thing that needs to be taken into consideration is that this is related to Un/Managed DNS and has to be enabled /disabled based on that I used the PEERDNS approach during this time as a workaround ad worked as expected. My recommendation would be to take this approach. -------------- Additional Clarification ------------------------------- If DNS is set to managed: Ensure in ifcfg-ethX interfaces that peerdns=no and that nm_controlled=no No DNS servers should be entered in ifcfg-ethX files All DNS servers should be in /etc/resolv.conf These settings should be verified before system services start... each time. If DNS is set to unmanaged: ifcfg-ethX interface files should not be changed end user is free to put DNS entries into ifcfg-ethX files the system should not modify /etc/resolv.conf the system should not modify /etc/named.conf the system should not modify any bind zone files in /var/named
XX-10834	DST change did not trigger a "send profiles"	Fix to help address reports that the DST change did not trigger a full rebuild of the mongo database and schedules were off by 1 hour. We need to investigate why this happened. Added some logging to code
XX-10877	User ~~id~media (freeswitch) has exposed password in snapshot	Fixed an issue hat caused the Freeswith server to expose the password of the user media in the snapshot. Now, the password is no longer displayed in the sipX_profile.xml.

X X -1 0 8 99	OpenFire restarts on nearly every configuration change	Fixed an issue that caused the Openfire server to restart each time the administrator performed changes to the configuration This issue caused the incoming calls to be rejected. Now, Openfire no longer automatically restarts when configurations are modified.
X X -1 0 9 16	When saving a phone onSave() is triggered twice	Fix to address when saving a phone either on create or on edit, the method onSave() is called twice.
X X -1 0 9 17	When saving a phone which is part of a group or groups, there will be multiple calls to onSave()	Fixed an issue that caused the system to send multiple requests to the onSave()event when administrators saved or edited a phone that was part of a group, or multiple groups. This issue prevented the Alarms->History page log any events. Now, the snmp trap is sent correctly and alarms over the email are sent accordingly.
X X -1 1 0 82	Internal error when accessing Licensing tab after Restore on a different domain	Fixed an issue that prevented the Licensing page from displaying the content and caused the system to generate an internal error. This issue occurred when the licence used by the system was incorrect. Now, a job status error is still generated, but the Licensing page is correctly displayed.
X X -1 1 1 11	Polycom SIP Servers transport settings not applied	Fix to address SIP Server transport settings not being applied to configuration files. Administrators can set transfer on a phone using three pages: 1. SIP Servers > Outbound proxy > transport 2. Line > Registration > Outbound proxy transport 3. Line >Registration > Primary/Second registration server > transport However, it is recommended to use an outbound proxy only if administrators want to redirect SIP traffic through a server other than the registration server. You can set the outbound proxy globally on the phone and particularly on each line. The value set on the line will override the value set globally. For example, for a phone with 4 lines and an outbound proxy set on the SIP server setting globally, but on line 4 there is a different outbound proxy, the first 3 lines will use the globally defined outbound proxy and the fourth line will use its own setting.
X X -1 1 1 22	Up and down arrows not working when adding new call forwarding entries	Fixed an issue with the Unite User Portal that prevented the up and down arrows next to the options on the Call Forwarding tab from being displayed. Now, the arrows are correctly displayed and allow users to add entries for the call forwarding options and or to change their order.
X X -1 1 1 25	Phone Outbound Proxy does not override Line Outbound proxy	Fixed an issue where setting the value of the Server on the Phone>SIP server page for a Polycom phone or phone group, did not override the value of address for the Primary and Secondary Registration servers on the Line>Registration page. Now, the phone value always overrides the line value.
X X -1 1 1 33	Cannot access Conferencing page after deleting a server from cluster	Fixed an issue that prevented access to the conferencing page when in an HA environment the administrator deleted from the cluster a server that had a conference bridge.
X X -1 1 1 41	Scroll bar is missing in the Voicemails window	Fixed an issue with the Unite User Portal that caused the Voicemails tab to be displayed without the scroll bar.
X X -1 1 1 70	FreeSwitch uses some CPU when idle	Fix for freeswitch performance problems. FreeSwich performance has been dramatically improved, as it now requires 1 to 2% of CPU usage when it is in idle mode and only Media Services are running. Previously, FreeSwitch required approximately 7% of CPU. The assessment was performed on a virtual machine with a 3.4 GHz processor assigned with 2 cores with x1.7GHz and 3 GB of RAM.

