sipXcom 15.10

Summary

New Features:

- Locations Feature to Block Calling between Locations
- Call Back on Busy now Highly Available
- Set Polycom Background Images in Admin GUI
- Logging of Authorization Codes to Call Detail Report
- Provision Jitsi from Admin GUI
- Obfuscate CDR Records

Enhancements:

- New eZuce Logo on Polycom Phones
- Audit Log Performance Enhancements
- Configure Outbound Port for Exchange Voicemail
- Update of Freeswitch to 1.4.20
- Included VP8 Codec in Media Services
- Yealink phone enhancements / fixes
- Polycom phone enhancements / fixes
- Allow for Polycom firmware up to version 5.4.0 (still recommending SoundPointIP at 4.0.7 and VVX at 4.1.7 however)

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 15.10 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 - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/15.10/CentOS_$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 name	RN Content	Enhancement/Fix/Known Issue	Keywords

SIPX-142	Make TCP default transport for Polycom phones	In many cases recently it has become apparent that continuing to use UDP for SIP signalling with Polycom phones hasn't really worked: * Server Failover in HA environments * Large BLF lists TCP is also inherently more reliable which results in a more consistent customer experience. This ticket is for two configuration items: * Make TCP the default transport protocol for Polycom Phones * Enable TCP Fast Failover by default	Enhancement	Polycom
SIPX-167	Disable ipv6 using cfengine promise	Fixed an issue with RPM installations to disable IPV6. This was being done properly for ISO installations. Ensure the existence of: net.ipv6.conf.all.disable_ipv6 = 1 net.ipv6.conf.default.disable_ipv6 = 1 is within /etc/sysctl.conf until we have completed ipv6 support.	Enhancement	Config
SIPX-226	Option to enable Core Dumps for Freeswitch	Added a new option in System -> Media Services -> Advanced Settings to enable core dumps. in /etc/sysctl.conf add fs.suid_dumpable=2 kernel.core_pattern=/tmp/core then /sbin/sysctl -p /etc/sysctl.conf verify properly set [root@openuc ~]# sysctl fs.suid_dumpable fs.suid_dumpable = 2 In /etc/init.d/sipxfreeswitch add -core switch when starting up, e.g. DAEMON_START_ARGS="-nc -nonat -core \ -u sipx \ -g sipx \ $DAEMON_ARGS" then restart freeswitch service as service sipxfreeswitch restart. Force core dump as: fs_cli freeswitch@internal> fsctl crash Socket interrupted, bye! Core file generated as [root@openuc ~]# ls -la /tmp/core.20319 -rw------- 1 sipx sipx 28315648 Jul 21 04:06 /tmp/core.20319 (pattern is core.PID) To see what did the crash (mind that freeswitch-debuginfo RPM should be installed and same version as freeswitch): gdb freeswitch /tmp/core.20319 then type bt, you will get something as (gdb) bt #0 0x00007fe36c705625 in raise () from /lib64/libc.so.6 #1 0x00007fe36c706e05 in abort () from /lib64/libc.so.6 #2 0x00007fe36ed59c61 in switch_core_session_ctl (cmd=SCSC_CRASH, val=0x0) at src/switch_core.c:2506 #3 0x00007fe35a3d49c5 in ctl_function (cmd=<value optimized out>, session=<value optimized out>, stream=0x7fe34943cae0) at mod_commands.c:2219 #4 0x00007fe36ed873de in switch_api_execute (cmd=0x7fe34c00bd80 "fsctl", arg=0x7fe34c00bd86 "crash", session=0x0, stream=0x7fe34943cae0) at src/switch_loadable_module.c:2438 #5 0x00007fe36ed3153b in switch_console_execute (xcmd=<value optimized out>, rec=0, istream=0x7fe34943cae0) at src/switch_console.c:391 #6 0x00007fe35b999a75 in api_exec (thread=<value optimized out>, obj=0x7fe34943d470) at mod_event_socket.c:1503 #7 0x00007fe35b99b4cd in parse_command (listener=<value optimized out>, event=0x7fe34943d758, reply=0x7fe34943db70 "", reply_len=512) at mod_event_socket.c:2253 #8 0x00007fe35b99d9b7 in listener_run (thread=<value optimized out>, obj=0x7fe36000b208) at mod_event_socket.c:2677 #9 0x00007fe36d0bd9d1 in start_thread () from /lib64/libpthread.so.0 #10 0x00007fe36c7bb8fd in clone () from /lib64/libc.so.6 or bt full thread apply all bt thread apply all bt full	Enhancement	Config, Media
SIPX-228	Provision Jitsi	Be able to provision Jitsi as a phone and XMPP client from within sipXcom. In phone and phone groups be able to create a Jitsi phone and associate a user with it. Enable the ability to add XMPP capability if the user has sipXcom XMPP configured. Configuration should work much like existing Bria configuration capability. https://jitsi.org/Documentation/ConfigureJitsiWithProvisioning Note: Jitsi doesn't seem to go to the user inbox properly to check VM, Dial 101 as workaround (see jira.sipxcom.org/browse/SIPX-375)	Enhancement	Jitsi
SIPX-239	Upgrade FS to latest version	Upgrade FS to latest version (1.4.20)	Enhancement	Config, Media
SIPX-242	Custom gateway settings :: CallerID, do not alter name, just number	Contribution to just keep the "Display Name" portion of the From header when transforming extensions. We could potentially check the User Entity and Gateway Entity for caller ID name before using the Display Name in the original From header, but I don't think that's necessary since this will most likely only be used when 1) Sending user info to another PBX 2) Mapping users to a block of DIDs owned, in which case the name portion will probably just be ignored. I think this is better than having no name portion kept, but if you think it should check the gateway and user entity first let me know and I can change it. gniculae	Enhancement	Config, Gateway

