GRANDSTREAM NETWORKS

Firmware Release Notes
Firmware Version 1.0.8.6
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date October 16, 2015

CHANGES SINCE FIRMWARE RELEASE 1.0.8.4

BUG FIXES

- Fixed Dial plan prefix does not works when dialing a number from call history
- Fixed Currency XML application does not work
- Fixed Inverted LCD Hebrew Strings in Account Name and Incoming Calls
- Fixed Hide passwords from command line
- Fixed Admin password can be changed using end-user login
- Fixed Device rings dial tone after establishing conferences with the LINE key
- Fixed Pressing Voicemail button makes the phone Freeze
- Fixed SIP TLS Private key and certificate cannot be provisioned using XML
- Fixed Show SIP Message text content on LCD screen 3

ENHANCEMENTS

- Added "Delete All Contacts" Button on Phonebook
- Add ability to dial directly LDAP contacts via softkey or dial button
- Adjusted Hebrew Font to a smaller size
- Added ability to filter characters from dialed numbers
- Updated logo for web UI

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Firmware Version 1.0.8.4
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date May 6, 2015

SUMMARY OF UPDATES

The main purpose of this release is to address the remaining issues observed in previous releases.

NOTE:

Once upgraded to 1.0.7.8 and above, downgrading to any previous firmware version is not supported.

CHANGES SINCE FIRMWARE RELEASE 1.0.7.11

BUG FIXES

- Fixed phone crash issue associated with SIP OPTIONS
- Fixed phone sends MOH URI INVITE with incomplete From header
- Updated Hebrew language missing strings
- Fixed Secondary SIP Server was modified successfully after turning on provider lock
- Fixed SIP TLS Private Key and Certificate cannot be provisioned using XML file

- Fixed phone will use random port as REGISTER and INVITE port after unplugging and re-plugging network cable
- Fixed phone cannot receive incoming calls under certain conditions
- Fixed phone freezes on conference page after long time conference on PCMU
- Fixed phone behaves abnormally if entering audio loopback when enable HEADSET Key Mode
- Fixed Phonebook export/upload are accessible without authentication request
- Updated New Zealand time zone
- Updated Russian Time Zone settings
- Fixed phone does not stop the RTP after BYE and get mixed on subsequent calls
- Fixed changing configuration does not take effect under certain conditions
- Fixed rejecting a call on "Audio Loopback" page would result in sound lost
- Fixed volume setting is incorrect if adjusting it under headset mode
- Fixed phone fallbacks to PCMU upon re-invite during the session
- Fixed phone crashed after switching audio mode during a call
- Fixed DNS-SRV connection failed if first option is unavailable using TCP
- Fixed Phonebook containing Cyrillic characters is not sorted by alphabetical order
- Fixed phone cannot enter Audio Loopback when it is in headset mode
- Fixed audio channel in audio loopback is error after enabling Toggle Headset/Speaker mode
- Fixed there was no audio after 14min conversation when using SRTP enabled and forced
- Fixed phone crashes when configured with long dial plan after registering to SIP server
- Fixed phone drops calls on re-INVITE
- Fixed no dial tone and unable to make or receive calls after weekend
- Fixed SRTP with TLS SIP Transport - no voice speech path after secondary server is disconnected while having an active call on primary server
- Fixed TCP port 4 is open when logging in from phone UI
- Fixed no language cursor change when switching Auto/Downloaded to the language same as current one in LCD mode
- Fixed failed to change LCD language after factory reset
- Fixed pressing Line key when phone is downloading phonebook will lead abnormal display on LCD
- Fixed there is no audio channel icon and no sound on audio loopback page after entering and exiting audio loopback page continuously
- Fixed LCD display is abnormal when transfer a call
- Fixed phone plays busy tone in audio loopback mode after remote party ends the call
- Fixed failed to dial Voice Mail via MPK in dial page
- Fixed audio does not stop after sending number with * and no call feature is set

ENHANCEMENTS

- Added option to ignore Alert-Info header when used for distinctive ringtone
- Added dual Outbound SIP Proxy support
- Added support for sending SIP Option messages to verify connectivity to the SIP server
- Added option to select Call Pickup mode
- Added support of OPTION 160
- Enabled the HTTP(S) ID/Password prompt
- Added option to disable the Call park subscription
- Added support to check the status of registration to Action URI

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Firmware Version 1.0.7.11
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105