X X -1 1 1 82	Default serial fork expiration and user call forwarding interfere with call queue/agent settings	Fixed an issue with CallQueue so that the agent setup timers would work independent of system default serial fork expiration. By default the openUC system will have SIP Proxy -> Default Serial Fork Expiration set to "20". Also under Users -> Users -> User X -> Call forwarding value is set to 20 These settings will have an impact on how the call queue agent No Answer Delay Time and Max No Answer (count) UC-2055 , probably over other settings as well.
X X -1 1 1 90	Offline messages transmitted with every new login	Fixed an issue that caused the IM client to send users off-line messages from another user, each time the user logs in and logs out the IM client. Now, the off-line message is only displayed only ince, when the user logs in the system.
X X -1 1 1 92	Users members of multiple groups can only see the first group in their contacts list	Fixed an issue where users members of multiple groups can only see the first group in their contacts list, if the administrator previously changed the user groups. This issue occurred in a HA environment with Pidgin and Unite2 clients running on Ubuntu.
X X -1 1 1 96	Setting a custom status no longer works	Fixed an issue with the Open Fire service that prevented users from viewing custom messages set by another user in the Pidgin IM client.
X X -1 1 1 97	Changing User Group name several times in a row not visible until openfire restart	Fixed an issue with the Open Fire service that prevented the administrator from updating user group names in the IM client. Thi sissue has been fixed by implementing a mechanism for HA where only one node processes MongoDB events. Now,group name is succesfully changed in admin UI as well as in IM Client.
X X -1 1 2 13	Overwriting users table through Import ".csv" file damage mongoDB	HA - 3 nodes cluster: primary, secondary, arbiter. steps to reproduce: 1. Import "5000users.csv" file all users will be created without any errors. 2. Import for the 2'nd time the same file . This should overwrite the users table. Expected result: overwriting the users table should be done without errors. Actual result: Mongo damaged, primary server and secondary become UNAVAILABLE. ps: tested for 1 node on 14.04 and 14.10 tested for HA in 14.04 this bug is reproduced only for HA in 14.10 Downer you can find the errors from sipxconfig.log: [root@p151 ~]# tail -f /var/log/sipxpbx/sipxconfig.log "2014-06-26T08:23:52.440000Z":747:JAVA:ERR:p151.dluca.ezuce.ro:qtp2090424299-26: 00000000:UserDecorator:" failed to retrieve user profile Read operation to server s152.dluca.ezuce.ro/10.5.0.152:27017 failed on database profiles; nested except ion is com.mongodb.MongoException$Network: Read operation to server s152.dluca.e zuce.ro /10.5.0.152:27017 failed on database profiles" Thu Jul 10 14:20:35.598 [initandlisten] connection refused because too many open connections: 819 Snapshots attached for primary, secondary and arbiter. workaround: # nano /etc/security/limits.d/99-mongodb-nproc.conf mongodb - nproc 655350 # service mongod restart # service mongod status \\to find mongod <pid> # cat /proc/<pid>/limits
X X -1 1 2 25	CPU_THRESHOLD_EX CEEDED Alarm - isn't generated	Fixed an issue that prevented the system from generating the CPU_THRESHOLD_EXCEEDED when the CPU usage was higher than the one determined by the administrator. Now, the alarm is correctly generated in Alarms>History, the snmptrapd.log and it is sent by email.

