sipXecs 4.4

Improvement

- XX-5004 - support distinctive rings via Alert-Info
- XX-5367 - Party invited to a conference by owner must still enter PIN
- XX-6048 - Provide alarm for expiring certificate
- XX-6120 - "Per-user timer for call forward" should get its default value from "Default serial fork expiration"
- XX-6294 - Enable building sipXecs components in a chroot environment (such as mock or the SUSE build service)
- XX-7419 - LDAP import: Add SSL/TLS support
- XX-7813 - Let all external interactions (remote worker and ITSP) go through a single external port to 5060.
- XX-8089 - Add profile and auto-provisioning for Polycom IP 5000 model
- XX-8173 - Support CentOS 5.4
- XX-8209 - Dialplans are unnecessarily replicated twice in certain senarios
- XX-8218 - User Portal: Change "Conference" to "Conference Recordings"
- XX-8332 - Add support for HTTP PUT to supervisor mangement APIs
- XX-8334 - Remove the Devices -> SBC Routes configuration entity
- XX-8398 - Upgrade stunnel to 4.28 for CentOS 5.4 support
- XX-8474 - Improve RLS scaling capabilities for large installations
- XX-8507 - Allow for Language selection in the Startup Wizard
- XX-8632 - Bria 3.1: "See this person's Jabber online presence" should be enabled.
- XX-8645 - t.38 Fax support in sipxecs
- XX-8651 - track rpms built from information from git repo, not svn
- XX-8757 - RLS server to improve scalability
- XX-8884 - Configure media server to utilize mod_spandsp from freeswitch (in FS 1.07)
- XX-8932 - Fully support all traffic on port 5060 to be diverted to sipXbridge
- XX-9027 - Secure web service APIs from sipXivr
- XX-9028 - Additions to voicemail webservice APIs
- XX-9029 - Way to identify voicemail email messages uniquely from other emails
- XX-9071 - Need to generate .keystore in sipXivr
- XX-9209 - Enable TLS for polycom
- XX-9311 - Allow sipxconfig plugins to contribute own Tapestry pages
- XX-9316 - Default configuration information for Nortel 11xx phones
- XX-9319 - openACD agent security to be managed from sipxconfig
- XX-9320 - Pull in current openacd release management changes
- XX-9351 - Should not display a message "XXX is invited to conference" from Admin protal participant screen when owner is unassigned
- XX-9352 - sipXivr: MailboxServlet: xml response should follow XML specifiation
- XX-9362 - Integrate queues and clients in Call Center Line
- XX-9368 - Change default LDAP / AD settings to better suit AD by default
- XX-9371 - Create URL to expose the root sipx certificate for download
- XX-9409 - Extend sipXconfig license pluggable mechanism to support multiple license files
- XX-9463 - Call transfer: getting ongoing calls for current user
- XX-9530 - IVR: don't play thank you goodbye on AA transfer on failure

New Feature

- XX-4808 - Support SIP over TLS connections in sipXtackLib
- XX-4989 - Fax service
- XX-5357 - Allow remote database access to specified address for Billing Applications
- XX-6279 - import contact information from LDAP
- XX-7650 - Support for Unidata Phones
- XX-8501 - Authorization Codes
- XX-8529 - Add a per-User "Fax Extension" setting, replicate to sipXproxy
- XX-8530 - sipXproxy support for fax to email
- XX-8533 - Add Authorization Code configuration
- XX-8614 - Add NAT Traversal support for TLS port in sipXconfig
- XX-8666 - Update default ldap port for TLS should not have to be entered
- XX-8667 - Test / Update Counterpath Bria 3.1 softphone
- XX-8682 - Enable G.729 codec for media services
- XX-8685 - Allow Google Chrome to be a supported browser
- XX-8704 - Publish language packs
- XX-8711 - Update link for the Help function in admin and user UI
- XX-8725 - Polycom plugin patch: administrator usable custom features hook
- XX-8726 - Aastra phones - disable GRUU
- XX-8739 - French-Canadian localization by Abitibi-Temiscamingue
- XX-8755 - getting STUN alarms even though the alarms are disabled
- XX-8770 - Russian localization package
- XX-8772 - Called conference invitee needs to authenticate against system
- XX-8776 - unable to add users to hunt group
- XX-8778 - Plugin Support for Karel IP11X Phones (Yealink IP Phones)
- XX-8780 - Move over stories from backlog.ezuce.com to track.sipfoundry.org
- XX-8781 - Use LDAP to authenticate in OpenFire
- XX-8782 - Add an 'About' menu to the diagnostics tab
- XX-8783 - phase 1 - Manual config of call flows to use sipXecs as the proxy with FreeSWITCH as the media server
- XX-8784 - Improve syncrhonization with AD/LDAP: Simplify and add importing contact info

