sipXcom 17.08

July 28, 2017

Summary

eZuce is pleased to announce the General Availability of sipXcom 17.08.

We've been busy since 17.04! While we've been working hard on the next generation 'Docker-ized' version sipXcom (see 17.08.docker branch) we've added quite a large collection of new features and enhancements for the 17.08 release. These new features include SIP Proxy congestion management tools and configuration support for some new phones. The new user portal continues to get additional enhancements including the ability for users to upload their greetings as MP3 or WAV files.

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

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

In all 45 issues (enhancements / fixes) are addressed for sipXcom in this beta release.

In a break from our releases every 4 months, we're planning on a 17.10 release as our next release. This will be a release focused on a new option for anchoring calls in freeswitch.

Highlights

sipXcom New Features:

- Zoiper Provisioning Support
- G.729 Codec included
- Optional Retry-After header in Proxy 503 Responses when Overloaded
- Proxy Congestion Management feature
- Grandstream 2130, 2140 and 2160 Phone Templates
- Allow MP3 or WAV User Greeting Upload in New User Portal User Settings

sipXcom Improvements:

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

Notes

1. Full Release Notes with installation information are located here: http://wiki.sipxcom.org/display/sipXcom/sipXcom+17.08

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

Specific Issues Addressed

Specific issues can be located in the detailed release notes in the wiki at: http://wiki.sipxcom.org/display/sipXcom/sipXcom+17.08

New Installs

A new ISO is available for 17.08 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 - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/17.08/CentOS_$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

Issues Resolved

	JIRA name	RN Content	Enhancement/Fix/Known Issue	Keywords
SIPX-159	Yealink auto-provisioning	An administrator would like Yealink phones to auto-provision in the same manner as Polycom phones.	Enhancement	Yealink
SIPX-573	Grandstream profile fails if advanced parameter is used	Fixed an issue generating Grandstream phone profiles if an advanced parameter was used. Steps to reproduce: 1. devices -> phones -> create new, grandstream gxp2010 2. Under custom configuration -> show advanced, add a P Value of P2910=1, apply 3. send profiles Expected: phone profile to be created Actual: phone profile fails to be created	Fix	Grandstream

SIPX-604	Upgrading sipxgrandstream doesn't delete all cfg files	Fixed an issue caused when installing sipxgrandstream 17.08 where some config files are left in the config directory.	Fix	Grandstream
SIPX-608	Polycom UseTelUriAsLabel improvement	An administrator would like to allow the use of the Polycom Firmware parameter for UseTelUriAsLabel. This parameter was limited to 4.1.8 template. This parameter is now available in newer Polycom Firmware >= 5.3.1.	Enhancement	Polycom
SIPX-611	Yealink Power Saving	An administrator would like to be able to configure the power saving settings for Yealink phones. This should be available for Firmware 8.x and later.	Enhancement	Yealink
SIPX-612	Yealink CDP Provision	An administrator would like to be able to configure CDP (Cisco Discovery Protocol) parameters in the Yealink Phone Configuration Plugin static.network.cdp.enable (default 0) static.network.cdp.packet_interval (default 60)	Enhancement	Yealink
SIPX-613	Settings evaluation with regular expression	A developer would like to be able to utilize regular expressions in provisioning plugins. This enhancement will make regular expressions available for all settings evaluation and support the old writing with || Current Settings Evaluation utilizes compare and supports optional settings separated by || Some Provisioning Plugins are already using Regular Expressions to do this.	Enhancement	Polycom sipXconfig
SIPX-615	Polycom Firmware 5.x.x	An administrator would like to be able to specify Polycom Firmware up to version 5.5.2.	Enhancement	Polycom
SIPX-616	Yealink CP-Source Parameter	Enhancement to allow a Yealink phone to use the Contact Header to display the calling party source. In Yealink 8x (x.80.250.x) the parameter cp_source has been improved. Now phones can use the Contact Header to display the calling party source. This is useful if you want to show the callee number after pickup or from hunt groups.	Enhancement	Yealink
SIPX-617	Make phone line label configurable	An administrator would like to control a phone or phone group to either have the phone line label left null in the Polycom phone configuration or have the phone configuration populated with the line number. UC-3903 implemented a change for Polycom phone line labels in reaction to Polycom changing the default behavior of their firmware. In 4.x and earlier firmware if the line label was blank, the line extension was used. In 5.x and later if the line label was blank the name associated with the line was used. Our change for UC-3903 looks at the database settings for a phone and if that is null or empty we put the line number into the phone configuration to defeat Polycom's change. This should continue to be the default behavior.	Enhancement	Polycom
SIPX-618	Polycom config parameter for new Polycom phones	Fix for some parameters in the phone provisioning that are model based. Example: Background of VVX 500 phone has specific parameter and for VVX 501 its the same. Current checks ignore the 501's	Fix	Polycom
SIPX-620	Jitsi preferred transport	A fix to correct parameter name in Jitsi provisioning plugin. PREFFERED_TRANSPORT -> PREFERRED_TRANSPORT	Fix	Jitsi
SIPX-622	Add opensource G729 codec to Freeswitch	An administrator would like to be able to enable/disable the use of the belladonna communications open source version of G729 codec in Freeswitch. The g.729 protocol is now unencumbered and customers are free to use this. https://github.com/xadhoom/mod_bcg729 To Enable: 1. Login to Linux box as root and execute: "yum install mod_bcg729" 2. In Media Settings page select the G729 codec driver 3. Send profile to Server, Freeswitch configuration should pickup latest changes	Enhancement	g729