X X -1 1 2 26	CPU_THRESHOLD_RE COVERED Alarm - isn't generated	Fixed an issue that prevented the system from generating the CPU_THRESHOLD_RECOVERED when the CPU usage was higher than the one determined by the administrator. Now, the alarm is correctly generated in Alarms>History, the snmptrapd.log and it is sent by email.
X X -1 1 2 30	The proxy MUST remove the user-provided P-Preferred-Identity header from any message it forwards.	The proxy now removes the user-provided P-Preferred-Identity header from any message it forwards.
X X -1 1 2 32	STUN binding requests interpreted by SIP transport as EOF reads	STUN (Session Traversal Utilities for NAT) binding requests are now treated by ports (port 5060 by default) as NOOPs. Previously they were processed as zero length reads and caused the SIP transport protocol to generate assertions if they the number of packets was higher than 20.
X X -1 1 2 38	Authentication Rules should not authenticate calls to registered users from an SBC or gateway	If an SBC handles remote worker registrations and it is also configured as a gateway with permission, calls to registered users will be authenticated. This will fail if the call is coming from the PSTN. Note: This is untested by QA in 14.10. It is disabled by default. Full delivery pending QA in 15.04.
X X -1 1 2 40	Grandstream phones are unable to disable NAT TRAVERSAL feature	Fixed an issue that prevented administrators from disabling NAT traversal features for Grandstream phones.
X X -1 1 2 52	Extra LF in dhcpd.conf file causing restart problems	Fixed an issue with the dhcpd.conf file that prevented the Dynamic Host Configuration Protocol (DHCP) server from running. This issue occurred because of an extra line feed in the file.
X X -1 1 2 61	DNS advisor - A records missing	Fixed an issue that caused the DNS Advisor to erroneously generate errors regarding missing A records, even though they were present in the default view.
X X -1 1 2 96	Remove Outbound Proxy Auto-fill from Polycom and Audiocodes Plugins	Fixed an issue that caused the system to auto fill the outbound proxy setting for Polycom and Audiocodes Plugins with the SIP domain. Now, the Outbound proxy transport field on the Line > Registration is blank by default.
X X -1 1 3 02	User cannot add comments to forwarded voicemail when mp3 used	Fixed an issue that prevented users from adding comments to a forwarded voicemail when the system was set to MP3 encoding. Now, then message is correctly forwarded with both the mp3 file and the new message added by the user.
X X -1 1 3 05	Create SSL connector in Jetty for web socket	A SSL connector and a firewall rule have been added to the Jetty HTTP server on port 8076 for web sockets. This has been implemented because we cannot proxy wss packets over httpd port 443 with the current version of apache httpd server.
X X -1 1 3 08	Changing avatar will cause a lot of log messages to be sent	The following changes have been performed to prevent the system from generating an unnecessary large number of log messages when users changed the avatar: -re-enabled avatar operations log listener, -removed unused caches, -removed Delete Avatar job, -removed MD5 verification of old and new avatar, -used VCardEventDispatcher.dispatchVCardUpdated in VcardUpdateJob to clear caches then broadcast presence.

