sipXcom 15.06

Summary

This is a small update for sipXcom. After 15.06 we're going to move to 15.08 and try a 2 month cadence for releases. One month is a little too quick for us, while removing work on the QA team we added load to the build team. We'll see how a 2 month cycle works for a couple releases.

New Features / Improvements:

New Feature - SIPX-128 - Authentication Rules should not authenticate calls to registered users from an SBC or gateway

New Feature - SIPX-17 - Create proxy plugin to validate URI/HOST/IPAddress field entries

New Feature - SIPX-2 - Proxy PlugIn: Advice of charge (AOC)

Improvement - XX-11572 - Voicemail: download with file extension in new user Portal

Improvement - UC-445 - Set password for postgresql user postgres

Improvement - SIPX-125 - Voicemail: add header when downloading

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 15.06 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 - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/15.06/CentOS_$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 Addressed

JIRA ID	JIRA name	RN Content	Enhancement/Fix/Known Issue

UC-3521 SIPX-177	Fax'ing Doesn't work through Sangoma SBC	Fixed an issue that caused faxing to not work through Sangoma SBC. During the sip signaling process for fax, Sangoma changes a port and FS does not follow this port change. Instead, in continues sending media to old port. The fix is to add the sofia template parameter as described below. <param name="disable-rtp-auto-adjust" value="true"/>	Bug
SIPX-130	Openfire and RLS doesn't subscribe to sqa when sqa restarts	Fixed an issue when restarting State Queue Agent (sqa) which resulted in: - openfire doesn't attempt to resubscribe, the only way to make it subscribe is to restart it - RLS subscribes after 8 minutes	Bug
SIPX-127	System audit sorting by User column triggers exception	Fixed an issue with the new Audit feature where if the administrator tried to sort the audit by the User column the page would crash.	Bug
SIPX-126	Misleading INFO log in sipregistry about contacts' Expires period	Fixed a logging issue where the INFO log messages for sipregistrar were misleading because it looks like the time until that binding will expire is very large; in fact, that Expires refers to the time in seconds since epoch until the binding is available.	Bug
SIPX-119	sipXproxy should respond with 503 for server failures in HA setups	Fixed an issue to help the phones to do failover to the next available proxy in cases when the current proxy is unable to handle the request due to internal issues (i.e. mongo timeout, dns lookup failures), the proxy should respond with the 503 error code - currently it uses the 500 code for this kind of situations. See: https://tools.ietf.org/html/rfc3261#section-21.5.4	Bug
SIPX-55	Both PCMA and PCMU are marked as G711 in the Yealink profile	Fixed an issue in Yaealink template where Under Device->Phones->"Yealink phone"->Line->Codecs there is no way of telling which codec is G711 U-law and which is A-law since both are named G711	Bug
SIPX-38	Start syslog with reverse lookup disabled	Fixed an issue where phone logging causes Syslog to flood named with reverse lookups. These fail because we don't create reverse lookup DNS zones. We changed syslog start parameters to disable reverse lookup.	Bug
SIPX-34	SipMessage::getContactEntry() fails to parse contacts if user part of the uri contains comma characters	Fixed an issue that caused parsing of contacts to fail if there was a ',' in the User part of the URI.	Bug
SIPX-29	Freeswitch pid file and xml file in /var/log/sipxpbx folder	Fixed an issue where freeswitch.pid and freeswitch.xml.fsxml should not be in this folder : /var/log/sipxpbx/ Related to SIPX-129	Bug
SIPX-24	Label change for mediaservices codec support	Change text on list of available codecs to reference sipxcom wiki for g.729 info.	Bug
SIPX-21	make setup.sh will generate a setup.sh which installs too much	Accepted a patch from "Niek Niek" to fix sipxecs.spec.in which required all the sipx projects without a capital x. That caused the exclude in the mak/list-dependencies.mk (build/mak/20-list-dependencies.mk) to not work.	Bug
UC-3574	missing /etc/sipxecs.cfg causes snapshot creation error and local backup failure	Fixed an issue where if /etc/sipxecs.cfg was missing then there were issues creating snapshots and local backups would fail.	Bug
UC-3569 SIPX-176	CSV export from 14.04, import on 14.10 caused sipxconfig crash	Fixed an issue such that with LDAP enabled, and then users imported (through CSV not LDAP), the csv import caused a sipxconfig exception related to system audit. After disabling the system audit, the import ran successfully. The issue was related to a system audit setting that was not present in 14.04.	Bug
SIPX-129	Freeswitch pid file in /var/log/sipxpbx/	Fixed an issue with freeswitch in 14.10 where it was creating a pid file in /var/log/sipxpbx/ instead of the expected /var/run/sipxpbx/	Bug

XX-11572	Voicemail: download with file extension	Changed the voicemail files to download with an extension so that they appear properly in browsers	Improvement
UC-445	Set password for postgresql user postgres	Added the ability for the administrator to set the postgresql user. If the administrator disabled the firewall in the system, postgresql was able to be queried without a password.	Improvement
SIPX-125	Voicemail: add header when downloading	In order for browsers to not follow a voicemail download url the following header needs to be present in the HTTP response: Content-Disposition: attachment This way the browser will download the voicemail. Added new url for voicemail download (instead of inbox write download): - https://<fqdn>/sipxconfig/rest/my/redirect/media/201/download/100000001.mp3 - force the file to download - https://<fqdn>/sipxconfig/rest/my/redirect/media/201/inbox/100000001.mp3 - navigate to a new window which plays the file Also keep in mind that in both cases you can use just the id of the message (100000001) or the id with the file extension (100000001.mp3)	Improvement
SIPX-128	Authentication Rules should not authenticate calls to registered users from an SBC or gateway	Fixed an issue where if an external SBC handles remote worker registrations and it is also configured as a gateway with permission, calls to registered users will be authenticated. This would fail if the call is coming from the PSTN.	New Feature
SIPX-17	Create proxy plugin to validate URI /HOST/IPAddress field entries	Create a new proxy plugin to validate URI/HOST/IPAddress field entries in SIP messaging. OSS_CORE has a very nice ABNF validator implementation. it can validate the syntax of anything that has a known ABNF rule. Use this code as a plugin to check the Request-URI, contact-URI and from-URI which we all use in routing SIP messages and reject messages with BAD syntax with a 400 Bad Request.	New Feature
SIPX-2	Proxy PlugIn: Advice of charge (AOC)	Accepted a new feature from IANT that is a plugIn for the sipXproxy to parse AOC Information and store it in the MongoDB. AOC Information is provided by the ISDN provider during / at the end of the call. The AOC comes to the Proxy inside of INFO Messages or responses (1xx/2xx) to ACK or BYE with Content Type "application/vnd.etsi.aoc+xml" and is specified by ETSI TS 183 047. If the PlugIn is installed, the Proxy will load it on startup and there is no config for it. It checks every incomming message if it is a valid SIP Message for AOC with an AOC Content Type. The Amount will be parsed out of the Message and be stored into MongoDB (Database iant, Collection iant_billing) mapped with the Call ID. The Messages will not be redirected, the PlugIn just scans the content and will not modify any part of the SIP Message. A present this new plugin must be installed with: yum install sipxaocbilling	New Feature