sipXcom 16.02

Release Notes

Summary

sipXcom New Features:

- Easier portal customization with 3 layers of templating

sipXcom Improvements:

- Change some system defaults related to call transfers
- New User Portal - Sort Voicemail by date
- New User Portal - Only show Conference tab if user has a Conference Bridge
- Polycom SoundPoint IP 4.0.8 and Later Firmware Support
- Yealink plugin updates (contribution)

Phone Software Supported:

- Polycom- 4.0.9 for SoundPoint IP, 5.2.5 for VVX

Who Should Install?

This release is recommended for all 4.6, 14.XX and 15.XX installations.

New Installs

A new ISO is available for 16.02 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 - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/16.02/CentOS_$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	Keywords
SIPX-397	Polycom SPIP 4.0.8 & Later firmware	Added an enhancement for some new parameters to firmware 4.0.8 and later firmware for the SoundPoint IP phones. One in particular causes the phone to close TCP socket connections which can break remote phone operation. Add the following new parameter and ensure phone is configured with the value set as 1. tcpIpApp.keepalive.tcp.sip.persistentConnection.enable 0 or 1 (0 = default) If 0, the TCP Socket connection is closed after 1 minute. When the phone sends a new SIP message, a new connection is opened. If 1, the TCP Socket connection remains open indefinitely. Large installations may want to be able to set the value to 0 to keep TCP socket connections lower.	Enhancement	Polycom
SIPX-400	3 level plugin overwrite support	Added an enhancement so that GUI plugins would work to 3 levels making customization of portal easier. Ensure a depth of 3 for plugin overwriting level 1 - sipxplugin2.beans.xml level 2 - sipxplugin.beans.xml level 3 - sipxplugin0.beans.xml level 2 overwrites level 1 level 3 overwrites level 2 level 1 beans will be loaded level 2 beans will be loaded and will overwrite any beans with same name from level 1 level 3 beans will be loaded and will overwrite any beans with same name from level 2	Enhancement	Config
SIPX-424	Yealink plugin V8X	Enhanced the Yealink plugin with new phones, add firmware version 8 model files, and correct number of lines on T4 phones.	Enhancement	Yealink
SIPX-92	Change Polycom Phone QOS Default for Call Control so DSCP = 24	Change Polycom defaults for QoS for call control so DSCP = 24. In SipXConfig Polycom Phones QOS settings page The default TOS settings equate to a DSCP of 44. Call Control (current) Prec Dly Thpt Rlb 5 Chk UnCk UnCk 101 1 0 0 seventh bit (Cost) 0 As DSCP = 44 Call Control (proposed) Prec Dly Thpt Rlb 3 UnCk UnCk UnCk Seventh bit (Cost) Uncheck 011 0 0 0 seventh bit (Cost) 0 As DSCP = 24 RTP equates to a DSCP of 46 and if this Jira is implemented, that the Call Control settings equate to 24. A call control setting of 24 is consistent with recommendations by eZuce and Cisco.	Enhancement	Polycom
UC-3859	Defaults Related to Transfers should be changed	System enhancement to change the following 3 values to the opposite of what they default to in order to avoid transfer issues. We should change the default values of these options. Parameters and current defaults: System-->Media Services (Allow Blind Transfer) default: unchecked System-->Media Services (Simplify Call After Transfer) default: checked System--Voicemail (Transfer by Bridging the call) default: false Parameters and proposed new defaults: System-->Media Services (Allow Blind Transfer) default: checked System-->Media Services (Simplify Call After Transfer) default: unchecked System--Voicemail (Transfer by Bridging the call) default: true	Enhancement	Config
UC-3897	CDR Reports must be able to be reported by time zone	Allow CDR Reports and Downloads to have time reported by a particular time zone. In Diagnostics -> Call Detail -> Historic and Diagnostics -> Call Detail -> Reports be able to select (and remember) a Time Zone to display call records in. Download of CSV must also be in selected time zone.	Enhancement	Config