SIPX-256	Make call back on busy HA	Make call back on busy a highly available service.	Enhancement	Config
SIPX-267	Change System -> Branches to System -> Locations	As part of the new Locations feature, change the Branches menu item to be Locations.	Enhancement	Locations
SIPX-269	In Locations (Branches) page add Calling Restrictions	Added the ability to restrict calling based on a user's defined Location. The Administrator would like to be able to configure which other Locations users in a particular Location are allowed to call. The Administrator would like to be able to control what features users from other Locations are allowed to access for a particular Location (i.e., Auto Attendants, Hunt Groups, Intercom, Paging, Conference Bridge, Call Queue, Park Orbits, Authorization Codes).	Enhancement	Locations
SIPX-275	Polycom Configuration improvements	Added some new configuration features to the Polycom configuration plugin. - Added ACD Agent login-logout - Added admin and user password configuration - Added labels for Polycom firmware 5.2.1, 5.2.2, 5.2.3, 5.3.0, 5.3.1, 5.4.0 - Added configuration for local processing of DND and FWD, when server-based feature control is active - Making device backgrounds editable (SoundPoint and VVX series) with enable/disable feature Tested with 15.10 and VVX 300/310/410/500	Enhancement	Polycom
SIPX-285	Add option to assign location(s) for additional resources	Add enhancement to support Locations for system services. AA Auth Codes Call Park Call Queue Conferences Hunt Groups Paging others... Rename Location to Server in Call Park page	Enhancement	Locations
SIPX-287	Add fields to confirm passwords on Admin Settings and SNMP pages	Add confirmation fields for default password,default PIN and default psql password on Admin Settings page and confirmation field for community string on SNMP page.	Enhancement	Config
SIPX-314	Obfuscate CDRs	Added a new feature to obfuscate (masquarade) parts of dialed phone numbers. Larger companies sometimes have concerns about the data privacy of the cdrs. Settings can be found in System -> CDR -> Advanced Settings Enable Masquerading - enable / disable the freature Minimum length of extension that muse be masqueraded - This value is the minimum length of extension that should be covered with a * Prefixes excluded from masquerading - Certain numbers prefixed with these values will show the entire dialed number and not be covered with a *	Enhancement	CDR
SIPX-318	Users in different locations are allowed to call each other by default	Users in different location should not be allowed to call each other by default. 1.Created two Locations. 2.Created two user groups with one user in each group. 3.Assigned a different location to each user group. 4.Verified in mongo that users are assigned to correct location. 5.Called between these two users Result : users can successfully call each other. Expected : they should not be able to	Enhancement	Locations
SIPX-321	Add more info on Locations page	By default calling between different locations will not be allowed. This could be important information to have in the admin gui either on the locations page or under the Add location page. Perhaps there is more relevant info that should be added..	Enhancement	Locations
SIPX-323	Yealink ACD and SIP Server Types	Added some new configuration parameters for Yealink Phones: ACD inital state on login ACD auto login after reboot SIP Server types UCAP, Genesys and Genesys Advanced	Enhancement	Yealink
SIPX-326	Add vp8 codec config for media services	Enabled the vp8 codec in media services	Enhancement	Config, Media
SIPX-329	Configuration option for Push and State Polling User, Password and Type	Added the configurable parameters for Polycom plugin so that user and password for push and polling messages could be configured. Used for remote control and tracking of phone status Tested with 15.10, Polycom VVX 500, VVX 410 Parameter on the phone WebGUI of VVX Phone > Settings > Application > Phone State WebGUI of VVX Phone > Settings > Application > Push Parameter in config gui Devices > Phone Group > Polycom VVX... > Applications Moved UCDesktop in Menu Applications	Enhancement	Polycom
SIPX-362	SIP Core related changes for Location Permissions for Inbound	Note the "loc_assoc_inbound" new field that contains the locations where the external inbound calls are permitted The old loc_assoc field is now restricted to locations where internal calls only are permitted Core code needs to be changed to ackwnowledge these new field.	Enhancement	Locations
SIPX-369	Location fallback permissions	Added an enhancement so that in the Locations panel there will be a new tab named Fallback Calling Permissions with a dropdown listing all locations. When a location is picked it will be replicated in Mongo as loc_assoc_fallback, single value. If no location selected there will be an empty value replicated as loc_assoc_fallback. The fallback permissions control call transfers.	Enhancement	Locations
SIPX-370	SIP Core - Location fallback permissions	Added an enhancement to support adding permissions to the fallback call routing in support of permissions for call forwarding.	Enhancement	Locations
SIPX-374	Make T.38 reinvite on fax receive optional	Created an option to send a T.38-ReINVITE when a fax was detected by tone detection to deal with a FreeSwitch bug that was adding additional SDP information into a dialog.	Enhancement	Config

