sipXecs 3.10

New (nicer) skin

We added a new default skin for sipXconfig and moved away from the traditional yellow background. All of sipXconfig is now easily skinnable including the creation of a custom login page.

Even easier installation / device discovery

In addition to plug & play management of phones and gateways, this release adds an auto-discovery function for devices. Phones and gateways are found automatically and presented in a table from where they can be added to the database in one click only. Also in this release a new network services test capability has been added. When sipXconfig starts all the necessary network services, such as DHCP, DNS, NTP, TFTP, FTP, HTTP, are tested for correct configuration and operation. Detailed error messages are printed with troubleshooting information. The test suite can also be downloaded to a laptop and run under Windows. That way the tests can be run on the same subnet the phones are connected to,

Extended User Portal / time based find-me / follow-me

The sipXecs user portal is available to every user of the system and allows individualized management of key user features. Tn addition to the management of unified communications and voicemail, the user portal now also supports time-based find-me / follow-me, personal call history, personal phone management, and personal management of phone book, speed dial, and presence subscriptions.

Personal auto-attendant / Individual zero-out capability

Every user gets a personal auto-attendant that can be configured on the user portal or by the admin. When a caller is redirected to the user's voicemail, the caller will hear an individually recorded greeting that provides instructions on how to reach the user or to leave a voicemail. The user can define individual keys, such as press 1 to get forwarded to my cell phone, press 2 to get transferred to my assistant, press 3 to reach my girl friend and press 4 to leave a voicemail. Also, it is possible to define an individual transfer extension for the 0 key, which is usually the operator or a personal assistant.

Import from and export to Excel

During the planning phase before an installation, many users create cut-sheets that identify users, extensions, phone models, passwords, and other necessary parameters. Once this information is captured in Excel it can be uploaded into sipXconfig, greatly simplifying the installation process. At the same time this information can now be exported to Excel as well.

Localization of the Media Server

The last release brought about localization of the Config Server as well as the voicemail user portal. In this release we are adding localization of voice prompts for the auto-attendant and voicemail systems for a first set of languages. German, Italian and Polish are currently in process with others to follow. We will define a simple format for language packs, so that localization can be easily done in the community.

Busy Lamp Field (BLF) and Presence

In release 3.8 we got BLF almost right and we added a new SIMPLE based presence server. However, because of a bug in the Polycom 2.x firmware, BLF still does not work reliably under all use cases. Release 3.10 will see improvements in the BLF implementation that will make the feature less dependent on phones and extend the capability to phones that comply with the SIP stndard (e.g. LG-Nortel phones).

Integration with Microsoft

Release 3.10 provides a unified communications solution integrating with Microsoft Exchange 2007 as well as Active Directory. Microsoft Exchange 2007 can be selected as an alternative voicemail system directly in the dialplan. This provides a speech enabled voicemail system integrated with the Exchange email and calendar system. Synchronization of users and their credentials can be done automatically using the integration with Active Directory.

Time-Based Routing

We are introducing a time-based routing capability into the dial plan. This is based on a new redirector plugin and allows all kinds of time dependent features and feature interactions.Every dialing rule has now an optional schedule attached.

Paging Server

Based on the specification we published some time ago we added a group paging server to the sipXecs system. The paging server is added as a distributed component where several paging servers can be added to the system, either on the same host as the rest of the sipXecs system or on separate HW. The paging server allows group paging of SIP phones. Different announcement audio can be selected to announce a page. Regular SIP phones that provide auto-answer capability can be used or dedicated SIP-based speakers (e.g. in-ceiling)

Overhaul of the ACD server

The ACD server has been overhauled and made a lot more stable. Additional features include agent wrap-up time as well as an agent auto-sign-out capability in case the agent does not answer a call. Also, the overflow mechanism has been enhanced with a better algorithm and more destinations. E.g. it is now possible to use a queue, a hunt group or an individual extension as an overflow destination. If no agent is signed in the call can overflow to voicemail.

Improvements to the Auto-Attendant

Several important improvement to the auto-attendant subsystem have been queued up for quite some time. In particular we added transfer rules and targets to handle invalid response. Also, the auto-attendant can now transfer to external numbers with proper permissions.

Improvements to Hunt Groups

More flexibility is added to the management of hunt groups so that it is possible to specify destinations for no answer. Such destinations can include voicemail, auto-attendant, an extension or SIP URI, or another hunt group. See XCF-831. In addition, the difference in behavior between transferring consultative or blind to a hunt group will be eliminated. On a per hunt group basis the admin can now configure whether user call forwarding rules shall be followed or not. This allows disallowing forwarding of calls to e.g. user's cell phones as part of a hunt group.