SIPX-624	Yealink Resource List Subscription	Yealink plugin should check if a phone line has BLFs before setting the resource list subscription URI	Enhancement	Yealink
SIPX-625	Polycom Call Waiting	Add parameter to configure call.callWaiting.enable and call.callWaiting.ring for Polycom phones	Enhancement	Polycom
SIPX-626	Polycom RealPresence Trio POE configuration	Add parameter to configure POE for LAN out port and USB charging on Polycom RealPresence Trio poe.pse.enabled usb.charging.enabled poe.pse.class	Enhancement	Polycom
SIPX-627	Polycom RealPresence Trio Firmware	The Polycom RealPresence Trio uses a special firmware revision. This is an Enhancement to add a seperate version tree for Trio to seperate Trio Firmware from other Polycom phone firmware. Example: FW 5.4.0 for VVX/SoundPoint/SoundStation/... FW 5.4.0_Trio for RealPresence Trio Trio starts with Version Number 5.4.0. The latest current release (Mar 31, 2017) is 5.4.6	Enhancement	Polycom
SIPX-628	~~sipXprovision user has to much permission by default	The sipXprovision user which is used for auto provisioning has to much permission. To reduce load to MWI in Polycom provisioning there is an additional check if a user has voicemail permission. If they don't have voicemail permission, the phone isn't provisioned to subscribe to voicemail. This was added for Yealink as well. As the autoprovisioning feature for Yealink was added a developer noticed that the voicemail settings are provisioned. After some research it was determined that the provisioning special user has all default dialing permissions which isn't necessary. All dial permissions for the provisioning user are now disabled.	Fix	sipxconfig
SIPX-638	sipxcom 17.04 social links not in footer	The social links are not displayed in the footer within any tab beneath System -> Servers, or Diagnostics -> CDRs	Fix	sipxconfig
SIPX-639	Users invited to conference rooms use only g711u	In europe g711a-Law is mostly used so some ISPs only accept g711a but not g711u. If a non-encoding SBC is used and a user is invited to a conference room via Web Portal, than the config server produces a Invite with u-law only. Some of the bigger ISPs in Germany refuse such requests with "not acceptable here" (with a cause of no supported codec). Look for the configured codecs of media services and use them only but as workaround it would be sufficient to add a-law with lower priority.	Enhancement	Conference
SIPX-640	Polycom Device Base profile	Add a new config parameter for Polycom devices device.baseProfile Values are: Generic, Lync Why this is necessary: Normaly the default value is "null", so Generic is choosen. If you buy a phone which is marked as "Lync", this is automatically choosen. We need to ensure that "Generic" is configured to get those phones working with SipXcom/UniteMe Parameter available in Firmware 5.3.x and later.	Enhancement	Polycom
SIPX-641	Grandstream GXP 2160 phone profile XML errors	Fix for 2 errors in the xml config file that is created for Grandstream GXP 2160 phones. <!--OpenVPN Client Key --> <#P9904></#P9904> <!--Prefix for dialing password 0 --> <26049></26049> This should be : <!--OpenVPN Client Key --> <P9904></P9904> <!--Prefix for dialing password 0 --> <P26049></P26049>	Fix	Grandstream