SIPX-84	Log Accepted authorization code.	Added the ability to log the accepted Authorization Code and store in CDR.	Enhancement	CDR, Auth Code
SIPX-88	Polycom firmware 5.1.1 default administrator password	Added the ability to set the Admin user and password for Polycom phones. After upgrading to Polycom release 5.1.1 rev B a warning message is displayed to the user: "Default admin password is in use, please contact your administrator"	Enhancement	Polycom
UC-3244	zen 3420 : Desktop integration missing from VVX 1500 template	Added the desktop integration portion of the Polycom template is for the VVX 1500 template, whereas it is there under all other VVX templates.	Enhancement	Polycom
UC-3666	Feature request - ability to assign port number to exchange voicemail dial plan	Added the ability to set a port in the Voicemail dialplan when using Exchange Voicemail Server as an option. Needed the ability to change the port to 5080 when sending and not use 5060. This eliminated the need to use an SBC to communicate to Office 365 Exchange Unified Messaging	Enhancement	Config
UC-3756	Update Polycom - ezuce custom background images	Updated eZuce background images that Polycoms load on their displays. Update the ezuce logo to use the new one and change color scheme - currently background is white (this is an issue because the menu options are also written with white). VVX400 does not have custom bg (works for vvx410) VVX600 needs to have the ezuce logo differenty - currently on the main screen , you cannot see the ezuce logo because it is blocked by the black bar where softkeys reside.	Enhancement	Polycom
UC-3763	Change System -> Emergency Dialing -> Locations text box name	As part of adding the new Locations feature we changed the 'Location' text box name to read "Emergency Response Location" in Emergency Dialing so as not to cause confusion with Locations. Also modify description under text box that currently reads "Physical location details (such as room, floor, building, street address). This should be as fine grained as possible. Third party applications may use this to update the Public Safety database." to read "Physical emergency response location details (such as room, floor, building, street address). This should be as fine grained as possible. Third party applications may use this to update the Public Safety database."	Enhancement	Locations
SIPX-114	Translations in the login screen don't work	Fixed an issue with the login to the admin GUI so that it would use the default system language. When the browser default language is for example Dutch and translations for the login screen are available they won't be used in some cases, the text will still be in English.	Fix	Config
SIPX-240	Creating Snom Group results in Error screen	Fixed an issue with phone groups other than Polycom. It was reported initially that when adding a group for snom phones - snom320 in this case, but same with any of the others as well. Select snom320 from list of devices. Make a change in the group - select save - go to error screen with the following - An internal error has occurred. Click here to continue. When you go back to the group, your changes are saved in the profile.	Fix	Config
SIPX-243	Polycom Configuration Server Feature Control DND/FWD	Fixed an issue where the Polycom Plugin generates wrong Parameter in Server Feature Control for DND and Forward Parameter is serverFeatureControl.doNotDisturb.enable not serverFeatureControl.doNotDisturb.enable.enabled Same for Forwarding Related to SIPX-36	Fix	Polycom
SIPX-274	Conference maximum legs field , description update	Fixed an issue that when creating a new conference, the "Maximum legs " field is set by default to "0", and has the following description : "The maximum number of call legs to be allowed by this bridge. 0 means unlimited." Reported problem : Setting up "1" in the "Maximum legs" field will have the same effect as setting "0" - unlimited users will be able to join the conference. *Setting up "2", will permit two users to join the conference,then the 3rd one will be asked to enter PIN,and upon entering he will hear a prompt "Locking conference" and get disconnected.F	Fix	Conference
SIPX-278	MoH not working when phone behind NAT	Fixed an issue with remote phones would not get MoH. This does not replicate on setups where servers and phones are on the same network.(no NAT) Configuration: Server is installed in AWS (behind NAT) Phones are behind their own NAT. Added two set of logs. In one set, call is between two Polycom phones (fw 4.1.7) - there is no MoH. In the other set,call is between a Polycom and a Bria softphone - there is MoH only when Polycom holds the call. Fail scenario <Polycom phones>--Nat--Internet--Nat--<openUC> : 1.Polycom 10.2.0.51 - extn 2801 calls Polycom 10.2.0.52 - extn 2802 There is voice path both ways 2.2801 holds call Result Fail : 2802 does not hear Moh (also tested when 2802 was the one to hold the call,there was no moh as well,but i did not include that in the logs/capture) Semi-Fail scenario <Polycom + Bria >--Nat--Internet--Nat--<openUC> : 1.Polycom 10.2.0.51 - extn 2801 calls Bria 10.2.0.200 - extn 2804 There is voice path both ways 2.2801 holds call Result Success : 2804 hears Moh 3.2801 unholds call There is voice path both ways 4.2804 holds call Result Fail : 2801 does not hear MoH	Fix	MoH, NAT
SIPX-283	sipXcom version fails to start when installing from RPMs	Fixed an issue where cfengine fails to run due to missing local_cp_by_digest bundle	Fix	Config
SIPX-293	Configuration fails to deploy due to IVR exception	Fixed an issue with a java.lang.NullPointerException when backup server was set to a different IP.	Fix	Backup