- XX-8786 - 64bit CD ISO
- XX-8787 - Allow user portal authentication using LDAP bind
- XX-8788 - Authorization codes to place calls
- XX-8789 - FINISH: Use LDAP to authenticate in OpenFire
- XX-8790 - LDAP preview for new fields
- XX-8791 - Make LDAP server config more prominent
- XX-8792 - Option to enforce LDAP only authentication in user portal
- XX-8794 - Upgrade RPM and CD ISO builds to CentOS 5.5 / RHEL 5.5
- XX-8795 - Allow queriing user's presence based on email address
- XX-8796 - Enable REST service for Presence
- XX-8797 - Plugin support for Karel phone
- XX-8798 - Cisco Unity call diversion support
- XX-8799 - Improve sipXbridge to accommodate To field routing and P-Preferred identity for CallerID
- XX-8800 - Allow for shared mailboxes
- XX-8801 - Add 'stunnel' to list of packages built for CentOS 5. (XX-6619)
- XX-8804 - Use LDAP to authenticate in OpenFire
- XX-8806 - phase 1 - Manual config of call flows to use sipXecs as the proxy with FreeSWITCH as the media server
- XX-8813 - LDAP preview for new fields
- XX-8814 - Make LDAP server config more prominent
- XX-8819 - Enable REST service for Presence
- XX-8820 - Plugin support for Karel phone
- XX-9069 - Managed Phone support for isphone softphone
- XX-9072 - REST API Call to get user's conference number if configuured
- XX-9110 - Snom m3 plug-in
- XX-9203 - Manage OpenACD queue and queue groups from admin interface
- XX-9205 - Manage OpenACD skills from admin interface
- XX-9206 - Associate OpenACD customer identity to calls from admin interface
- XX-9364 - Add a Start and Stop Button In Config UI for server in Services Page

Task

- XX-6671 - upgrade lucene 2.0 => 3.0.2
- XX-6893 - ITSP needs us to provide CLID in P-Preferred-Identity header not in From header and put username in From header
- XX-7889 - Fix failing ManageVoicemailTestUi testMove() and testDeleteFriendlyUrl() UI unit tests
- XX-8250 - Remove branch column from SBC database table and provide upgrade capability
- XX-8339 - Remove un-used Discover Devices code from sipXconfig
- XX-8885 - Improve development environment
- XX-8930 - Retest Bria XMPP issues
- XX-9067 - Postgres 8.1 is current DB for config storage, but 8.1 is now EOL November 2010
- XX-9070 - Spam in the comment section of a wiki page
- XX-9289 - Support Fedora 14
- XX-9323 - Freeze sipXconfig DB schema
- XX-9331 - Remove DimDim support from the user portal