DATE APRIL 12, 2015

CHANGES SINCE FIRMWARE RELEASE 1.0.7.8

BUG FIXES

- Updated New Zealand time zone settings
- Updated Russian Time Zone settings
- Fixed phone does not stop the RTP after BYE and get mixed on subsequent calls

ENHANCEMENTS

- Added support to disable the Call park subscription
- Added support of DHCP OPTION 160
- Enabled HTTP(S) ID/Pass prompt

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Firmware Version 1.0.7.8
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date February 20, 2015

CHANGES SINCE FIRMWARE RELEASE 1.0.7.4

BUG FIXES

- Fixed potential security issue for unauthorized access to product configuration

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Firmware Version 1.0.7.4
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date November 4, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.6.11

BUG FIXES

- Fixed cannot change Web UI display language to Czech
- Fixed BroadSoft interop: CONF+BLF use Broadsoft ID instead of Extension number
- Fixed BroadSoft interop: TRANS+BLF use Broadsoft ID instead of Extension number
- Fixed proxy server failover with DNS NAPTR/SVR and TCP SIP transport failing
- Fixed Disable Call Waiting also worked in ringing status
- Fixed [Anatel]: BYE request sent to the wrong proxy/IP based on the route set
- Fixed crash after press transfer key in auto-attended transfer
- Fixed phone cannot send out TCP packets after changing IP
- Fixed phone doesn't display the complete extension in "dial" screen, if the extension consists of more than 10 digits
- Fixed phone does not play custom ring tone if phone receives a call on line 2
- Fixed when Ethernet cable is disconnected, "NETWORK DOWN" status will disappear after DHCP lease time passed
- Fixed phone failed to handle refer and early media during transfer with Genesis platform
- Fixed no audio after 14min conversation when using SRTP Enabled and Forced
- Fixed SIP Server and Outbound Proxy cannot be configured with host address like x.x.x.0
- Fixed BLF get stuck with Portaone Softswitch

- Fixed phone should not update the called party name from UPDATE message during an active call
- Fixed audio mode was wrong after recovering held call
- Fixed 3CX Call Park Retrieval (Shared Parking)
- Fixed high noise level when using RJ9 headset
- Fixed phone always display "loading user data" when viewing call history and pressing up\down\left\right arrow are all invalid
- Fixed Web page displayed abnormally after pressing Save button directly on web access page
- Fixed Fonax: new password in web access page can be reset after resetting phone
- Fixed need to wait for NOTIFY upon transfer
- Fixed phone does not apply the ZeroConfig Server Path coming from the SIP NOTIFY
- Fixed Device crashed after downloading config file
- Fixed Device crashed in Public mode when download phonebook after reboot
- Fixed Name on Account general settings page didn't work
- Fixed LCD wouldn't display MUTE icon when pressing MUTE button if "Enable Idle Mute" function has been enabled
- Fixed clearing Outbound Proxy will return to the default value
- Fixed Setting public mode to "No" does not take effect after reboot
- Fixed Device worked abnormally when log in Public mode
- Fixed LCD popped up login page after reboot in Public Mode
- Fixed Unregister on reboot option should only clear phone contact

ENHANCEMENTS

- Updated Spanish LCD language strings
- Added Multicast Paging support
- Added support to disable SIP NOTIFY Authentication
- Added support to Map Polish characters in phonebook entries

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Firmware Version 1.0.6.11
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date August 5, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.6.7

BUG FIXES

- Disabled RTCP temporarily to avoid the no audio issue only when SRTP is used
- Checked the LCD PN number to display network icon and network boarder icon
- Fixed HTTP GET provisioning request displays HTTP username/password in plain text
- Fixed phone shows IP 0.0.0.0 randomly
- [Siemens IOT] Added an option, Disable Multiple m line in SDP, to send only 1 m line or multiple m lines
- Fixed phone switched audio channel from handset to speaker automatically when initiating a quick conference call
- Fixed phone crashes on receiving incoming calls with remote-party-id header
- Fixed conference fails when using # as send key
- Fixed device worked abnormally after downloading configuration file
- Updated the wrong string number on the short string file (applies only to gxp1150/1160/1400)
- Added missing tool tip for flash writing option on web settings/call features page
- Added an attempt to fix random crash issue
- Fixed phone will do a Blind Transfer if "End Call" soft button is pressed during auto-attended transfer
- Fixed SIP NOTIFY with "resync" event rebooted the phone
- Fixed phone cannot dial out if starts with * in on-hook dialing mode