SIPX-307	Yealink configuration crashes on apply	Fixed an issue with Version 15.06 - 15.10 that caused the Yealink Configuration to crash on apply (similar to SIPX-240) Failure occurs in "org.sipfoundry.sipxconfig.site.phone.EditPhoneDefaults.java), inside the apply Method the "DeviceVersion" throws a null-pointer exception. The DeviceVersion is null, because inside the Yealink-Plugin the a phone is tagged with "yealinkPhone" and the configuration looks for "yealink" To handle this, it is necessary to change "yealinkPhone" to "yealink" to make it work (plugin folder and bean_id) Made some changes to do this (pull request follows), but this means the bean_id will changes, so older entrys in the database will crash the web GUI. To fix this a change inside the postgres is needed, which is not part of the commit source. Tested with 15.10. SQL Statement to fix after update UPDATE "public"."phone" SET "bean_id"='yealink' WHERE ("bean_id"='yealinkPhone')	Fix	Yealink
SIPX-309	Yealink DND mode.	Fixed a problem with DND mode on Yealink phones. "Feature Key Synchronization" - must be unchecked in phone's web interface or by autoprovision fille.	Fix	Yealink
SIPX-310	Problem receiving faxes in 15.06 and later & sipXbridge Specs Violation	Fixed an issue with sipXbridge that caused certain SDP values to not be passed. This was manifested by a FS but that's adding additional SDP settings. During a T.38 re-INVITE sipXbridge strips a custom payload from the m line sent by FS. The resulting m-line has no payload at all which of course is a syntax error. This seems to be a direct result of the regression pointed out here:https://freeswitch.org/jira/browse/FS-6954. This may cause SBC's that don't pass the m-line in SDP to not be able to receive faxes.	Fix	SIP Trunk
SIPX-313	References in from settings_with_location table not removed when node is deleted	Fixed an issue that caused the settings_with_location table is not cleaned up when locations deleted	Fix	Locations
SIPX-328	Sipxcom 15.08 Install Errors using Centos minimal Method	Fixed an issue caused when user ran 'yum install sipxcom' instead of 'yum install sipxecs'. Workaround for 15.08 - use 'yum install sipxecs'. Installed Centos minimal and then installed Sipxcom using this methodhttp://wiki.sipxcom.org/display/sipXcom/Installing+on+Fedora+and+CentOS. The following errors were generated following execution of sipxecs-setup. System is unstable and cannot be used. Centos installation method is used for remote locations where transmission of ISO files is easier with Centos ISO than Sipxcom ISO because image is smaller. Application settings: Primary server : yes Host : pbx SIP Domain : sipxtest.com Network Domain : sipxtest.com Would you like to change your application settings? [enter 'y' or 'n'] : n Initial setup, this may take a few minutes... error reading information on service vsftpd: No such file or directory error reading information on service vsftpd: No such file or directory error reading information on service vsftpd: No such file or directory error reading information on service mysqld: No such file or directory error reading information on service mysqld: No such file or directory error reading information on service mysqld: No such file or directory Finishing system configuration, almost done... error reading information on service vsftpd: No such file or directory error reading information on service vsftpd: No such file or directory error reading information on service vsftpd: No such file or directory error reading information on service mysqld: No such file or directory error reading information on service mysqld: No such file or directory error reading information on service mysqld: No such file or directory error reading information on service xinetd: No such file or directory error reading information on service xinetd: No such file or directory error reading information on service xinetd: No such file or directory done.	Fix	Config
SIPX-331	Call Queue service labeled as "beta"	Remove the (beta) tag on Call queue in Admin Portal. Go to System -> Services -> Telephony Services "Call Queue (beta)" service should be renamed to "Call Queue".	Fix	Call Queue
SIPX-36	DND disable does not work	Fixed an issue that caused the Polycom disable DND feature to not work. There was a typo in the syntax that is used in the config file generated for each phone. The config file contains: feature.doNotDisturb.enable.enabled="0" But it needs to be feature.doNotDisturb.enable="0" This is controlled by the setting under Features > Do Not Disturb in the phone or group config.	Fix	Polycom
SIPX-371	Unite Web Redirect to insecure link	Fixed an issue where if an administrator went to https://servername/sipxconfig/ it redirected the browser to http://servername/sipxconfig/app where it should have redirected to https://servername/sipxconfig/app.	Fix	Config