Bug

- XX-5120 - voicemail caller identification uses "Outside caller" when callers user address has a leading '+'
- XX-5730 - upper case in profile file name for spa942/spa962
- XX-7201 - Running the command "sipx-trace -a -o <filename> <call-id> " displays error messages
- XX-7244 - Loop is configured when a conference name is the same as its extension
- XX-7610 - Device files are not included in a backup
- XX-7625 - Nortel phones: Caller gets disconnected upon consultative transfer to voicemail.
- XX-7765 - XMPP SRV records are not generated in zone files when DNS is enabled Onboard
- XX-7891 - Same user entries may show up more than once in webdav.user.password
- XX-7986 - sipxconfig-ftp – scriptlet failed with exit status 1 on any update
- XX-8291 - localization package files are not installed due to permission problem
- XX-8402 - sipXivr Menu and PromptList fixes
- XX-8419 - Call dropped from phone containing shared user
- XX-8528 - sipxconfig not generating useable firewall settings for Bria Pro 3.0
- XX-8531 - psql scripts should use CREATE TEMP TABLE AS instead of SELECT INTO
- XX-8570 - non-us locale in VM message digest file format causing exception in user portal
- XX-8609 - Localization: MyBuddy needs to know the sipXecs locale string
- XX-8629 - Localization Package Fails to install - "sipxlocalization: '/usr/share/java/sipXecs' is not a writable directory."
- XX-8630 - sipXbridge: No audio upon answering a call which is blind transferred after a directed call pickup (if Skype is the ITSP)
- XX-8633 - Bria 3.1: "See this person's Jabber online presence" parameter gets disabled on every login.
- XX-8634 - Bria 3.1 : Call to User added under Instant Messaging enabled user Group fails
- XX-8640 - Attended transfers to FreeSWITCH services fail
- XX-8646 - PIN from TUI cannot be changed
- XX-8650 - Internal error adding device files for polycom
- XX-8653 - Device files are not correctly created for Nortel 1120e /1140e phones
- XX-8655 - Order of gateways in dialplan not being followed correctly
- XX-8657 - ldap authentication authenticates EVERY user both as the BIND USER, and using the credentials being given it
- XX-8658 - English prompts copied in the localized directory
- XX-8659 - sipxsupervisor doesn't stop cleanly
- XX-8672 - sipXconfig BuildRequires: sipx-jasperreports-deps shall be removed
- XX-8673 - Setting the global logging level to DEBUG causes an exception
- XX-8785 - stunnel doesn't update cleanly

