sipXcom 16.12

Summary

We're continuing to focus more on fixes and minor improvements in 16.12 as work ramps up on the next generation of sipXcom.

Also as always, thanks to the Dev & QA team at eZuce for their excellent work on this release. Thanks also to IANT for a number of fixes to Polycom and Yealink devices.

In all 28 issues (enhancements / fixes) are addressed for sipXcom in this beta release.

The next sipXcom release will be 17.04.

Highlights

sipXcom New Features:

- Conference Bridge Entry / Exit tones configurable
- Conference Bridge Voice Announcement of Entry / Exit
- Grandstream 2130, 2140 and 2160 phone configuration support

sipXcom Improvements:

- Autoprovision for new 301, 311, 401, 411, 501 and 601 phones

Phone Software Supported:

- Polycom- 4.0.9 for SoundPoint IP, 5.2.5 for VVX
- (Note that 4.0.6 for SPIP may be preferable for BLA and it is reported that 5.5.1 for VVX seems to be working properly)

Who Should Install?

This release is recommended for all 4.6 and later installations.

Questions

Please post to the sipXcom-users google group if you have questions.

https://groups.google.com/forum/#!forum/sipxcom-users

New Installs

A new ISO is available for 16.12 at: http://download.sipxcom.org/pub/sipXecs/ISO/

Update

To update please edit your /etc/yum.repos.d/sipxecs.repo file and reference the new download server (download.sipxcom.org). The repo should look as follows:

```
[sipXcom]
name=sipXecs software for CentOS $releasever - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/16.12/CentOS_$releasever/$basearch
gpgcheck=0
```

To edit this file, login to your sipX server as root and then use either vi or nano (easier).

vi /etc/yum.repos.d/sipxecs.repo

or

nano /etc/yum.repos.d/sipxecs.repo

Once the repo file is modified, run:

yum clean all

yum update

	JIRA name	RN Content	Enhancement/Fix /Known Issue	Key words
SIPX-389	Cannot acess HTML reports	Fixed an issue caused by upgrading a 15.08 production site to 15.10. The administrator cound not access the HTML reports in Call Detail Records > Reports. PDF, CSV, and XLS files can be accessed without problems. HTTP ERROR 500 Problem accessing /sipxconfig/reportsService.svc. Reason: Server Error	Fix	Config
SIPX-509	Polycom Auto Provisioning missing default firmware	Fixed an issue with Polycom firmware 5.2 or 5.3.1's default value for provisioning. The 000000000000.cfg will be generated without default firmware. The default firmware for some VVX phones will not set to 5.x too. The missing firmware caused bootloops with formated phones if you don't have the old firmware that is hard coded inside the template	Fix	Polycom
SIPX-517	Mongo plugin not entirely ported to open source	Fixed an issue with sipXcom where the plugin hazelcast.jar and some mappings in openfire.properties were missing.	Fix	IM
SIPX-521	Prompt for user name on entry to conference	Added a new feature to have the system be able to prompt a user to speak their name and then be announced into a conference. This should be optionally enabled for a conference room in the sipXcom / Uniteme Admin GUI. When a user dials a conference bridge with the Voice Announce Arrival option enabled the user will be prompted to speak their name and press #. The system records the name and plays it to entire meeting as the user is brought into the meeting. The caller will first hear "Please say your name after the tone" which will utilize the canned Freeswitch audio file ivr-say_name.wav. A pound symbol (#), a total of 3 seconds of recording time or 2 seconds of silence shall end the recording. Playback states "{recorded name} has joined the meeting." This will utilize the canned Freeswitch audio file conf-has_joined.wav. This should utilize the localization included with sipXcom / Uniteme. We'll need to store these names somewhere along with inbound sip uuid to have Feature #2 work. Names will need to be erased after if SIPX-521 not enabled. English only is ok. Example Freeswitch Code: <extension name="Record Name and schedule conf announce"> <condition field="destination_number" expression="^55(3\d\d\d)$"> <action application="answer"/> <action application="set" data="namefile=/tmp/${uuid}-name.wav" inline="true"/> <action application="sleep" data="1000"/>> <action application="playback" data="voicemail/vm-record_name1.wav"/> <action application="playback" data="tone_stream://%(1000,0,500)"/> <action application="record" data="${namefile} 1"/> <action application="playback" data="ivr/ivr-call_being_transferred.wav"/> <action application="set" data="res=${sched_api +1 none conference $1-${domain} play file_string://${namefile}!conference/conf-has_joined.wav}"/> <action application="transfer" data="$1 XML default"/> </condition> </extension>	Enhancement	Conference
SIPX-522	Play name of person leaving conference	Added a new feature to have the system be able to announce the departure of a user from a conference. This should be optionally enabled for a conference room in the sipXcom / Uniteme Admin GUI. This feature can only be enabled if Feature #1 is enabled. Playback states "{recorded name} has left the meeting." This will utilize the canned Freeswitch audio file conf-has_left.wav. This should utilize the localization included with sipXcom / Uniteme. Erase recorded names on user exit.	Enhancement	Conference
SIPX-523	Make meeting entry and exit tone optional	Added a new feature to make meeting entry and exit tones individually optional for conference bridge. At present tones are played on conference entry and exit. These can be overwhelming for large conferences. These two tones should be optionally enabled and disabled in the sipXcom / Uniteme Admin GUI.	Enhancement	Conference