SIPX-50	When saving a group onSave() is triggered twice	Fixed an issue that caused group settings saves to be done twice.
		
		The problem was that when calling from UI we did:
		
		Group group = getGroup();
		if (equalsIgnoreCase(User.GROUP_RESOURCE_ID, group.getResource())) {
		getCoreContext().saveGroup(group);
		} else if (equalsIgnoreCase(Phone.GROUP_RESOURCE_ID, group.getResource())) {
		getPhoneContext().saveGroup(group);
		} else {
		getSettingContext().saveGroup(group);
		}
		
		In CoreContextImpl we have:
		
		@Override
		public void saveGroup(Group group) {
		m_settingDao.saveGroup(group);
		}
		
		both methods were intercepted and onSave event was published.
			Fix	Config

SIPX-78	Polycom VVX 1500 Missing Desktop Integration	The Polycom VVX 1500 phone did not have the Desktop Integration settings available in the template. Added the settings.	Fix	Polycom

UC-2305	GUI crash in a regions partitioned setup (intermitent)	Fixed an issue with Regions such that when Cluster was in a partitioned state after making config changes that apply to all servers, GUI would crash because httpd service crashed. Setup : 8 servers in HA with 3 regions (2+1 default) Steps : 1.Default region is no longer available - partitioned 2.From the GUI make a config changes that applies to all servers Reported issue : After making config changes that apply to all servers, GUI will crash, httpd service will crash Workaround : service sipxconfig restart	Fix	Regions
UC-2407	Error message after changing User ID needs an update	Fixed an error message to indicate that Password needed to be changed and not PIN. Steps to reproduce: 1.Go to Users->Users->User 200 and changes its User ID to 250 2.Apply Reported issued : you will receive an error : "When changing user name, you must also change PIN" Expected : When changing user ID , you must also change Password	Fix	Config
UC-2763	Remove IM Chat Room section from Conference page	With UC-2671 Decouple Conferences and Chat Rooms - getting fixed, the IM Chat Room section is no longer needed under the Features->Conferencing->"Server"->Conferences->"Conference" These settings were removed.	Fix	Config
UC-3228	Linksys spa942 phones need template change	Fixed an issue where for mwi the voicemail box (mailbox id) should be set to the user extension and voicemail server should be blank. If many of these phones were configured the old setting caused a flood of subscribes to the server.	Fix	Config
UC-3596	Add nodes IP and FQDN in /etc/hosts	We received some tickets where although named is starting prior to mongod it fails to serve IP address of the other nodes resulting in following errors on mongod.log. We've added server node names and IP addresses to servers' /etc/hosts file to alleviate the problem.	Fix	Config
UC-3663	eMail notification checkbox has opposite effect	It was reported that when you did not check the box to notify the user of account creation that the user was notified. When you add a new user , the NOTIFIED (and not "notify" - there's some confusion) checkbox will be by default unchecked. That means the user will be informed by his user creation Otherwise if you check it , the system considers the user to have already been notified , so the email is not sent. We changed the label from "Notified" to "User was notified".	Fix	Config
UC-3714	Internal Error when Adding PSTN to Audiocodes Mediant 1000 Gateway	Fixed an issue when attempting to add a PSTN Line to an Audiocodes Mediant 1000 PRI gateway profile the error below is presented in sipxconfig.log. Caused by: org.springframework.orm.hibernate3.HibernateObjectRetrievalFailureException: No row with the given identifier exists: [org.sipfoundry.sipxconfig.gateway.FxoPort#-1]; nested exception is org.hibernate.ObjectNotFoundException: No row with the given identifier exists: [org.sipfoundry.sipxconfig.gateway.FxoPort#-1] at org.springframework.orm.hibernate3.SessionFactoryUtils.convertHibernateAccessException(SessionFactoryUtils.java:677) at org.springframework.orm.hibernate3.HibernateAccessor.convertHibernateAccessException(HibernateAccessor.java:412) at org.springframework.orm.hibernate3.HibernateTemplate.doExecute(HibernateTemplate.java:411) at org.springframework.orm.hibernate3.HibernateTemplate.executeWithNativeSession(HibernateTemplate.java:374) at org.springframework.orm.hibernate3.HibernateTemplate.load(HibernateTemplate.java:551) at