Overhaul of the security and authentication system

The security system of sipXecs for call authorization has been overhauled. This should eliminate previous restrictions on call tromboning or other external forwarding (blind or consultative transfer of an external call to an external number) while strengthening the security of the system. Gateway templates now automatically configure Access Control Lists (ACL) to prevent unauthorized LD calling.

Improved E911 call routing

Resiliency of emergency call routing has been improved. Phones able to directly route emergency calls to a gateway without requiring the sipXecs server to be operational are now automatically configured to use this feature. Emergency calls, therefore, will now succeed even if the sipXecs server is not available as long as the phone can talk to the gateway.

Simplified dial plan configuration

Gateways can now be added to dialing rules directly from where gateways are managed. A single click adds a newly created gateway to a dialing rule. Removing a gateway automatically deletes all its references in the dialing rules. Gateways continue to offer trunk redundancy and automatic failover in case of busy or unavailable. sipXecs therefore supports more than one gateway per dialing rule.

Registered phones displayed per user

Managing a large number of users, several hundred to several thousand, can be a difficult task. sipXconfig already offers elaborate search capabilities to filter reports. In this release there is now a very simple way to just display phones registered for a specific user. This is possible both by the admin in the admin portal or the user using the user portal.

New device category: SBC

In addition to phones and gateways, sipXconfig can now also manage Session Border Controllers (SBC). A new category of a managed device has been introduced. SBCs are used for Internet call routing rules, remote workers, as well as SIP trunks.

Automated restore from backup

The current restore from backup functionality will be integrated into Config Server.

Server and application statistics, reports, and alarms

We are implementing SNMP / MRTG based statistics into Config Server that allows improved monitoring, alarming and reporting of performance and problems. In addition, the sytem will allow integration into data center management applications.

Support for new Polycom 320 / 330 phones / Polycom 2.2.2 firmware

We are adding support for plug & play management of new Polycom phones. In addition, the plug & play management system has been updated to support firmware 2.2.2. Older phones IP300 and IP500 can no longer accommodate 2.2.2 firmware because of memory constraints acording to Polycom.

Plug & Play Management Support for Linksys Phones

We are adding support for Linksys SPA941 and SPA942 phones fully integrated into the sipXconfig management system thanks to a community contribution.

Plug & Play Management Support for IpDialog SipTone V Phone

We are adding support for the IpDialog SipTone V phone fully integrated into the sipXconfig management system thanks to a community contribution.

Plug & Play Management Support for LG-Nortel 1535 Video Phone

We are adding support for the LG-Nortel 1535 Video phone fully integrated into the sipXconfig management system thanks to a community contribution. This is a new and very attractive desk video phone.

New Report: Login history

sipXconfig now provides a report on the login history. This includes successful and unsuccessful logins from all users (superadmin as well as logins of ordinary users into the user portal).

Symmetric signaling / merged proxy

We introduced symmetric signaling, which is a first step towards supporting NAT traversal natively in sipXecs. This was achieved by merging the two proxies (forking proxy and authentication proxy) into one combined proxy server that communicates on default port 5060.

SIP loop detection

sipXecs proxy server is now able to detect loops and will abort them. We implemented a new IETF draft RFC for this important feature. Previously a call, under certain conditions, could loop indefinitely in the system.

Port to PowerPC (PPC)

sipXecs was ported to the PowerPC (PPC) platform with all the big endian handling for audio processing and other issues.

Port to FreeBSD

A new port was done to FreeBSD. We are still looking for a new maintainer who would be able to maintain this port in the FreeBSD ports library. Refer to XECS-108 for the port files and FreeBSD port for sipXecs 3.10 for documentation..

New XML RPC process management API

sipXconfig now uses a new XML RPC based API to manage processes on the master and slave hosts. Additional security and efficiency is provided over the old CGI based solution. This is a pre-req for the cluster management coming in the next release.

Detailed List

New Feature

- XCF-831 - flexibility in specifying "no answer" huntgroup destination
- XCF-1605 - Provide personal auto-attendant (per user) that offers options for forwarding targets
- XCF-1668 - Process Management Enhancement
- XCF-1787 - define forwarding schedules on a per group bases
- XCF-1788 - dial plan rules for configuring Exchange as Voice Mail server
- XCF-1792 - support both sipX and Exchange voicemail servers on a single system
- XCF-1748 - plugin for Linksys SPA941/942 phone
- XCF-1790 - add generic configuration test page to diagnostics menu
- XCF-1806 - Extend Scheduling to the System Dialing Plan
- XCF-1835 - generate and replicate domain-config file
- XCF-1840 - end user phonebook
- XCF-1387 - configuration for ENUM plugin
- XCF-1590 - Serviceability: Support server and application statistics inside sipXconfig, allow IT integration with common reporting and alarming tools
- XCF-1597 - Add plug & play support for new Polycom 320 / 330 phones
- XCF-1676 - superadmin logins should be logged with timestamp (successful and unsuccessful)
- XCF-1809 - Display user specific CDR records on the user portal
- XCF-1871 - Extend Scheduling to the Oher Dialing Plan Rules
- XCF-1810 - Add UI for paging server
- XCF-1920 - installable localization package
- XCF-1956 - sipXconfig UI for preflight