X X -1 1 3 11	Multiple groups of users can cause problems when EN / DIS IM services	Fixed an issue where multiple IM groups of users could cause problems with enabling or disabling IM services. Steps to reproduce: - Enable for each group all IM services and MyBuddy services - Login via Pidgin with "200" and "201" users accounts (expected results: users logged successfully, they can see both "MyGroup" and "MySndGroup") - For "MyGroup" go to Users->User Groups->Instant Messaging and uncheck / disable "Enable IM account" , "Add user group as IM group" and "Add MyBuddy to the IM group" - For safety even if i'm not sure this is the correct practice, go to Pidgin and logout/login "200" and "201" users Expected results: "MyGroup" should disappear from contact list Actual results: "MyGroup" still present in contact list. If is not reproducible with first try insist many times (2 times of enable/disable IM services) Additional testing: Delete users 200 and 201 from MyGroup - after this practice the group will disappear from Pidgin Workaround: disable IM services for the second group also - both groups will not appear anymore in Pidgin list as expected
X X -1 1 3 15	removing regions left shardId != 0, registrations are lost	Fixed an issue that caused the Mongo registrardatabase to display two contacts with different expires for the same registration. This issue occurred in the Mongo registrar database when there was one registration for the same phone on two different regions, and consequently two different shard Ids. Now, the phone refreshes registration on the same shardId (0) in order not to lose registrations.
X X -1 1 3 17	Push-To-Talk and Paging Groups" page crashes upon edit	Fixed an issue that caused the Push-To-Talk and Paging Groups section to generate a server related error when applying changes to Polycom phones. To replicate this issue in your environment, follow these steps: 1. In the main menu, click Devices>Phones. 2. Click on a Polycom phone in the Phone column. 3. Click the Push-To-Talk and Paging Groups option under the Identification section. 4. Perform one or several changes on the page an d then click the Apply button. The following error was generated: javax.servlet.ServletException: java.lang.IllegalStateException: Form too many keys
X X -1 1 3 20	Permission for retired feature still on the available list - System : SBC (Cisco Integration)	The System : SBC (Cisco Integration) Read permission has been removed from the user interface of the administrator, as it is no longer supported. To review the changes follow these steps: 1.Go to Users>Admin Roles 2.Click the Add new role button. 3.Under the Permission section, notice that the System : SBC (Cisco Integration)Read permission is no longer available under the Available column.
X X -1 1 3 27	Config does not start due to PolycomPhone null pointer exception when replicating	Fixed an issue that caused the Phones page to fail and generate a Polycom Phone null pointer exception when performing the Send Profile action when the phone list contained a Polycom phone.
X X -1 1 3 28	Location delete fails with internal error if no conference role enabled on that location	Fixed an issue that caused the system to generate internal error messages when users deleted a location that did not have an enabled conference role. Now no errors are generated and locations are succesfully deleted.
X X -1 1 3 56	Apache regular expressions break Grandstream config download	The regular expression "AliasMatch" parameters located in the /etc/httpd/conf.d/sipxconfig.conf path caused files that begin with "cfg" to return a "404 not found" error message when requested through HTTP from an OpenUC server. This issue has been fixed by replacing AliasMatch /(.*).cfg "/var/sipxdata/configserver/phone/profile/tftproot/$1.cfg" with AliasMatch /.*cfg.* "/var/sipxdata/configserver/phone/profile/tftproot/$0" . Also, two new matches were added: AliasMatch /gs_config/.* "/var/sipxdata/configserver/phone/profile/tftproot/gs_config/$0" AliasMatch /gs_phonebook/.* "/var/sipxdata/configserver/phone/profile/tftproot/gs_phonebook/$0"
X X -1 1 3 59	Import csv does not ignore empty vociemail pin	The Import CSV functionality no longer ignores empty fields when performing updates. If a value is missing, it does not updates the user properties and the value already defined in siXecs is displayed. For example, if a voice mail pin is empty in CSV, the pin is not updated after CSV import.

