

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 scenarios
- [XX-8218](#) - User Portal: Change "Conference" to "Conference Recordings"
- [XX-8332](#) - Add support for HTTP PUT to supervisor management 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 portal participant screen when owner is unassigned
- [XX-9352](#) - sipXivr: MailboxServlet: xml response should follow XML specification
- [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.ezuze.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 synchronization 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 querying 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 configured
- [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](#) - Voicemail NullPointerException error
- [XX-8828](#) - sipx-freeswitch rpm revision number needs increment 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_ADDADDRESS_OF_THE_PHONE-sipx-phone.cfg when external line is added 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 with 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 Authorization Code, Paging and SipRegistrar
- XX-9432 - Openfire Cannot Authenticate 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 an Error message 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 transferred 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 softphones (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