SIPX-646	Yealink BLF 7x vs 8x	Fix difference between 7x and 8x for BLF configuration	Fix	Yealink
SIPX-648	Yealink Play Local DTMF Tone	Enhancement to add a parameter to enable/disable local playback of DTMF tones for Yealink phones. Parameter is: features.play_local_dtmf_tone_enable Currently there is parameter with this description. This is not correct. From current property file: features.AUDIO.features.partition_tone.label=Play Local DTMF Tone	Enhancement	Yealink
SIPX-72	Add support for Grandstream GXP 2130, 2140 and 2160	Enhancement to add support for Grandstream GXP 2130, 2140 and 2160 phones.	Enhancement	Grandstream
UC-3669	Feature request - Allow greetings to be uploaded from user portal	A users would like to have the ability to upload pre-recorded greetings in WAV or MP3 via the unite web interface.	Enhancement	UniteWeb voicemail
UC-4360	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to change the port that SIP Redirector is bound to in sipXconfig. This is config part of the work for UC-4361: Extend SIP Registrar configuration for pickup redirector plugin bind port with port 5085 as the default. In registrar config file (/etc/sipxpbx/registrar-config) it should look: SIP_REDIRECT.100-PICKUP.BIND_PORT : 5085	Enhancement	SIP sipxconfig
UC-4361	Make SipRedirectorPickUp bind port configurable	An administrator would like to be able to configure the port that SIP Redirector binds to for Call Pickup. This is registrar part: Get value of SIP_REDIRECT.100-PICKUP.BIND_PORT and use it during initialisation of SipRedirectorPickUp. If no value present in config file use current PORT_DEFAULT value and system will select port on it's own.	Enhancement	SIP
UC-4383	Active CDR Page broken when umlauts are used	Fixed an issue with the Active CDR page in Admin GUI where when a User Agent has umlauts in the name, sipXconfig would not display the call detail properly. The problem is on sipXconfig side. If the UA encodes special characters the problem doesn't reproduce. Polycom seems it is not encoding and SIPXCDR API such that config calls to retrieve active calls works as expected. There seems to be a problem on sipxconfig side which may not be handling them properly.	Fix	CDR sipxconfig
UC-4397	Group firmware gets reset after upgrade	Fixed an issue with Group Firmware being reset to default when upgrading a system to 16.12. After the upgrade, in the Device > Phone Groups > any group > any phone, the Group Firmware version was reset to default.	Fix	sipXconfig
UC-4398	VVX 1500 profile voice codec typo	Fixed an issue with the Polycom VVX 1500 configuration plugin. There was a typo in the voice codec settings.	Fix	Polycom