- XCF-1957 - configure internal media server language on a per user basis
- XCF-1984 - Add SIP_REALM to domain-config
- XCF-1987 - SBC plugins
- XCF-1833 - Remove hard coded limit of 500 users from SCS – Licensing mechanism will deal with the user limit
- XCF-1892 - Emergency Dialing needs to be added for LG-LIP Sets
- XCF-1917 - Configure identities for the park and media servers
- XCF-1926 - Provide link to download InGate configuration tool
- XCF-1958 - 'first-run' mode: generate initial configuration when sipXecs is stared for the first time
- XCF-1993 - Need to add link which allows download of Windows installer for Preflight
- XCF-2092 - Add default permissions for the media server identity
- XCF-2093 - Configure identity and permissions for the ACD
- XCF-2100 - Add support for Nortel 1120/40 sets
- XCF-1733 - export of current user list
- XCF-1893 - Audiocodes Parameters for Globalization files needs to be exposed
- XCF-1626 - Ability to display / print a phone book from the user portal
- XCF-1731 - FreeBSD port changes for sipxconfig
- XCF-1745 - Improve flexibility of routing calls to the operator

Improvement

- XCF-1344 - allow save of phone configs, without restarting phones
- XCF-651 - It should be possible to restore from backup using the UI
- XCF-1687 - Autoattendant configuration should be more versatile and allow all the same options that can be currently be set in vxml files.
- XCF-1773 - Linksys 942 line settings patch
- XCF-1785 - Phone and gateway configuration files should not be browsable
- XCF-1786 - day of time routing schedules: "weekend", "week day" and "every day" as entries in day list
- XCF-1799 - suppress generating <base/> tag to allow accessing UI through HTTP 'reverse' proxy
- XCF-1802 - Backup & Restore: It should be possible to upload backup files using the GUI
- XCF-1364 - LDAP not compatible with Win 2003 Active Directory
- XCF-1804 - Backup & Restore: Warn if admin attempts to restore backup from older version
- XCF-1834 - configure voicemail operator extension on a per user group basis
- XCF-1853 - Some pages should add a Quick Help box
- XCF-1860 - Allow specification of custom port and/or transport when creating/modifying gateways
- XCF-650 - Allow backup files to be sent to an email address
- XCF-1593 - check validity of imported wav files
- XCF-1825 - improvements to custom rule screen
- XCF-1854 - Help text for Hunt Group / Fallback destination
- XCF-1868 - Internal Dial Rule should be renamed Voicemail Dial Rule
- XCF-1869 - sipX Voicemail Server should be renamed Internal Voicemail Server
- XCF-1882 - Extend sipXconfig SOAP interface
- XCF-1910 - German dial plan template uses a wrong parent for long distance beans
- XCF-1919 - Services page to allow control over new services dynamically at runtime
- XCF-1947 - load localized property bundles from independently installed jar files
- XCF-1965 - Intercom feature should be extended to support other phone vendors
- XCF-1988 - plugin UI: add ability to display links to file resources on device configuration screen
- XCF-1768 - GUI should display user aliases in a consistent order
- XCF-1780 - check if phone or user have active registration
- XCF-1844 - Intercom Feature screen help text state only Polycom is supported. Need to change for LG
- XCF-1879 - user portal navigation is too wide
- XCF-1903 - Improvement of the SIP Trunk GW configuration
- XCF-1951 - changing region on localization resets dial plans without warning
- XCF-1952 - hide SNMP community on MRTG configuration page
- XCF-1953 - mrtg summary graphs should be clickable
- XCF-1969 - sipx server's IP addr should be a domain alias by default
- XCF-1996 - Display number of users in paging group
- XCF-2049 - Unnecessary code introduced in DomainManagerImpl.java
- XCF-1570 - Voice Mail Web UI should inform a user that he or she does not have a voice mail box if he or she does not have voicemail permissions.
- XCF-1778 - need easier way to add the same gateway to multiple dialing rules
- XCF-1798 - WebUI for ACD overflow destination needs to be cleaned up
- XCF-1803 - Backup & Restore: Progress and success message should be added for restore operation
- XCF-1841 - *.page and *.jwc files not ready for localization
- XCF-1867 - Add help text to Internal Dial Rule screen that explains how to use Exchange as the VM server
- XCF-1883 - automatically configure direct emergency gateways for phones that support it
- XCF-2050 - JAR message source does not use files with language variants in their names
- XCF-2057 - add ability to disable phonebook management
- XCF-2079 - Mediant 1000 should be changed to Mediant
- XCF-2103 - 'personal-auto-attendant' permission support
- XCF-2106 - Configuration tests should run automatically on first startup
- XCF-2115 - Improve "search for java" to enhance portability
- XCF-2137 - new color scheme for default sipXconfig look and feel
- XCF-2138 - search box improvements
- XCF-2139 - replace Home/Help/Logout link with icons
- XCF-2140 - improve help/quick links look and feel
- XCF-2141 - vertical navigation changes
- XCF-2142 - Login page improvement
- XCF-1144 - Allow AA to transfer to a SIP URI in addition to an internal extension
- XCF-1502 - Make voicemail web page the home page for user portal
- XCF-1844 - Intercom Feature screen help text state only Polycom is supported. Need to change for LG