UC-3907	BLF and MWI URI fields should be empty on phone profiles when not used	Added an enhancement so that BLF and MWI URI fields are empty for phone profiles that don't use BLFs or need MWI. Set "Polycom phone"->Line->Messaging->Subscribe field to blank if the User assigned to this phone has User->Permissions->Voice Mail disabled. ex: "Hoteling User has MWI, normal User on Phone has no MWI Hoteling User will login on the phone and has a waiting Message Hoteling User will logout and the MWI is still there" And if the user does not have any Speed dials with presence or group speed dials with presence defined, then on his phone set "Polycom phone"->BLF->Subscription URL field to blank. For example: "User 200 is registered on phone A with the BLF 201 and 202 The User 201 has no BLF and no permission to use BLFs If User 201 use hoteling to log in on phone A, he will get the BLF of User 200 (BLF 201, 202), because the attendant uri don't change and he tries to get the BLF of user 200."	Enhancement	Polycom
UW-168	Allow upload of avatar from the user portal	Added an enhancement to allow users to be able to upload avatar from user portal.	Enhancement	Unite
UW-319	Don't Display Conference Bridge Section in Unite	Added an enhancement to not display Conference bridge section if user has no conference bridges defined. Currently Unite will show a conference bridge section even if there are no conferences defined for the user. In that section it will show what appears to be an error message stating: Feature Not Available Please Contact your Administrator This causes confusion for the users since and they do indeed "contact their administrator" even though there is nothing wrong but rather the feature is just not turned on for them. We should remove that section all together in the Unite display if there are no conferences for the user.	Enhancement	Unite
UW-321	In UniteWeb and Unite Lite voicemail is not sorted by date	Added an enhancement to sort voicemail messages by date instead of by name.	Enhancement	Unite
SIPX-158	Main cdr fixes	Fix an issue with CDR to properly handle forwarded calls. When a call goes out of sipx through a gateway or a SIP Trunk, we rely on the contact that other part sent back to us to build cdr, and this is often useless to get information about the called party. This is even worse when you have some call forwarding active, in this case you miss the actual entity reached from the call. Often can be useful to know about which gateway/trunk is engaged when a call exit from sipx.	Fix	CDR
SIPX-279	zen 4787 : user registration page error on user with multiple bria phones assigned	Fixed an issue with registration page for an user assigned to more than one Bria phone. Workaround is to search registrations in global page for that particular user using the web browser's find function.	Fix	Config
SIPX-335	In mongo, users point to the old location name after a location name change	Fixed an issue caused by changing a location name for a user, the old location name would remain. Steps to reproduce: 1. Create a location 2. Assign user group to this location and verify in mongodb User points to the correct location 3. Change location name to a new one and verify in mongo Issue: Users still point to the old location name. Workaround: Click Apply on the user group	Fix	Locations
SIPX-37	Consultative transfer to park orbit does not work	Fixed an issue with users performing a consultative transfer to park orbits. Steps : 1.Define a park orbit extension from Features->Park Orbit : extension 888 2.2012 calls 2048 3.2012 parks 2048 to the park orbit extension via a consultative transfer (not blind transfer) Expected : 2012 to be able to park the call Actual : 2012 hears a busy tone when trying to park the call Note: Consultative transfer to park orbit may disrupt BLF, recommend blind transfer only.	Fix	Park
SIPX-398	Remove IMAP settings from Unified Messaging settings in User Groups	IMAP is no longer supported. Remove IMAP server host, IMAP server port and TLS from User and User Group settings.	Fix	Config
SIPX-434	Yealink 7x/8x misleading tooltip	For Yealink Firmware Template 7x/8x Preference > Backlight Time (seconds) Currently the config "Always" means backlight always off Configuration of 0 seconds will mean always on Updated tooltip to make it more reasonable	Fix	Yealink