ENHANCEMENTS

- Added support for HTTP CTI feature
- Added support for inserting pauses into speed dials and phone book entries
- Added PC port VLAN support
- Added support for Svenska language
- Added phone power special feature to gxp116x/gxp14xx
- Added support for incoming subscription with dialog event to send NOTIFY once after 202
- Added time and date menu

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Firmware Version 1.0.6.7
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date May 5, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.5.58

BUG FIXES

- Fixed 3CX prompt not fully heard on the phone
- Fixed phone sends BYE if it does not receive ACK after 200 OK response for update
- Fixed GXP116x digit gets dialed twice randomly
- Fixed gxp21xx lost side tone under some conditions
- Fixed 3CXv12 CTI - Slow Dialing

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Firmware Version 1.0.5.58
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date February 21, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.5.32

BUG FIXES

- Fixed phone plays incoming call ringtone with low volume when volume level set to zero
- Fixed random crash with parsing NOTIFY messages with reg event
- Fixed Dial Plan cannot remove ' ' from LDAP phone number entry
- Fixed Click-to-dial feature cannot be disabled
- Fixed phone does not display the booting info normally when the Display language is Russian and some other languages
- Fixed phone sends Subscribe after timeout to wrong destination under certain conditions
- Fixed BLF call pick up does not use the remote elements
- Fixed phone does not fetch new firmware when firmware prefix parameter is changed
- Fixed time zone setting via config file does not take effect until after manual reboot
- Fixed callee cannot hear the DTMF voice when caller is using MPK to dial DTMF
- Fixed LDAP search result display is incomplete
- Fixed incoming calls are answered in mute mode under certain conditions
- Fixed phone ignores UPDATE message and does not change the calling party info in call transfer case
- Fixed EHS issues reported by Plantronics
- Fixed WEB UI does not show complete configuration of the Extension Board
- Fixed Music on hold not working. Added an web option for "Hold Method" (P value 2361)
- Fixed Fetch data using cURL vulnerability
- Fixed CNAM display has been reduced to 10 characters
- Fixed phone got stuck after upgrade/downgrade using HTTP MV-IPTEL server

- Fixed GenBand Interop: GXP21xx N-way Conference does not send out REFER
- Fixed phone displays the conference setup screen when phone is already in 3 way conference
- Fixed phone displays "Talking To" when using other languages
- Fixed phone does not take its time from SIP REGISTER message
- Fixed phone plays dial tone when switched audio mode from EHS to handset
- Fixed some info display error on web page
- Fixed incorrect Chinese translation for headset

ENHANCEMENTS

- Added support for Second Dial Tone
- Added support to send A,B,C and D characters as RFC2833 DTMF
- Added Phonebook sorting option in Web UI
- Added support for Public Mode auto logout
- Added RTCP support
- Added option to enable/disable Crypto life time when using SRTP
- Added support to display PAI in UPDATE
- Changed from sendonly to inactive on hold INVITE SDP with new option "Hold method" with RFC 3264
- Updated OenSSL CA bundle
- Updated LCD and Web UI language string files
- Added Handset RX Gain (dB) for GXP2124
- Added handset TX gain 12/18 dB
- Added UCM active feature code support
- Added UCM server detection feature.
- Added support to display NTP server information on both LCD and web UI

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Firmware Version 1.0.5.32
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165
Date September 27, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.5.26

BUG FIXES

- Fixed inverted LCD Hebrew string display
- Fixed audio is cutoff upon answering a call
- Fixed phone cannot set Japanese via Web interface
- Fixed phone is not able to resume the call using hold button if headset softkey is pressed on hold call
- Fixed Contacts are not synchronized between phonebook on Web interface and phonebook on phone when changing phone to public mode
- Fixed When there is an account registered, after cancel a direct IP call, callee still rings
- Fixed automatic Upgrade (daily or weekly) is not obeying the set parameters
- Fixed check New Firmware Only When F/W pre/suffix changes function does not work
- Reduced audio delay when SRTP is in use

ENHANCEMENTS

- Added warning window display when key pad is locked with star key
- Added "-" in input string
- Added HTTPS support for WebUI access
- Changed "Network" to "Network Config" in LCD main menu
- German translation update (GUI/LCD)

- French translation update (GUI/LCD)
- Russian translation update (GUI/LCD)
- Hebrew translation update

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Firmware Version 1.0.5.26
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165/1100/1105
Date: June 28, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.5.24