- XCF-1873 - Improvement of the LDAP Import page
- XCF-1995 - Usability enhancements for Configuration Diagnostic page

Bug

- XCF-1230 - Able to add Duplicate permission Name, on system wide permission page
- XCF-1522 - New voicemail UI : strings and special characters, comma seperating 2 users can be added to Distribution list
- XCF-1569 - New voicemail UI : User without VM permission and an invalid user extn can be added to Distribution list
- XCF-1752 - Internal exception when filtering the Imported users in LDAP
- XCF-1354 - daily and weekly schedules do not account for DST
- XCF-1826 - WorkingTime: not all overlapped cases are detected
- XCF-1678 - Firefox Pwd Mgr mangles PIN on User Identification page
- XCF-1861 - Duplicate schedule names are accepted
- XCF-1874 - Dialplan activation AuthRules do not reflect gateway changes
- XCF-1783 - change true/false into localized "Active/Inactive" in the 'Device Files' table
- XCF-1836 - Failed in progress calls do not get cleared from the active call display
- XCF-1839 - Hunt Group: Last call does not go to voicemail, if sequence set is 'If no response'
- XCF-1851 - Backup & Restore: Not possible to upload the saved backup files
- XCF-1870 - Voicemail server hostname should be a required field for non-internal voicemail servers
- XCF-1875 - Voicemail mapping rules broken
- XCF-1877 - exception in search for phonebooks that contain entries with empty (null) fields
- XCF-1878 - user phonebook page look and feel needs improvement
- XCF-1905 - There is no user explanation text on how to use the Schedules for Dial Plan in UI
- XCF-1908 - Doing a Restore > Upload File from a PC with Internet Explorer 6.0 or 7.0 Fails
- XCF-1909 - Generated .../etc/sipxpbx/domain-config is incorrect
- XCF-1928 - Hunt Group: Calls go directly to the voicemail when Use voicemail check box is enabled
- XCF-1934 - Dial plan : Activating the dialplan hangs the config UI
- XCF-1942 - Dial plans with both fieldparams and headerparams are not schema-valid and don't work
- XCF-1654 - add 'overload queue' column to the the queue table
- XCF-1808 - Configuration files are not getting created for AudioCodes MP-FXO,AudioCodes Mediant 1000 and AudioCodes TP260 gateways in the TFTProot directory
- XCF-1900 - Internal exception while resetting the Dial Plans
- XCF-1930 - Internal exception in Configuration Diagnostic Tests page
- XCF-1945 - When Config Server activates dialplans, it restarts the servers multiple times
- XCF-1955 - adding external line to Cisco IP phone causes exception
- XCF-1964 - "/usr/bin/sipxconfig.sh --database reset-superadmin" gives error
- XCF-1355 - typo in label for Polycom "daylightSavings.stop.date" field
- XCF-1823 - Internal exception while activating dial plan
- XCF-1832 - LDAP preview: display error message instead of Internal exception
- XCF-1849 - ACD: Internal exception while Signing-in the ACD Agents under ACD Presence tab
- XCF-1876 - Caller Sensitive Emergency Routing not integrated with dial plan activation
- XCF-1885 - Changes made to Personal Auto Attenadant when logged in as admin does not reflected in the user portal and vice versa
- XCF-1888 - Internal exception when clicked on the Add Alias link under Manage Domain page
- XCF-1914 - Setting for "voice mail type" may not be initialized properly during upgrade
- XCF-1918 - Not possible to login to config UI after installing build 393011006
- XCF-1921 - LG devices the "Phone Password" field must be a minimum of 4 digits, it accepts 3
- XCF-1931 - monitoring UI disabled by default (even after sipxconfig-mrtg RPM is installed)
- XCF-1932 - Internal exception when Caller sensitive forwarding is activated
- XCF-1938 - Internal exception when sorting Service by status
- XCF-1949 - Fake errors are displayed when sipxchange is restarted from the console
- XCF-1976 - Paging Server config needs some changes
- XCF-1982 - cannot define multiple emergency dialing numbers for Polycom v2.x firmware
- XCF-1985 - IP address lable for polycom phones is not seen in the config UI
- XCF-1986 - CDR historic report page causes an exception if sipxproxy-cdr is not installed
- XCF-1990 - Internal Error seen when a Paging Group is added under Features>Paging Groups
- XCF-1994 - Adding Call Forwarding extension to one User adds the same call forwarding extension to all the users added
- XCF-2003 - Internal exception when a large number is added as Page Group number
- XCF-2012 - Deleting the schedule which is used for call forwarding does not display error message.
- XCF-2018 - exception when editing User (changing PIN, changing name etc.)
- XCF-2030 - Gateway accept the invalid format of IP Address
- XCF-2031 - Internal exception while opening the Dialplan.
- XCF-2043 - e911 routing asks for 407 Proxy Authentication
- XCF-2046 - Accepts duplicate Page Group numbers.
- XCF-2081 - Activating dialplan does not restart SIPXProxy process
- XCF-2097 - sipxconfig db setup skipped under rare condition
- XCF-2099 - ZipUploadTest fails when executed on NFS share
- XCF-1685 - Internet explorer gives an error while uploading the ACD audio(Welcome and queue audio)prompts for the first time
- XCF-1749 - Click OK button after clicking Apply prints error message in the Add Phonebook page.
- XCF-1767 - CDR stats: disabled "Filter by" option is selectable in IE and selecting it causes Internal Server Error
- XCF-1777 - Extract time pattern from the page locale
- XCF-1898 - Drop-down menus with strings that cannot be localized
- XCF-1992 - Configuration Diagnostics Tests screen throws org.apache.tapestry.StaleLinkException on automatic refresh
- XCF-2002 - Help text is missing in Localization screen
- XCF-2008 - Internal exception while adding an existing user.
- XCF-2009 - Incomplete help text in SBCs screen
- XCF-2010 - Call Forwarding: Error messages are not displayed when invalid inputs are given for forward to and ring for
- XCF-2059 - Editing the refresh time causes exceptions on auto refreshed screens.
- XCF-2082 - Tabs defined via the settings mechanism (XML files) remain "active" along with the main tab
- XCF-2085 - "Caller" column is not displaying uri of caller's number on ACD Historic Reports page.