SIPX-59	Scheduled backup by specified day does not work	Fixed an issue caused when scheduling backup for one particular day and time. Steps to recreate: 1.Go to Backup and setup a scheduled backup : let's say Monday 17:00 PM (if current day is Monday) 2.Wait for specified time to arrive Reported issue : backup is not triggered. Additional info : scheduled backup setup as "Everyday" 17:PM will work. Backup Now also works. There is nothing related reported in the backup/sipxconfig.log	Fix	Backup
SIPX-61	Call Forwarding CDR problem	Fixed an issue where if an extension (e.g. 200) is forwarded to mobile phone, and another extension or incoming number dials 200, and talks to the mobile number, mobile phone number of the callee cannot be displayed in the CDR.	Fix	CDR
UC-1382	Consultative transfer must point to the exact gateway that accepted the initial INVITE	Fixed an issue caused by multiple gateways on a single dial plan, consultative transfer needed to point to the exact gateway that accepted the initial INVITE. When there are multiple gateways configured to handle the call, there is no guaranty that the initial INVITE and the transferred INVITE with replaces will follow the same forked path. The proxy should be able to tag the REPLACEs header with the absolute URI of the gateway that accepted the initial INVITE. The registrar fallback plugin should be able to parse this tag and return it as a solitary contact. How to replicate: 1. Create 2 gateways in the same branch. Let's call them gw01 and gw02 respectively. 2. Point a dial plan to both gw01 and gw02 as tandem failover 3. Bring down gw01 4. Call a user and put it on hold. 5. Using line 2, call the dial plan you have created (gw01 is down so gw02 should accept the call). 6. Bring up gw01 7. Transfer the call. Since gw01 is now online, the transferred call will go to gw01. This will fail because the call being replaced is in gw02. This is just a manual scenario. In a real deployment, all sorts of timeouts can happen making forks failover which increases the chance that an attended transfer hits a different gw.	Fix	SIPCore
UC-3153	Do not allow presence subscriptions to self	Fixed an issue caused when users subscribed to their own presence. Changed config webui to not allow users to subscribe to themselves. If the user has "Use Group Speed Dials" checked, this should also be filtered to exclude the user him/herself from the group list. Slightly related to UC-134.	Fix	Config
UC-3226	For distribution lists, increase setting_value table from its "value" field currently limited to 1000 characters	Fixed an issue caused by creating very large distribution lists whereas the field that contained this only allowed for 1000 characters.	Fix	Config
UC-3628	REST API for call action doesn't escape + char	Fixed an issue with REST call action API such that it didn't escape the '+' character properly.	Fix	API
UC-3812	Call Park BLF and Line Appearance - Wrong CallerID	Fixed an issue with the CallerID displayed when a call was retrieved from the park orbit. The CallerID would show what was dialed to unpark the call. 1. Call from 2159663354(PSTN) -> 2676380305(alias for x201) 2. Answer call. 3. Blind Transfer from x201->5007(Call Park) 4. MoH to caller and Call Park BLF light and Line Appearance popup on x203 are as expected. 5. x202 fetches CP call via *45007 successfully. 6. BUT x202 does NOT see CallerID of the remote party (2159663354) but rather *45007. 7. ALSO, the CP BLF on x203 is STILL LIT, even though call has been fetched out of park	Fix	Park
UC-3864	/var/log/messages " There was a mount error, trying to mount one of the filesystems on this host."	Fixed an issue with the new SIP packet capture tcpdump based service when running on multiple servers. Log entry noted on fresh install Uniteme 15.12 /var/log/messages seems to be related to new tcpdump service .. : [root@uc1 ~]# grep -c 'mount' /var/log/messages 169 [root@uc1 ~]# grep 'mount' /var/log/messages | tail -n 5 Jan 11 11:42:59 uc1 cf3[1656]: There was a mount error, trying to mount one of the filesystems on this host. Jan 11 11:42:59 uc1 cf3[1656]: There was a mount error, trying to mount one of the filesystems on this host. Jan 11 11:42:59 uc1 cf3[1656]: There was a mount error, trying to mount one of the filesystems on this host. Jan 11 11:42:59 uc1 cf3[1656]: There was a mount error, trying to mount one of the filesystems on this host. Jan 11 11:43:00 uc1 cf3[1656]: There was a mount error, trying to mount one of the filesystems on this host.	Fix	Config
UC-3896	Webui rejects longer TLD names as invalid	Fixed an issue with the tapestry email validator where it doesn't allow for top level domain names with more than 4 characters. Fixed by changing validator regular expression.	Fix	Config
UC-3938	Searching for external numbers in the call history often fails	Fixed an issue when searching for external incoming calls in the CDR doesn't return reliable results. The calls are listed in the CDRs, but it is sometimes not possible to search for specific numbers - the search either returns the correct results, or no results at all. The search results are consistent; when one number is entered into the search, it always appears while another number is never found even though it is also listed in the call history. Because the CDRs are much larger for some customers, they rely more heavily on the search function. We've been able to recreate this issue on many software versions; 14.10, 15.06, 15.08, 15.10.	Fix	CDR
UW-206	Settings: change pin or password - no validation message	Fixed an issue where if a password is input with less than 8 chars the input changes border color, but there is no validation message. If a user is LDAP managed they should not be able to change their password and there should be a message as such. When changing PIN there is no validation message either.	Fix	Unite

SIPX-430	Unable to remove Homer /sipxhomer	On V15.12, with the introduction of Network Packet Capture, Homer cannot be removed due to: dependency on package sipxecs-- root@s01 ~]# yum remove homer sipxhomer sipxhomer-config sipxhomer-proxyplugin Loaded plugins: product-id, rhnplugin, security, subscription-manager This system is receiving updates from RHN Classic or RHN Satellite. Setting up Remove Process Resolving Dependencies --> Running transaction check ---> Package homer.x86_64 0:3.2.5-2.el6 will be erased ---> Package sipxhomer.x86_64 0:15.12-4124.49451 will be erased --> Processing Dependency: sipxhomer >= 15.12 for package: sipxecs-15.12-7798.ee34c.x86_64 ---> Package sipxhomer-config.x86_64 0:15.12-4124.49451 will be erased ---> Package sipxhomer-proxyplugin.x86_64 0:15.12-4124.49451 will be erased --> Running transaction check ---> Package sipxecs.x86_64 0:15.12-7798.ee34c will be erased --> Finished Dependency Resolution Dependencies Resolved == Package Arch Version Repository Size == Removing: homer x86_64 3.2.5-2.el6 @openuc 1.6 M sipxhomer x86_64 15.12-4124.49451 @openuc 449 k sipxhomer-config x86_64 15.12-4124.49451 @openuc 18 k sipxhomer-proxyplugin x86_64 15.12-4124.49451 @openuc 389 k Removing for dependencies: sipxecs x86_64 15.12-7798.ee34c @openuc 0.0 Transaction Summary == Remove 5 Package(s) Installed size: 2.4 M If you proceed with the removal of *homer, then attempt to reinstall sipxecs, *homer is pulled back in.		Config