XX-11371	Resource list (speed dials) not updated on extension removal	Fixed an issue affecting the SSS service that prevented the speed dial list from correctly updating when the administrator deleted a system extension that was part of a resource list (speed dials) of a user. Now, when a user extension, conference room, park extension, or others are deleted, the speed dial list correctly contains with up to date information.
XX-11374	sipXcallqueue selects random FS Server in cluster	Fixed an issue that caused the sipXcallqueue server to select a random FreeSwitch instance in the cluster when a call queue is added to the system. Now, the call queue is automatically redirected to the local FreeSwitch.
XX-11395	Sipxrecording running on server where it is not enabled	Fixed an issue that caused the sipxrecording process to start and use memory even though the process was stopped on the server. This issue occurred when running a sipp subscribe load test on one server part of a three nodes cluster. Now, the sipxrecording process only runs on the server where it's enabled.
XX-11402	Registrar should preserve callgroup param in contacts	When the the HTTP INVITE request contains the parameter named "callgroup" set to "user", the registrar adds it to every contact returned in a 302.
XX-11404	sipXconfig should add hunt group extension as a param to all its contact	Fixed an issue that caused the system to store hunt groups in the IM database as aliases. Now, the system correctly stores hunt group extensions to the contact URI as a parameter using a new parameter called "callgroup". For example, the aliases previously existed in the MongoDB as: { "id" : "300", "cnt" : "<sip:206@dev-ezuce.ro;sipx-noroute=Voicemail?expires=30>;q=0.95", "rln" : "userforward" }, { "id" : "300", "cnt" : "<sip:207@dev-ezuce.ro;sipx-noroute=Voicemail?expires=30>;q=0.95", "rln" : "userforward" } Now, they are stored as: { "id" : "300", "cnt" : "<sip:206@dev-ezuce.ro;callgroup=300;sipx-noroute=Voicemail?expires=30>;q=0.95", "rln" : "userforward" }, { "id" : "300", "cnt" : "<sip:207@dev-ezuce.ro;callgroup=300;sipx-noroute=Voicemail?expires=30>;q=0.95", "rln" : "userforward" }
XX-11405	Unable to set TCP transport method for Bria Mobile	Fixed an issue that prevented the system from sending the TCP transport method for Bria Mobile. This issue occurred because the transport setting set in UI by the administrator was not correctly translated from .ini to .xml.
XX-11425	empty mp3 file created when voicemail set to mp3 and call hanged up before leaving message	Fixed an issue that caused an exception to be generated when the mp3 file was set to Mp3 and the caller hanged up before leaving the message, thus preventing users from determining the call duration. The issue occurred because the system was not performing disk clean up, thus the temporary messages were still kept on the disk. Now, in case of an empty file, the duration is set to 0 instead of throwing exception. The system generated the following exception: "2014-10-22T10:40:22.357000Z":6:sipXivr:ERR:openuctest.ezuce.ro:Thread-16:00000000:sipxivr:"SipXivr::run" java.lang.RuntimeException: Message::getDuration Problem determining duration of /var/sipxdata/mediaserver/data /mailstore/201/deleted/temp_recording_5288578795931161421.mp3 at org.sipfoundry.commons.util.AudioUtil.extractMp3Duration(AudioUtil.java:48) at org.sipfoundry.voicemail.mailbox.TempMessage.getDuration(TempMessage.java:89) at org.sipfoundry.voicemail.Deposit.clearChannelUUID(Deposit.java:266) at org.sipfoundry.voicemail.Deposit.runAction(Deposit.java:220) at org.sipfoundry.voicemail.VoiceMail.voicemail(VoiceMail.java:53) at org.sipfoundry.voicemail.VoiceMail.run(VoiceMail.java:42) at org.sipfoundry.sipxivr.SipxIvrApp.run(SipxIvrApp.java:31) at org.sipfoundry.sipxivr.SipXivr.runEslRequest(SipXivr.java:61) at org.sipfoundry.sipxivr.eslrequest.EslRequestScopeRunnable.run(EslRequestScopeRunnable.java:24) at java.lang.Thread.run(Thread.java:701) Caused by: org.jaudiotagger.audio.exceptions.InvalidAudioFrameException: No audio header found withintemp_recording_5288578795931161421.mp3 at org.jaudiotagger.audio.mp3.MP3AudioHeader.<init>(MP3AudioHeader.java:138) at org.jaudiotagger.audio.mp3.MP3File.<init>(MP3File.java:379) at org.jaudiotagger.audio.mp3.MP3FileReader.read(MP3FileReader.java:39) at org.jaudiotagger.audio.AudioFileIO.readFile(AudioFileIO.java:286) at org.jaudiotagger.audio.AudioFileIO.read(AudioFileIO.java:150) at org.sipfoundry.commons.util.AudioUtil.extractMp3Duration(AudioUtil.java:44
XX-11446	Call does not roll to voicemail	Fixed an issue that prevented the IVR from sending a call to the voicemail when the callee rejected the call. Previously, the busy tone was sent to the caller. To replicate this issue in your environment, follow these steps: 1. In OpenUC, perform a call from an internal line to another internal line. 2. Reject the call and notice that the busy tone is sent to the caller.