SIPX-524	Yealink DND Local Processing	Enhancement added to take advantage of newer Yealink Firmware's (>=x.x.193.95) behaviour of handling DND and Forwards, when application server feature sync enabled. In this mode Yealink requieres that the PBX will process the DND. TIn upcoming releases there will be a config parameter to change this to the old behaviour and make it work again with SipX Parameter is: features.dnd.feature_key_sync.enable Default value 0 Change in behaviour can only be seen if feature sync is enabled	Enhancement	Yealink
SIPX-525	Path to mod_com_g729 freeswitch module hardcoded to the old/wrong fs module directory	Fixed an issue where cfengine tries to detect mod_com_g729 in /usr/lib64/mod rather than /usr/lib64/freeswitch/mod where these modules are now.	Fix	Config
SIPX-529	Verify BLF Method has side effects	Fixed an issue that was introduced with UC-3956 and it showed up only when having System->Admin Setting->"Allow subscription to self" enabled (which is disabled by default). A wrong list of BLFs will show up once you configured an user with a mix of Speed dials with presence(blfs) and some without presence. Add the following Speeddials/BLF for User 200 201 (BLF) 202 (Speeddial) 203 (Speeddial) 203 (BLF) on the phone of User 200 you get 201 (BLF) 202 (BLF) 203 (BLF) 203 (BLF) 202 (Speeddial) 203 (Speeddial)	Fix	BLF
SIPX-530	Autoprovisioning of Polycom VVX 301, 311, 401, 411, 501, 601	Fixed an issue with autoprovisioning with new Polycom VVX models.	Fix	Polycom
SIPX-72	Add support for Grandstream GXP 2130, 2140 and 2160	New Feature to add support for Grandstream GXP 2130, 2140 and 2160.	Enhancement	Grandstream
UC-4001	Strip leading characters from username in LDAP import	Enhancement so that an administrator can now strip leading characters from the userid as it is imported from LDAP. Make the number of characters to be stripped configurable from 0 to X. This setting would be on the resulting username field. Default will be 0.	Enhancement	LDAP
UC-4158	REST API Enhancement for partial user data update	Enhancement to allow only certain fields to have to be updated when adding or updating users. For example when addding new users the admins should be able to set only the required updated fields.	Enhancement	API
UC-4216	Unite Web speed dial bugs	Fixed a couple issues with Unite Web & Unite Web Lite. 1) Feature codes can't be entered as speed dials. They must fail validation and you can't save them. If you have feature codes (starting with asterisk) already programmed, and the user logs in and tries to change speed dials, the feature codes are not shown and the user cannot save speed dials unless they remove those containing feature codes or change them to something else. I have attached screen shots for reference. 2) If a user edits speed dials and subscribes to presence on an extension that they can't subscribe to presence on (it's not a valid extension, etc.) it doesn't give them any error and it won't save. In some browsers you can notice a very small "warning" box that pops up for less than a second and goes away, however the users don't know what is going on.	Fix	Unite Web
UC-4240	Enhancement to have system backups to run under cgroups	Enhancement request to run backups under cgroups to control resources available to mongodump for config/voicemail backups.	Enhancement	Backup
UC-4266	Voicemail backup does not cleanup <tmpdir>/dump directory	Fixed an issue that caused backups to be larger than they needed to be. Steps to reproduce: 1. Run a backup. 2. Run a backup again. When temporary files backup directory became configurable a bug was introduced related to voicemail backup: after voicemail backup is finished, the temporary dump file directory is not cleaned up (the default is /var/sipxdata/tmp/dump). As a result, if a configuration backup executes next, the vmdb dump directory will be included in the configuration backup	Fix	Backup
UC-4267	Config generation fails with external Line on Polycom Phones	Fixed an issue with adding a second line to Polycom phones. Steps to reproduce: Take a normal Polycom Phone (with internal Line or not) and add an external Line (must not exist on external system) After send profiles generations fails with null point exception	Fix	Polycom
UW-184	Conference bridge: participant: drop /mute/mute speaker does not work	Fix an issue with Unite Web on iOS devices where if a user clicks a particpant the controls on it do not show.	Fix	Unite Web
UW-201	In Unite Web remove obsolete options Moderated & Public	Enhancement to remove obsolete conference bridge settings. The "moderated" and "public" options became obsolete in 14.04 and should be removed from the Settings -> Conference Bridge page in Unite Web.	Fix	Unite Web
UW-231	Adding a bad speed dial from Safari doesn't display error message	Fixed an issue that was seen in Safari only where a small "warning" appears briefly if you are very attentive. Steps to reproduce: 1. Use Safari to login to new user portal (Unite Web or Unite Web Lite) 2. Navigate to Settings -> Speed dials. 3. Add speed dial with presence with a wrong extension will not show an error.	Fix	Unite Web
UW-347	Unite Web forwarding option should support e164 format	Enhanced number validation to allow users to enter e164 formatted numbers as speed dials in Unite Web.	Fix	Unite Web