BUG FIXES

- Fixed phone GUI crash when login under public mode if the records of call history are more than 200
- Fixed phone will block when it's called via sipp scripts
- Fixed Using configured IP in DNS mode is invalid
- Fixed Pressing left arrow key in Phone Book page ,it will come back to idle screen
- Fixed phone can log out under public mode via 3CX command when it is in a call
- Fixed calls from call log bypass the dial plan check
- Fixed phone turns on line LED with idle screen on pressing redial key when the redial number is set to *30123
- Fixed parked call pickup does work
- Fixed phone stuck in BLF RED Solid state
- Fixed GXP2124 crashed in idle stage
- Fixed CPE sends incomplete info to ACS server
- Fixed 3CX CTI issue when using * and #
- Fixed TR-069: Incorrect response for GPV for Device.X_GRANDSTREAM_Keys.MultiPurposeKeys.MultiPurposeKeysNumberOfEntries
- Fixed TR-069: SPV for time zone not applied correctly using UTC+0100
- Fixed TR-069: SPV for Nederlands language not working
- Fixed soft key displays error on Manage Group page
- Fixed downloaded language file not loading automatically after rebooting
- Fixed phone sends CANCEL to SIP server instead of OBP when "Remove OBP From Route" is set to "Yes"

ENHANCEMENTS

- Added the volume control support on headset
- Added "Disable Notify Authentication" when there is no SIP account configured
- Added support for CTI Feature
- Added support to fast blink the voicemail indicator LED for incoming calls for GXP11xx

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Firmware Version 1.0.5.24
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1160/1165
Date: April 29, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.5.23

BUG FIXES

- Fixed HTTP GET request uses HTTP 1.1 causing XML provisioning fail
- Fixed WEB limit for SIP TLS Certificate
- Fixed phone will crash after uploading a file size of 10M
- Fixed phone crashes when download ring file

- Fixed phone canceling another call will crash after Auto-Attended Transfer
- Fixed phone won't boot up after incomplete firmware file download and upgrade

ENHANCEMENTS

- Added ability to enable/disable LLDP and/or LLDP-MED
- Added option to enable/disable MUTE key as DND

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Firmware Version 1.0.5.23
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1100/1105/1160/1165
Date: April 2, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.5.15

BUG FIXES

- Fixed LLDP feature not working
- Fixed Account 5 ring timeout on web refers to the wrong pvalue
- Fixed phone plays ring back tone after call is answered
- Fixed phone stuck on white screen after firmware upgrade under certain conditions

ENHANCEMENTS

- Added 3CX CTI support
- gs_cpe updated to 1.0.1.31
- Removed SIP user ID and SIP server's information from the status page
- Added support for contacts with "#" and "*" in Phone Number field on Web GUI

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Firmware Version 1.0.5.15
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1100/1105/1160/1165
Date: December 31, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.5.14

BUG FIXES

- Fixed 3-way conference not working on GXP110x with incoming calls
- Fixed BLF doesn't update LED status after rebooting GXE
- Fixed GXP21xx phones stop on aux when doing auto upgrade

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Firmware Version 1.0.5.14
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1100/1105/1160/1165
Date: December 17, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.4.23

BUG FIXES

- Fixed BLF LEDs do not light up on EXT Board
- Fixed GXP116x does not show status for click to dial function
- Fixed BroadSoft Interop: Call Features configuration blocked for Shared Line account
- Fixed Broadsoft Interop: Authenticate Incoming Invite option does not exist in web UI
- Fixed MWI does not light up after reboot when there are voice mail message(s)