- XCF-2110 - Services page: no Start/Stop feedback since rev 10635
- XCF-2112 - Generated domain-config is incorrect
- XCF-2122 - Cannot use "Use caller sensitive forwarding" function
- XCF-2146 - Internal Exception error when exporting phones with external lines
- XCF-2150 - Quick link on the "Devices/SBC" page should be to "Internet Calling" and not "Dialing Plans"
- XCF-2163 - LG Phones - Under sip parameters configuration parameter is named "Outbound proxy address". Should be "Proxy address"
- XCF-2164 - web tests fail on system w/o mrtg installed
- XCF-2165 - Many Admin E-Mails seen on new build system : ERROR: unable to open config file: /etc/mrtg/mrtg.cfg
- XCF-1742 - Additional E-mail address is copied to the next page when Create another user checkbox is selected
- XCF-1819 - Internal error ocured
- XCF-1848 - Polycom phones w/firmware 2.2.0 not receiving dial plan from sipxconfig
- XCF-1991 - Unable to login to the config UI after installing ecs-ingate-395008442-i386.iso build
- XCF-2052 - sipxconfig restarted a while after starting - sipxconfig 396011404
- XCF-2080 - all records duplicated in Agent Availability list, on customer's 'ACD Historic Reports' screen
- XCF-2158 - ACD: Not able to disable the "agents-wrap-up-time" parameter

Task

- XCF-1756 - freeze DB schema before 3.8 release
- XCF-1889 - Unable to declare Gateway objects via Spring configuration metadata
- XCF-2000 - Unit tests should not have a dependency on the database
- XCF-2091 - Configure identity for the ACD server
- XCF-2128 - freeze DB schema before 3.10 release