X X -1 1 4 58	Openfire certificate not corectly deployed	Fixed an issue with the Openfire certificate. The Openfire certificate was incorrectly deployed by the system with ssl-web.crt + ssl-web.key as a keystore in /opt/openfire/resources/security/keystore. Note: the trusted authority needs to be added under System/Certificates/Certificate Authority tab, in order to be validated by Openfire.
X X -1 1 4 62	Bring back old log rotate behavior where log file gets recreated every 5 seconds	Fixed an issue with the sipXportLib library that caused the snapshots to generate 0 bytes log files. Now, the log file is rotated on a daily basis and snapshots are correctly generated.
X X -1 1 4 83	14.04 Update 3 Can't add names to Voicemail Directory	Can't add recorded names to the voicemail for a user and have it work in the Directory. In mailstore, under the extension, the name is recorded as an MP3 file, even when system is set to WAV. The MP3 seems to be malformed and can't be played. Deleting it, and copying a good name.wav file seems to fix things. This seems to be okay for names that are imported. Once you delete the working file, you can not create a new one, nor record a new one to replace the working one. Effected only sipXcom and not openUC.
X X -1 1 4 86	Obsolete sipxevent (event notifier) service should be removed	The sipxevent (event notifier) service has been removed.
X X -1 1 4 94	Sec service using a lot of CPU after update to 14.10 until sipxlogwatcher restart	When upgrading OpenUC from 4.6 Update 11 or 14.4.3 to 14.10 follow these steps to ensure the SEC service is correctly running: 1. Upgrade from 4.6 Update 11 or 14.4.3 to 14.10. 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.
X X -1 1 5 06	OPTIONS messages not being sent to remote phones	Fixed an issue that prevented the NatMaintainer service from sending OPTION type messages to remote phones.
X X -1 1 5 14	myBuddy does not connect after reboot	The real issue here is when imbot is enabled and system is rebooted, mybuddy won't connect, unless service sipximbot restart is executed after system is rebooted. This is because JAVA DNS lookup does not return correctly on reboot. After a reasonable delay after reboot JAVA DNS lookup performs correctly. There is a DNS lookup cache in JVM, but on false results should expire by default in 10 seconds. However, it still does not perform correct lookup, because Java does not react when /etc/resolv.conf is updated with correct data: https://community.oracle.com/thread/1148912?start=0&tstart=0
X X -1 1 5 30	Upgrade to 14.10 fails due to old dependency	Erlang version was changed and obsolete openacd package still use old Erlang version Error: Package: openacd-2.0.0-67.60c2b.x86_64 (@sipXcom) Requires: erlang = R15B Removing: erlang-R15B-03.2.el6.x86_64 (@sipXcom) erlang = R15B-03.2.el6 Updated By: erlang-R16B-03.9.el6.x86_64 (sipXcom) erlang = R16B-03.9.el6 see https://www.sipfoundry.org/topic/14-10-update-errors-erlang/ Marked openacd as obsolete
X X -1 1 5 43	Exception in Call Queue Settings page	Fixed an issue that caused a page crash on the Features -> Call Queue -> Settings page
X X -1 1 5 44	IVR mailbox REST: Content-type header is not sent	Fixed an issue with the REST service that retrieves voicemails in that it did not set the response "content-type" header . Some consumers (such as IE) will not read the audio file if the header is not set.