- Fixed BroadSoft Interop: Feature Key Sync, Call Forward Always Ring Splash shows wrong missed call number
- Fixed P value is not using number for HTTP/HTTPS proxy
- Fixed phone stops on aux when doing auto upgrade performance.
- Fixed downloaded language file does not load automatically after rebooting
- Fixed missing 3 way conference option in GXP110x
- Fixed "Disable" function of "Click-To-Dial Feature" does not work
- Fixed only account 2: WEB will display blank value when configuring "Use P-Preferred-Identity Header " with "Yes"
- Fixed continuous clicking "save" for a few times results in adding many same contacts
- Fixed display Language on Language page error
- Fixed phone does not send 200OK on time on incoming calls
- Fixed group name is overlapped when its name is very long
- Fixed year of system time displays incompletely
- Fixed phone behaves abnormally when uses EHS to answer second call
- Fixed phone not sending RTP when voice code choose ILBC and the iLBC frame size choose 20ms
- Fixed phone not handling RFC3891 correctly
- Fixed IP displayed as 0.0.0.0 on telnet status and Phone GUI when VLAN Tag is used
- Fixed phone loses VLAN tag randomly and loses the network connection
- Fixed HTTP/HTTPS language file downloading does not use authentication
- Fixed phone does not send its certificate for MutualTLS authentication to download language file/XML Idle Screen and Phonebook
- Fixed LDAP Number Filter's value can't change to any other value when configuring LDAP Name Filter together
- Fixed LDAP Name Filter and LDAP Nunber Filter are all changed to a syblem (when configure LDAP Name Filter with (&(telephoneNumber=5384)(userPassword=1234))
- Fixed TR-069: Connection Request Username/Password fields don't accept dash
- Fixed DTMF cannot be recognized when pressing fast with duplicate digits
- Fixed phone sends back multiple codecs in 200 OK
- Fixed GXP110x do not send invite after pressing MPK which configure with 3-way conference to invite the third party
- Fixed PhoneBook entries in Arabic are not displayed correctly
- Fixed some contacts with special chars seem empty in phonebook
- Fixed LCD display error account info under public mode
- Fixed phone soft key display abnormal after some operation
- Fixed GXP110x does not have 3-way conference options
- Fixed GXP2100's SwitchSCR soft key displays incompletely after downloading and erasing SCR XML
- Fixed phone will not play a "ring tone" when the first call is on hold via pressing CONF button ,what it plays is "call waiting tone"
- Fixed GXP1400 V1.6A phone's PC port fails to connect to RTL8319 PCI Ethernet card
- Fixed MPK doesn't display correct status when server send "notify-state" instead of "state"
- Fixed phone does not use the complete user name configured in the LDAP bind Request
- Fixed LCD don't display the option edit and dial when we do search on phone book
- Fixed phone behaves abnormally when phone is at dialing status and answers a call
- Fixed phone GUI crashes when delete idle screen and during this time answers a call
- Fixed LCD doesn't change to answer screen when downloading idle screen and there is an incoming call
- Fixed update configuration on time zone results in a short-lived time
- Fixed changing language causes white screen display
- Fixed "Grandstream" characters on web appear and disappear after phone's configuration updated
- Fixed call feature is still set after the input number is cleared
- Fixed soft key doesn't work during some special operations
- Fixed can't hold the third party in the progress of making a conference
- Fixed creating new groups causes GUI freezing sometimes
- Fixed alert-info doesn't work if using name

- Fixed user needs to press MUTE twice to unmute
- Fixed line status change error when making a call from web UI
- Fixed web UI does not display the second call's status
- Removed "Downloading SCR XML" message when refreshing screen with SIP Notify
- Fixed Manage Group interface does not have dial soft key, so we can't make a call directly from Manage Group interface
- Fixed P value is not using numbers for HTTP/HTTPS proxy

ENHANCEMENTS

- Added support for Music on Hold via remote URI
- Added suggestions for Admin setting and end user setting options
- Added support for more 802.1x Authentication methods
- Added STAR key Keypad locking Settings on Phone's GUI
- Added a new P value to Enable/Disable "click to dial" function
- Added displaying page up/page down only if we cannot display the list within one page
- Added new web interfaces for configuration
- Added support for “Accept Incoming SIP from Proxy Only"
- Added display support for Power Source. If it is supported, a new option will be shown under LCD Status MENU
- Added support to update display after transfer
- Added "Phonebook" soft key on call screen
- Added support for DND button to be configurable from web GUI
- Added support to Increase/Decrease Stutter Tone gain
- Added support for prefix or postfix in ring files request
- Added support for special characters to be part of any password protection
- Added support to “Authenticate Incoming INVITE”
- Added "ringtone" instead of "call waiting tone" for incoming call while a first call is on hold
- Added IP information display on public mode login screen
- Added support to Increase/Decrease Call Waiting Tone gain
- Added time zone for Panama
- Added hard key support to exit XML Application on GXP2124
- Added feature only allow one call can use G729 codec in a conference
- Added BroadSoft call center feature

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Firmware Version 1.0.4.23
Name of Product GXP2100/2110/2120/2124/1450/1400/1405/1100/1105
Date: September 24, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.3.30

BUG FIXES

- Fixed phone does not send its certificate for MutualTLS authentication to download language file/XML Idle Screen and Phonebook
- FixedGXP1400 V1.6A phone's PC port fails to connect to RTL8319 PCI Ethernet card
- Fixed IP displayed as 0.0.0.0 on telnet status and Phone GUI when VLAN Tag is used
- Fixed Phone loses VLAN tag randomly and loses the network connection
- Fixed Phone does not authenticate itself with HTTP/HTTPS credentials when downloading language file
- Fixed rc.network.ipv6 overwrites wan_device with 'eth0' and ignores configured VLAN tag
- Fixed pressing digit on transferring and conference interface will cause LCD display some wrong icons
- Fixed GXP21xx should not have the 3_way_conference function for key mode or MPK
- Fixed MPK doesn't display correct status when server send "notify-state" instead of "state"