UC-4421	Default behavior for blind transfer on Polycom FW > 5.3.0	Due to changes in Polycom firmware, the old functionality for Blind transfer feature performed by hitting Transfer and then the Blind softkey is now gone. Users must now press and hold the Transfer softkey button, push down, push select "Blind Transfer" and then input their destination. If "Blind Transfer" is used more often than "Consultative Transfer" this can be set as the default option for transfer type in Preferences > Additional preferences > Default Transfer Type page in the Polycom configuration page. Also, the procedure used in the pre 5.3 versions to create a softkey with a macro(efk) to do one-button blind transfers seems to no longer work.	Enhancement	Polycom
UC-4435	Add Retry-After in 503 proxy response	An Administrator would like to be able to have UA's sending requests to proxy be able to back off if the proxy is overloaded. As determined by RFC 3261 - https://tools.ietf.org/html/rfc3261#page-176 and RFC 5390 - https://tools.ietf.org/html/rfc5390, in the 503 reply, we have added an optional Retry-After header such that phones that support this RFC will know when to retry. We currently don't do that, this is how the message looks at the moment: OUTGOING.. ...nSIP/2.0 503 Queue Size Is Too High From: sipxtest <sip:sipxtest@10.2.0.174>;tag=23289SIPpTag001683 To: 1238 <sip:1238@cluster.ezuce.ro>;tag=fzD_qG\r\n Call-Id: 1683-23289@10.2.0.174\r\n Cseq: 1 INVITE\r\nVia: SIP/2.0/TCP 10.2.0.174:8100;branch=z9hG4bK-23289-1683-1\r\n Date: Wed, 03 May 2017 13:39:44 GMT\r\n Content-Length: 0\r\n\r\n --------------------END	Enhancement	sipxproxy
UC-4436	Decrease default number of backups to retain	Currently the default number of backups is set to 50. If an administrator doesn't pay attention to that field they can quickly run out of space. Decrease this number to 5 by default.	Enhancement	backup sipxconfig
UC-4437	Phonebook REST improvements	An administrator would like to add entry-id endpoint to the my/phonebook rest call. This should return all entryId's so you should not be forced to get them from psql queries /my/phonebook/entry/{entryId}	Enhancement	REST
UC-4439	Add config support for Zoiper soft phone	A customer would like to introduce a soft phone client into their environment that is usable on a myriad of different operating systems. It has been determined that the most appropriate softphone for their purposes is Zoiper since it will work on linux, MAC, Windows, Android, etc. The customer worked with Zoiper to identify changes that will be required on their side and eZuce met with Zoiper to better understand how to best implement a provisioning model for their softphones. The result of that meeting was that an XML configuration file would be generated with all of the configuration variables in it. This XML file will then be downloaded and applied by the Zoiper client based on a configuration server value being set to our Uniteme cluster. The contents of this XML file have already been discussed and there is a sample XML file already available for review.	Enhancement	Zoiper
UC-4450	Implement proxy congestion protection - Config Work	Add new sipxproxy configuration options for the Administrator to be able to control which congestion policy is used and what the SIP Proxy Message Queue size will be. The administrator will be able to select a congestion policy: SIPX_PROXY_CONGESTION_POLICY : with default value : "SERVICE_UNAVAILABLE". Another possible value is "IGNORE" Also, add another option for the sipxproxy queue size (this is not a new setting but was not extended to web interface): SIPX_PROXY_QUEUE_SIZE:default 1024 Options to be placed under System -> SIP Proxy -> Advanced in the area of Reject calls on filled queue options.	Enhancement	sipxconfig sipxproxy
UC-4451	Add Retry-After in 503 proxy response - config work	An administrator would like to be able to allow the SIP Proxy to insert a retry-after header in replies when the SIP Proxy service becomes overloaded. This is config side work for UC-4435. The new setting is: SIPX_PROXY_RETRY_AFTER: default to 60. The new setting is located in System -> SIP Proxy -> Advanced. The purpose of this new option is to give the Administrator another method to help avoid proxy congestion. UA's that understand RFC 5390 (https://tools.ietf.org/html/rfc5390) will back off and retry after the set amount of time.	Enhancement	sipxconfig sipxproxy

UC-4472	Proxy Segfault caused by a malformed request URI	Fixed a problem caused by malformed SUBSCRIBE message that caused sipxproxy to crash. Core file analysis shows that reason for the crash was the first character of a SUBSCRIBE message was a space. This was being caused by an ALG at a remote site adding a space into SUBSCRIBE messages. This remote site had a cable modem where the ALG could not be disabled. The fix will ignore these malformed messages.	Fix	sipxproxy
UC-4481	Greeting REST API which returns filenames of different greetings.	An administrator would like a new rest API call which returns the filename for the standard, outofoffice and extendedabs greeting types.	Enhancement	REST
UW-297	Voicemail download does not work in Chrome with Unite Lite	Fixed an issue when trying to download a VM when logged in Unite Lite via Chrome on any OS. The download simply would not work. Valid only for Chrome and Unite Lite.	Fix	UniteWeb
UW-380	Upload MoH files in Unite Web	Fixed an issue with an error in the browser when up MoH files from Firefox and Chrome. Also addressed the issue of uploading either WAV or MP3. Both are allowed by MoH service.	Enhancement	UniteWeb
UW-389	Grey out or remove edit features when user group speed dials are selected	Fixed an issue where if a user goes to their speed dials and the "Use group speed dials" box enabled, the speed dials were not grayed out. The user might also think that he can add more speed dials and edit the ones he sees. If the user were to do that however, as soon as he would save the changes, they would not get saved and will instead refresh to show you the inherited group speed dials. While this behavior is correct, the user might not understand why he can't edit/add speed dials, so a solution might be to see the speed dials in "read-only" mode.	Fix	UniteWeb
UW-392	Display calling number and caller-id in voicemail page.	A user would like to be able to see the caller-id (name) and also the phone number that left the message. This number should also be clickable for click to call.	Enhancement	UniteWeb
UW-396	Make warning about closing UniteWeb configureable	An administrator would like the user to be able to have the warning when UniteWeb is closing optional on a per-user basis. The default value should be enabled.	Enhancement	UniteWeb