sipXcom 18.08

August 29, 2018

Summary

eZuce is pleased to announce the General Availability of sipXcom 18.08.

Important - Do not directly upgrade to 18.08 from any version before 18.04. The 18.04 upgrade must be completed first. Please read the release notes for 18.04 if you are not on 18.04 yet.

The two major features in this release are the Admin GUI Menu restructuring and the addition of the ability to age voicemail out of the system by user group.

The Admin GUI's menus have been updated to be sorted alphabetically and with some sub-menus. The menus have always been built dynamically based on which services were enabled. This lead to the system menu becoming as long as it was and menu items not being sorted. Well, we decided to address these issues and bring some structure to the menus. Hopefully, new users of the system will find things more easily and hopefully, our long-term admins adjust in a timely manner.

Highlights

sipXcom New Features:

- Automatic Voicemail purging by User Group

sipXcom Improvements:

- Admin GUI Restructuring
- Set file names for Network Packet Capture Service so they end in .pcap when downloaded.

Notes

1. Full Release Notes with installation information are located here: 18.08 Full Release Notes
2. The regular release of 18.08 will continue to use the upgrade script we started using in 18.04.

Who Should Install?

This release is recommended for all 4.6 and later installations. If you have a patch installed to your system a new patch may be required.

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

Questions

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

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

Specific Issues Addressed

Jira #	JIRA Name	RN Content	Enhancement /Fix /Known Issue	Keywords
SIPX-25	Cannot get BLA to work in release 14.10	When an incoming internal or external call is placed to the main line on a 14.10 system and one phone answers, there is no solid red light (BLA) that appears on any other phone. This configuration works as designed on 14.04 and lower systems. Fixed firewall settings to allow this traffic through by default.	Fix	bla

SIPX-403	P-Asserted-Identity of Proxy - SIP Side	Proxy must pass p-asserted-identity (default should be to not pass (as it operates now)) - proxy code change	Enhancement	sipxproxy
SIPX-404	P-Preferred-Identity of Proxy - gui work	Create a checkbox in proxy setup to pass p-asserted-identity (default should be to not pass (as it operates now): GUI change.	Enhancement	AdminPortal sipxproxy
SIPX-479	Live Attendant not updated in Mongo properly	When an AA is defined as live AA a new entry is added in MongoDB so registrar will look up routing information from there. When an existing live AA is disabled, MongoDB entry is removed and routing is done according to mappingrules.xml. In this case customer previously had 1900 configured as live AA, then disabled it. However it seems that in this version the mongo entry does not get cleaned up when doing this action and leaves inconsistent entries in mongo: { "_id" : "AttendantRule9", "ident" : "1900@DOMAIN", "uid" : false, "cnt" : "sip:1900@DOMAIN", "locations" : [], "vld" : true, "ent" : "attendantrule", "als" : [{ "id" : "1900", "cnt" : "<sip:1600@DOMAIN;sipx-noroute=Voicemail?expires=50>;q=0.933;sipx-ValidTime=\"88e:a8c:e2e:102c:13ce:15cc:196e:1b6c:1f0e:210c\"", "rln" : "userforward" }, { "id" : "9673", "cnt" : "sip:1900@DOMAIN", "rln" : "alias" }] } (other ways for this bug to manifest is generating null contacts as in: "cnt" : "<sip:null@DOMAIN;sipx-noroute=Voicemail;sipx-userforward=false?expires=0>;q=0.933",) Workaround: mongo use imdb db.entity.remove({"_id" : "AttendantRule9"}) Mind that the rule will get repopulated in Mongo when sending profiles to server	Fix	Auto-Attendant
SIPX-665	Duplicate Syslog Messages in /var/sipxpbx/phonelog/*	The rsyslog configuration is duplicating entries in /var/log/sipxpbx/phonelog/* logs. After examining the rsyslog config in /etc/rsyslog.conf, we discovered that this is because the phonelog.conf file is being included twice; once towards the top of the file when the entire rsyslog.d directory is included (line 31), and then again at the end (line 81). We modified the CFEngine scripts to prevent phonelog.conf from being included twice as a workaround.	Fix	logging
SIPX-741	Change file name extension for Network Packet Capture Service	Network Packet Capture Service generates file names as follows: filename.pcap0 filename.pcap1 filename.pcap2 ... This requires that an administrator rename each file to .pcap after download to open properly with Wireshark. An administrator would like to change the file names to: filename0.pcap filename1.pcap filename2.pcap ...	Enhancement	pcap
SIPX-744	sipxlang-pt.jar schema within spring bean xml uses the wrong classpath	If the Portuguese language is installed on the server, the system does not work without internet.	Fix	language
SIPX-746	Help text should be added to indicate that voting can be disabled only when priority is 0	In the Uniteme Admin GUI -> System -> Database -> Global database a help text should be added on the right after the last paragraph: "Voting - If you do not want a particular database to participate in the voting process, you can disable this feature for each database. A cluster is limited to 7 voting servers and 12 database servers in total." Which should be completed with the phrase: "Voting can be disabled only if priority is set to 0 first." Otherwise, the "disable voting" command does nothing, and the user can find what the error is only by looking manually at Mongo logs. The error in mongo.log reads: [js] replSetReconfig got BadValue: priority must be 0 when non-voting (votes:0) while validating More info on this: https://docs.mongodb.com/manual/tutorial/configure-a-non-voting-replica-set-member/	Fix	AdminPortal databases
SIPX-749	Menu Restructure	Reorganization of Uniteme / sipXcom menus is needed to simplify the interface.	Enhancement	AdminPortal
SIPX-750	User/Group based Voicemail cleanup	Extend voicemail_clean script to cleanup voicemails using days, path as parameters. Use this to implement User/Group based Voicemail cleanup in open-source sipXcom	Enhancement	Voicemail
SIPX-759	Jitsi preferred transport velocity file	There is an error in the Jitsi velocity file with the Preferred_Transport parameter. The correct parameter name in velocity file is: PREFFERED_TRANSPORT -> PREFERRED_TRANSPORT	Fix	Jitsi
UC-308	Add MoH timeout	There is a possibility that FS can miss a BYE or CANCEL and MoH will be played indefinitely and will eventually cause the system to crash. Administrators would like to add a timeout option for MoH channels. https://freeswitch.org/confluence/display/FREESWITCH/mod_local_stream#mod_local_stream-TimeoutExample	Enhancement	MoH

UC-4724	The user is unable to save prerecorded greeting unless menu is toggled	Upon testing, we found that the new Uniteme interface has a bug in the User portal -> Settings -> User settings. The drop-down menu for the Voicemail->Announcement selection does not have the "Name" field. Furthermore, the SAVE button at the bottom of the page is inactive until this drop-down menu is toggled. Looks like a webpage glitch which prevents the upload of prerecorded greetings if the drop-down menu is not toggled in any way.	Fix	Unite Web
UC-4729	g729 in 'selected' codecs by default	With g.729 added in the system by default, G.729 is in the 'selected' codecs. It should be set as 'available' by default.	Fix	g729