org.springframework.orm.hibernate3.HibernateTemplate.load(HibernateTemplate.java:545) at org.sipfoundry.sipxconfig.gateway.GatewayContextImpl.getPort(GatewayContextImpl.java:76) at sun.reflect.NativeMethodAccessorImpl.invoke0(Native Method) at sun.reflect.NativeMethodAccessorImpl.invoke(NativeMethodAccessorImpl.java:57) at sun.reflect.DelegatingMethodAccessorImpl.invoke(DelegatingMethodAccessorImpl.java:43) at java.lang.reflect.Method.invoke(Method.java:606) at org.springframework.aop.support.AopUtils.invokeJoinpointUsingReflection(AopUtils.java:317) at org.springframework.aop.framework.ReflectiveMethodInvocation.invokeJoinpoint(ReflectiveMethodInvocation.java:190) at org.springframework.aop.framework.ReflectiveMethodInvocation.proceed(ReflectiveMethodInvocation.java:157) at org.springframework.orm.hibernate3.HibernateInterceptor.invoke(HibernateInterceptor.java:111) at org.springframework.aop.framework.ReflectiveMethodInvocation.proceed(ReflectiveMethodInvocation.java:179) at org.springframework.security.access.intercept.aopalliance.MethodSecurityInterceptor.invoke(MethodSecurityInterceptor.java:64) at org.springframework.aop.framework.ReflectiveMethodInvocation.proceed(ReflectiveMethodInvocation.java:179) at org.springframework.aop.framework.adapter.MethodBeforeAdviceInterceptor.invoke(MethodBeforeAdviceInterceptor.java:52) at org.springframework.aop.framework.ReflectiveMethodInvocation.proceed(ReflectiveMethodInvocation.java:179) at org.springframework.aop.framework.JdkDynamicAopProxy.invoke(JdkDynamicAopProxy.java:207) at com.sun.proxy.$Proxy32.getPort(Unknown Source) at org.sipfoundry.sipxconfig.site.gateway.port.PortSettings.pageBeginRender(PortSettings.java:65) at org.apache.tapestry.AbstractPage.firePageBeginRender(AbstractPage.java:409) at org.apache.tapestry.AbstractPage.renderPage(AbstractPage.java:244) at org.apache.tapestry.engine.RequestCycle.renderPage(RequestCycle.java:400) ... 69 more Caused by: org.hibernate.ObjectNotFoundException: No row with the given identifier exists: [org.sipfoundry.sipxconfig.gateway.FxoPort#-1] at org.hibernate.impl.SessionFactoryImpl$2.handleEntityNotFound(SessionFactoryImpl.java:435) at org.hibernate.event.def.DefaultLoadEventListener.load(DefaultLoadEventListener.java:233) at org.hibernate.event.def.DefaultLoadEventListener.proxyOrLoad(DefaultLoadEventListener.java:269) at org.hibernate.event.def.DefaultLoadEventListener.onLoad(DefaultLoadEventListener.java:152) at org.hibernate.impl.SessionImpl.fireLoad(SessionImpl.java:1090) at org.hibernate.impl.SessionImpl.load(SessionImpl.java:985) at org.hibernate.impl.SessionImpl.load(SessionImpl.java:978) at org.springframework.orm.hibernate3.HibernateTemplate$3.doInHibernate(HibernateTemplate.java:558) at org.springframework.orm.hibernate3.HibernateTemplate.doExecute(HibernateTemplate.java:406) ... 92 more	Fix	Audiocodes
UC-3750	GRUU problem in 15.06	Fixed an issue with GRUU caused by an earlier commit (introduced in 15.06) that caused GRUU to break when used with wss transport. Customer was using GRUU address to target specific registration instances. Customer used to receive the contact's GRUU address as: ~~gr~BtXVZtl2D-I@server.fqdndomain.com;gr and now doesn't receive it on registration.	Fix	SIP
UC-3784	CDR Display name cut off on call queue call	Fixed an issue where certain characters (the '/' character in this case) in caller-id were causing issues with CDR. Worked around the issue by replacing '/' with '-' in callerid name.	Fix	Config
UC-442	Conference Control Labels incorrect	Fixed an issue on the conference config page where the DTMF digits for various conference operations, the "Volume" and "Microphone Gain" options are defined as they were labeled backwards. For example, a conference participant presses "6" to increase his volume into conference or "3" to increase the conference's audio to him. But the labels read the opposite. The Freeswitch config the GUI generates the following: <control action="vol talk up" digits="3"/> <control action="vol talk zero" digits="2"/> <control action="vol talk dn" digits="1"/> <control action="vol listen up" digits="6"/> <control action="vol listen zero" digits="5"/> <control action="vol listen dn" digits="4"/> The labels here seem to correspond to the GUI labels, "vol talk" for Mic Gain and "vol listen" for Volume. That makes sense. The actual behavior, however, is opposite.	Fix	Conference