XX-6791	Park server leaving MOH sessions and parking slots "stuck"	Fixed an issue that caused the Park server to drop from MOH sessions and the parking slots to become unresponsive for Polycom 550 and 650 phones. This issue has been fixed by porting the Park Server to the Free Switch server.
XX-9341	SIPXConfig Phonebook /PagedPhonebook REST API returns IM Passwords of Other Users	Fixed an issue that caused the /my/pagedphonebook and /my/phonebook URIs to generate the IM passwords of other users in the system when using the GET HTTP request. Now, passwords belonging to other users passwords are not returned any more.
XX-9355	Voicemail is blank when forwarded a second time	When a voicemail is forwarded a second time using IVR, the final recipient receives a blank voicemail. Call in -> User A VM - forward to -> User B VM - forward to - > User C VM User C will have a blank VM. There is a wav file, but it has no sound. It is the same if you listen to it on the web interface or through a handset. If the recipient has their VM set to forward to their email, the attached wav file is 0 bytes. The email notice and the info in the users web gui show 47 seconds (in my example) so it seems to to think the message is there, but it is not.
XX-9394	Remove bria port from Firewall	Bria provisioning port 12000 is not used any longer, remove this port from Firewall configuration.
XX-9504	After Modifying the Scheduling Period,It is not getting updated to the Dial Plan	Fix for schedule periods not being updated in the dial plan after being modified. Steps to reproduce 1.Add Scheduling say Testing for Everydays from 9 am to 5 pm in Dial Plan scheduling page 2.Enable the International Dial Plan with Scheduling Testing and Long distance Prefix =00 External no =Any digit 3.Between 9Am to 5pm Dial 009199862527263 from user(which International dialing Enabled) 4.The Call will Go through 5.Navigate to Scheduling page and modify the Testing scheduling Period from 9am to 6pm 6.wait for few minutes and Between 5.15pm to 6pm Dial 009199862527263 from user(which International dialing Enabled)

Expected Results 1.Either it should display warning message to restart of services when changes made in Scheduling page OR 2.Should get automatically updated to International Dial Plan and call should go through

Actual Results 1.There is no warning Message for Restart of Services 2.Not getting update automatically to International Dial Plan and call is not going through

Observation: Each time after modifying the Scheduling Period and want to navigate to Particular Dial Plan(for that scheduling) Click on OK button and Restart the services and then the changes made in Scheduling page is updated to the this Dial Plan |

Known Issues

UC-2915	Blind outgoing transfer fails through ITSP	An issue exists where a blind transfer through sipXbridge to an External Number does not properly complete. Call does not get dropped. Workaround: Use consultative transfer
UC-3306	Bria MAC users cannot see other users' XMPP presence after openfire restart	If Bria for Mac is used with XMPP enabled users cannot see other users's XMPP presence after openfire restart

The following entry can be seen in openfire db, if dropping this record and restarting openfire issue gets solved. |
| UC-3379 | Document that SAA works only on TCP w/ phones registered over TCP | The Shared Appearance Agent (SAA) service has the following limitations: - listens only on TCP; - requires that all phones that have shared lines to be registered over TCP. |