UW-350	Schedule not getting saved in Unite Web	Fixed an issue with user created schedules in Unite Web not saving. Steps to reproduce: 1. User 201 ans 202 must be registered on the system 2. Log in with user 201 in unite web 3. go to settings>call forwarding 4. click on set-up tab 5. create a call forward to user 202 6. after the call forward is created, go to the schedule tab 7. click on add new schedule 8. fill the fields with the data. 9. click add and save Issue: schedule is not saved.	Fix	Unite Web
UW-371	Validation not working when editing a speed dial	Fixed an issue with number validation when editing speed dials. The validation worked properly on entering a value. Steps to reproduce: 1. From Unite Web if you add a simple speed dial with incorrect value in Number field, the Number field will get Orange and not allow you to save the input. 2. Try to edit an existing simple speed dial(no presence) and you go and edit the Number field from Unite Web, and you are able to enter some bad input in there This did not happen if you edit a speed dial with presence. In this case you will not be allowed to save the input, but rather be informed through a not very intuitive "Warning" message - UW-231. This also did not happen if you edit speed dials with presence or not from Admin Portal - validation works in all cases here.	Fix	Unite Web
UW-372	When the user calls an extension by name this will return Number Not Found	Fixed an issue when a call is made from the system phonebook in UniteWeb Lite the call would not complete properly. The first user is called on the phone (first user is the same user which was logged in UniteWeb Lite), once the first user accepts the call the call will end immediately and in unite web lite will receive the following message: "Number not found". Two phones needed, users configured with first and last name, members of the same phonebook Steps: 1. Login with a user in Unite web Lite 2. Select Dialpad and search for the second user by his name 3. Click on match that was found 4. Click call button This issue is reproduced on all browsers.	Fix	Unite Web
UW-373	Disable override will not work	Fixed an issue where when the user checked "override default autoattendant" the setting would not stay enabled. Steps to reproduce: 1. Login with an user on unite web portal 2. Go to Menu>Settings>Personal Attendant 3. Check the "override default autoattendant" 4. Select any language 5. Save the modifications 6. Uncheck the "Override default AutoAttendant" 7. Save the modifications Issue: After saving you will see that the Override will be restored to enable. To disable override you must select EN language, than uncheck the override and than save the modifications This issue is reproduced on all browsers, and is reproduced on unite web and unite web Lite	Fix	Unite Web
UW-374	UniteWeb Lite - Search by name on dialpad will not work	Fixed an issue with UniteWeb Lite where searching by last name or first name will not return any results. Steps to reproduce: 1. Login with an user on UniteWeb Lite 2. Open dialpad 3. Search by last name or first name	Fix	Unite Web
UW-378	After accessing voicemail section a second time, current folder is set to an empty folder instead of inbox	Fixed an issue with UniteWeb that caused voicemail folder to appear empty. This occurred on all platforms/devices. Steps to reproduce: 1.Login to UW and go to Voicemails Result: Current folder shows "inbox" - and shows your voicemails if you have some 2.Close the voicemail section by clicking "X" in the top right corner 3.Go to Menu and go to Voicemails again Issue: Current folder shows " " nothing , and the folder is empty , does not have any VMs in it. You can tap the Current folder drop-down and choose - inbox. Now you will be inbox.	Fix	Unite Web
UW-381	Moderated and Moderator Pin settings not saving	Fixed an issue with UniteWeb that prevented the 'Moderated' check box and Moderator PIN to not be saved in the user's settings. Steps to reproduce: From Web browser, log into the Unite web portal. Go to settings, User settings. Enter a moderator PIN and save. or... check the moderated check box and save. Navigate away from this settings page (e.g. go to voicemails) and then return to the Settings - User Settings page. The setting changes that made are not there anymore.	Fix	Unite Web