UC-473	Change LDAP + PIN to LDAP + Password	Fixed an issue with error on login where the error message indicated PIN, it should reference Password.	Fix	Config
UC-492	Phone Book not Paginating Correctly	Fixed a problem with the 'Next' and 'Previous' page buttons not working (remain greyed out) regardless of the count of users within the phone book as best as possible with the control application that was used for this feature. If we limited pagination to 12 the search feature would only search those 12 for a user. We decided to set the page size to 500 to keep searching working. Next and Prev will work if you have over 500 users in phonebook.	Fix	Config
UW-313	Notifications no longer working	Fixed an issue that caused the red dot representing new IMs to no longer be displayed on the browser tab. Sound notifications also don't work. This is the Console error: TypeError: Notify.isSupported is not a function at Notify.myNotification.needsPermission (fc9f128370b1ea9770ccfadb.scripts.js:1) at fc9f128370b1ea9770ccfadb.scripts.js:1 at handleMessage (fc9f128370b1ea9770ccfadb.scripts.js:1) at Object.h.Handler.run (4e42ce69bdd61567355bfd36.modules.js:3) at 4e42ce69bdd61567355bfd36.modules.js:3 at Object.h.forEachChild (4e42ce69bdd61567355bfd36.modules.js:3) at Object.h.Connection._dataRecv (4e42ce69bdd61567355bfd36.modules.js:3) at Object.a.Bosh._onRequestStateChange (4e42ce69bdd61567355bfd36.modules.js:4)	Fix	UniteWeb
SIPX-332	Polycom VVX 5.4.0 only supporting 34 total lines	It was reported that Polycom firmware 5.4 supports 34 total lines+BLFs compared to 48 on firmware 4.1.7 e.g. On VVX 600 running fw 5.4.0 if you deduct the total number of line keys supported, which is 16 , you will only have 18 slots available for BLFs Compared to 4.1.7 - 32. Also valid for firmwares 5.3,5.2,5.1 From: http://support.polycom.com/global/documents/support/setup_maintenance/products/voice/EM_AG_Addendum.pdf Polycom phones can have multiple registrations; each registration requires an address or phone number. All VVX phones support up to 48 registration line keys and up to 34 line registrations when connected to three expansion modules.	Known Issue	Polycom
SIPX-377	Document Known Issue for SIPX-376	If a server in a partitioned region is rebooted before WAN comes up the Registrar service will fail to start. How to reproduce: Setup 2 regions, one region has 4 servers, and the other has one. Have phones registered in both. Partition off the regions from each other (as in a WAN failure). Reboot the server at the 1-server region. After the server reboots, the registrar service will fail to start. Registrar will start working properly once the WAN comes up.	Known Issue	SIP, Regions
SIPX-378	Document Known Issue for SIPX-335	If an admin changes a user's location the changes are not reflected in Mongo. Workaround: send server profiles.	Known Issue	Config
UC-3793	Issues with IC and Chrome in adding secondary database	Customer installed a 3 node cluster from 15.08 ISO. Using IE or Chrome, he was unable to add secondary databases in system -> database due to browser/certificate issues. This is not an issue if you've got FireFox (currently) so I haven't labeled it as a blocker, however I think it should be noted in the 15.08 / 15.10 release notes. Fix: Install a valid certificate on system or use FireFox.	Known Issue	Config