UC-3420	Retrieving a shared line from a held call stops rendering for both participants	Known issue with odd hold handling with Polycom phones in 5.1.1 and 5.2.0 firmware. Polycom firmware 5.1.1: Setup: 201 and 202 shared lines phone 1 : 201 phone 2: 201 and 202 phone 3 : 202 (3 lineKeys) 1. Call established between 201 (phone1) and 202 (phone3) 2. Press hold on both phone1 and phone2; on phone2 the lines are shown as being able to be picked up (rendering) 3. Resume call from 202 phone3, on phone2 the lines are both shown as busy, although 201 on phone1 is still holding the call Polycom firmware 5.2.0: a) 201 calls 202 b) 202 holds the call then retrieves it c) 202 holds again the call d) 201 holds the call also e) 202 retrieves the call : both shown as busy although 201 is still holding the call Leave the call ongoing: after 270 seconds (subscribe keep-alive expires:300 - 30) line 201 is shown as holding to reproduce in the same call repeat d) and e), otherwise, for new call, start from a)
UC-3436	Polycom DND does not work for shared lines	Known issue with Do Not Disturb and Polycom phones with shared lines. To reproduce: 1. Add a shared line on a Polycom phone(vvx500, vvx400) 2. Enable DND for that line 3. Make a call, the line will ring, and it should not. Looks like DND is broken if line is shared. DND works fine if line is not shared. This has been reproduced with Firmware 4.1.7.
UC-3445	Wrong Polycom phone behaviour when adding the same shared line twice compared with 1 line with 2 num-line-keys	Known Issue where if only one line is added with line-keys set to value 2, the phone display shows two lines, and it works as expected - both lines attempt line seize before a call. If the same line is added twice (by using the openuc interface -> click on the phone -> Lines -> add line) on a Polycom phone and line-keys is set to value 1, only 1st line (the first shown on phone display) will perform line seize NOTIFIES before a call and, also, only that line will generate NOTIFY messages to SAA. The 2nd line will act as if is not a shared line at all.
XX-11287	Records requiring update	Several system audit related records are not automatically updated after data has been changed. This issue currently occurs because because the property name includes information taken from code and not from the web interface.
XX-11401	Forcing primary database when not a majority, rewrites replica set configuration	When only one node remains operational in a HA environment, it becomes by default Secondary as there are no other servers to vote a Primary one. If users forced that remaining node to become Primary, the system creates a new replica configuration set that only contains that node and ignores the ones that are not functioning. It is recommended not to force a node to become Primary.
XX-11436	Recreating a region with phones registred results in doubled registrations for same phone (one of them expired)	Fixed an issue where deleting a local database with registered phones resulted in doubled registrations for the same phone , out of which one of them is expired. This issue occurred in a HA environment with multiple nodes with regions and local databases, in the following scenario: 1. Register a new phone with a new line in a region. 2. Delete all local databases in that region. 3. Click Diagnostics>Registrations in the main menu. Notice that the same registration is doubled: one available and another one expired. This occurres because when sipXconfig changes the shard IDs to when cfengine clears the local mongo registration db, the sipXconfig thread picked up the registrations again from the local database and replicated them into the global cluster resulting in the doubling of registrations in the global cluster.
XX-11487	If a user changes their password in User Portal config services should re-send profiles	If users changed their password in Unite Web, the configuration services do not check for users having Bria or Unite Mobile phones and do not re-send those profiles to get new password into XMPP configuration for the devices. The current workaround is for users to send the phone profiles using the Speed Dials page.

X X - 1 1 4 98	Apply needed on Gateways page to disable SIP Trunking service (server cluster related)	Known issue that prevents the calls from being connected when disabling SIP Trunk,Proxy and Registrar for a node in a HA environment. Currently, the workaround is to go to Gateways>Your sip trunk and click Apply in order for the incoming calls to be able to connect.
X X - 1 1 5 02	Local region only 100% / All other regions 100% fail over plan results in failed calls / unregistered phones	Several performance issues affect the system when using HA environment because of the MongoDB system. We need to next move to upgrading MongoDB is updated to version 2.6 to make Regions work as envisioned.
X X - 1 1 5 06	OPTIONS messages not being sent to remote phones	Reported issue wasn't OPTIONS but the NAT tcp keep alive setting which was not set. When the phones are behind NAT and registered over TCP, an extra TCP connection will be created with the public address ({public_host:public_port}) to be channel the communication between the phone and proxy. The problem is that, when using AWS (could be with other firewalls too), the public connection does not receive data on it for about ~20/30 secs, AWS considers the connection to be idle and closes it by force. Therefore, in order to prevent this kind of situations the NAT TCP keep alive interval should be set to something <= 10 seconds: from admin page, click on the phone(s) that is behind NAT, goto the 'NAT' settings --> click on 'Show Advanced Settings' --> clear 'SIP Signalling Port' and ' Media Port Range Start' --> set 10 in the 'Keepalive interval' box.
X X - 1 1 5 13	Openfire single node - failing scenario	Known issue that causes the system to fail to correclty enable an IM account when the following steps are performed by the administrator when OpenFire is running on a singe node: 1. Change IM group name. Notice that the IM clients successfully updated with new group name, roster and presence. 2. On the Instant Messaging tab, deselect the Enable IM Account check box. Notice that the IM users from that group are disabled. 3. On the Instant Messaging tab select the Enable IM Account check box. Notice that the system is not successful in performing the operation: users have IM enabled back again, but their roster is either missing or not complete.