- XX-8802 - Configure FTP port range so that phone profiles can be retrieved through a firewall
- XX-8803 - update java 1.6.0 to update 16, not 11
- XX-8825 - Configure FTP port range so that phone profiles can be retrieved through a firewall
- XX-8826 - update java 1.6.0 to update 16, not 11
- XX-8827 - Voicelmail NullPointerException error
- XX-8828 - sipx-freeswitch rpm revision number needs increament so that systems know to update
- XX-8829 - Internal error after importing users/phones
- XX-8832 - When phone profiles are generated via SOAP request affected phones are always restarted
- XX-8835 - sipxbridge From header is ITSP account info instead of originating caller
- XX-8842 - Install the Verisign certificate by default used for Google contact sync
- XX-8843 - @call does not work
- XX-8848 - VM storage directory of the user is not deleted after the user is deleted.
- XX-8878 - Polycom 550/560 phone template does not set DST flag to 1 in SNMP settings
- XX-8883 - Bria 3.1 Professional Pulls VCard information from OpenFire but is pulling user extensions and adding "(Work)" on the end
- XX-8888 - Stack bug with content type and quotes
- XX-8910 - sipXconfig creates empty <MAC_ADDDRESS_OF_THE_PHONE-sipx-phone.cfg when external line is addded to Polycom Phones
- XX-8912 - SAA service is mistakenly setup on the HA slave server
- XX-8917 - In Auto-attendant, Default values of Inter-DTMF Timeout and Overall DTMF-timeout are set to wrong values.
- XX-8924 - sipXproxy cores and restarts
- XX-8926 - Conference 'auto-record' does not work due to absence of /tmp/freeswitch/recordings directory.
- XX-8929 - Fully support transfer thru FreeSWITCH
- XX-8936 - sip password don't generated during LDAP import
- XX-9057 - Backup does not work on ISO and rpm installs because /var/sipxdata/backup data does not get created
- XX-9064 - 4.2.1 - "Delete message" link doesn't work
- XX-9085 - When we send profiles to the phones, the phones are not restarting
- XX-9086 - sipXecs path in sipxecs.repo set to incorrect value
- XX-9087 - In User-> MOH->Click Apply and Ok, Phone is getting Rebooted
- XX-9093 - sipxbridge.xml file is not generated when a SIP Trunk is created
- XX-9095 - Internal Exception occurs when the user stays on a page and restarts the server
- XX-9109 - Snom 3x0 plug-in and firmware 8.x - Corrupted address book
- XX-9111 - Contact list doesn't appear under Contact on Bria phone
- XX-9114 - LDAP page not showing error message
- XX-9127 - MyBuddy: Pickup command fails
- XX-9191 - User presence sync issue when using XML RPC api provided by sipXopenfire presence plugin
- XX-9193 - After yum update, the updated version is not getting reflected in the config ui
- XX-9199 - Failed unit tests in sipXproxy
- XX-9213 - One of the services(ConfigAgent) is not running after we installed the new build sipXconfig (0.0.4.3.2-ab500552010-10-29T20:13:27 build20)
- XX-9228 - Not able to upload the localization package from the config UI
- XX-9229 - Caller Id set on user account not sent out to gateway / ITSP (Aastra)
- XX-9235 - Figure out why successful faxes are not sent via mail
- XX-9236 - Memory not released after backup
- XX-9237 - Personal AA call is not transferred
- XX-9238 - IVR call is not getting disconnect after Pressing Logoff option
- XX-9239 - After every login of bria,the workgroup address disappears hence workgroup is not working
- XX-9241 - When I try to yum update the 0.4.4 ISO build, it comes up with error message
- XX-9242 - Voicemail login through AA cannot hear the voicemail Prompt
- XX-9243 - One of the services(SIPXProxy) is not running after we installed the new build sipXconfig (0.4.4-5b9dc0c 2010-11-23T05:33:41 build16)
- XX-9248 - SipXbridge Fails to start a session timer for reinvites after getting a 407 response
- XX-9250 - Blind Transfer to Voicemail box fails
- XX-9253 - MyBuddy / Listen & Pickup / INVITE / From display/username
- XX-9293 - core dumps is generated from sipxconfig-agent
- XX-9294 - Not able to log into the config UI when ldap is running
- XX-9295 - Google import fails with NPE - null rel for email
- XX-9299 - httpd: Syntax error on line 1087 of /etc/sipxpbx/httpd.conf: Could not open configuration file /etc/sipxpbx/httpd-sipxchange-common.conf: No such file or directory
- XX-9321 - Bad error message while trying to edit agent user and mongodb is not running
- XX-9322 - FS sessions not released due to linger command
- XX-9324 - error running sipx-snapshot: usr/bin/sipxecs-config: No such file or directory
- XX-9325 - Calls dropped after transfer
- XX-9326 - Error updating sipxpbx : /usr/bin/sipx-sendmail-configure: line 65: syntax error
- XX-9328 - LDAP import: handle byte arrays properly
- XX-9329 - ACD Historical Reports not working after moving role
- XX-9333 - Gateways order sequence is changing after press ok button on Gateway details page
- XX-9335 - core dumps is generated from sipxrls
- XX-9336 - memory leak in registrar in HA env
- XX-9338 - VM and AA DID is not working
- XX-9339 - Call is not Transferred through Dial by Name
- XX-9342 - conference Extension Call getting disconnect After entering pin more than 10 values...
- XX-9344 - Cannot Access superadmin after yum update shows HTTP ERROR 500
- XX-9345 - When Automatically created conference,the Participants pin is empty in Conference page in Config UI.But when I Call to this conference Extension its asking Pin
- XX-9347 - In Authorization Code Help text for code is mentioned wrong value
- XX-9354 - BasicXmlRpcAuthenticationHandler: fails with Null pointer exception when no user/password is provided
- XX-9356 - OpenACD startup error
- XX-9359 - If the "User Group attribute" (ou) has more than one value, multiple groups are not created.
- XX-9363 - Cannot Park the Call Again After the,Parked call is disconnected or retrieved
- XX-9367 - Internal Exception Seen when click on Report list down Option in Calls(CDR) Page
- XX-9384 - extra error messages in sip servers

- XX-9385 - sipXcdrLog: records wih caller/callee 1201 are returned when 201 is requested
- XX-9386 - LDAP import: adding to groups, not updating
- XX-9399 - OpenACD: 'Agent Groups' page: 'An internal error has occurred.'
- XX-9411 - After ISO Installation,setting the logging level to DEBUG the ACD server fails
- XX-9428 - Conference call ended with beep sound,when user Resume the call while putting it on Hold for second time by the same user
- XX-9430 - Error Message not displayed in Presence server
- XX-9431 - Not displaying Error Message "Conflict Code" for Authorizaton Code ,Paging and SipRegistrar
- XX-9432 - Openfire Cannot Authneticate against Active Directory
- XX-9437 - "which" package missing as dependency
- XX-9438 - Conference: 2-3 seconds delay in audio
- XX-9440 - Internal Error, duplicate personal attendants in postgres for random users
- XX-9444 - Not displaying a Error messge when Feature Code is set as Alias or DID
- XX-9448 - Cannot localize multiple Tapestry components - due to missing jwc
- XX-9453 - sipXecs setup scripts default text cannot be overwritten
- XX-9454 - GWT Components are not localized according to browser language
- XX-9455 - PluginSpecificationResolver should handle only pages within org.sipfoundry.sipxconfig.site.plugin package
- XX-9457 - Config plugin pages cannot be localized
- XX-9460 - Park server don't clear transferred or closed call
- XX-9468 - Consultative transfer gets disconnected while trying through auto attendant
- XX-9470 - Config UI allows to create UserID with special Character while import a CSV file from Import/Export option
- XX-9471 - Nortel 11xx malfunction during conference calling
- XX-9474 - Welcome audio prompt is not playing while calling to the ACD Extension
- XX-9475 - XML-RPC service failure when adding more than 61 users
- XX-9476 - No VM audio flow after yum update from 4.2.1 to 4.4.0
- XX-9477 - The call is not getting transfered back to the extension which parked the call after the time specified in Park timeout is elapsed
- XX-9478 - The Established call on ACD Extension is not getting disconnected
- XX-9481 - Extension and DID should not be present in User Group > Unified Messaging
- XX-9485 - For a higher search base in LDAP, the LDAP-Only authentication does not work
- XX-9486 - BasicXmlRpcAuthenticationHandler:"Could not authenticate since no username or password was provided seen in sipxopenfire.log
- XX-9490 - LDAP-openfire authentication is not supported for a LDAP anonymous configuration
- XX-9493 - The Name(server name) is not displayed correctly under conference , when we invite an external user
- XX-9495 - Conferences: consultative transfer doesn't work
- XX-9497 - Cannot fix the PPI on SIP Trunk to set value
- XX-9498 - Party invited to a conference by owner must still enter PIN in case of using sofphones (e.g. Bria, X-Lite)
- XX-9514 - sipXrest Basic/Digest authenticators use SIP_DOMAIN_NAME as authentication realm instead of using SIP_REALM
- XX-9521 - Call forward does not work with 4.4 and Verizon (Broadsoft)
- XX-9522 - Internal Exception seen when try to move agent from one group to another group while mongoDb is not running
- XX-9525 - OpenACD: Internal error when trying to delete an user assigned as agent
- XX-9526 - Displays the message "&MSG.CANNOT.CONNECT" instead of "Cannot connect to Call Center" in Agent groups Page
- XX-9528 - "Intermittent" Agent in OpenAcd should not get deleted when mongod services is not running