- Fixed pressing Page Up and Page Down softkeys are all invalid on call history and phonebook
- Fixed gxp21xx phones request gs_phonebook.xml instead of phonebook.xml on downloading phoenbook xml file

ENHANCEMENTS

- Keep missed call status icon after power cycle
- Added configuration parameter to block/hide DND from menu Preference → Do Not Disturb
- Changed language file name to language.txt
- Changed phonebook file name to phonebook.xml
- Changed idle screen file name to idle_screen.xml

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Firmware Version 1.0.4.9
Name of Product GXP2100/GXP2110/GXP2120/GXP1450/GXP1400/GXP1405/GXP1100/GXP1105
Date: July 3, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.3.30

BUG FIXES

- Fixed picking up the handset cannot answer the second call when pressing conference button
- Fixed phone reboots slowly when network cable is disconnected
- Fixed Caller ID not displayed on ringing status
- Fixed pressing line which configured key mode is speed dial via active account doesn't have any response
- Fixed phone gets in a reboot loop when using certain invalid values in config file
- Fixed XML application: Line key needs to be pressed twice to show call screen
- Fixed GXP14XX sending currency and stock updates requests
- Fixed phone can't delete the second entry when deleting an entry on phone book during a call
- Fixed LCD displays abnormaly when building a conference via MPK
- Broadsoft interop: Fixed GXP21xx does not display AoC information
- Fixed the icon of Feature Key Synchronization do not synchronous between phone and server
- GXP2124: Fixed phone displays hold softkey when set Auto_Attended transfer during a call
- Fixed DND function does not work after LCD shows the update warning widnow
- Fixed LCD displays abnormaly when line key is used as MPK and does other operations
- Fixed a redundant colon dispaly on web page
- BroadSoft interop: Fixed phone fails to update "No Answer Timeout" based on Feature Key Sync NOTIFY
- Fixed phone does not limit pressing letters when filter condition is “ Numbers"
- Fixed cancel softkey doesn’t work after pressing speaker on the building 4WC screen
- Fixed phone appears freeze after reboot under certain conditions
- Fixed GXP11XX: phone crashes when transfer to an non-exist number when phone starts Auto_Attended transfer
- Fixed MPK which configured with LDAP Search does not support filter contains &
- Fixed phone does not have ring tone and LED of MPK light turns off after Call Park
- Fixed GXP14XX: phone choose Arabic language , more softkey display black part on idle status
- Fixed GXP14XX: DND button do not work sometimes
- Fixed phone can't download phonebook with public mode
- Fixed phone GUI crashes when Auto_Attended transfer is enabled and some operation is done
- Fixed phone LCD does not display Reconf softkey when conference was put on hold
- Fixed at stock screen, the left arrow can't realize flipped over function
- Fixed phone audio loopback don't work when pressing Mute button during a call or during dialing
- Fixed phone stays at 'reading configuration' for about 40s after reboot

- Fixed cancel softkey does not work when continuously pressing CONF button and cancel softkey after phone builds a conference
- Fixed phone stays at 'reading configuration options' screen when seting configuration about idle screen
- Fixed Public Mode does not correctly
- Fixed GXP2120: MUTE icon does not disappear on dial status when exiting the function
- Fixed softkey displays abnormaly during a call
- Fixed phone requests gxp.txt using HTTP even if HTTPS is selected
- Fixed XML Application: AppendInputURL misses "&" to append input to existed variable
- Fixed phone freezes when download phonebook and auto answer a call
- Fixed phone on hook causes an abnormal result during call transfer
- Fixed some of the GUI update issues
- Fixed HTTPS option not available in upgrade via menu
- Fixed GXP21xx does not restart phone on backup SIP server change
- Fixed pressing dial softkey is invalid with special configuration
- Fixed pressing end call softkey is invalid when adding pressing phonebook button during calling
- Fixed cannot dial via voicemail key when phone is offhook
- Fixed phone crashes when IP server path write IP and certain special character
- Fixed pressing AnswerCall softkey or RejectCall softkey for second incoming call are all invalid
- Fixed GXP1400 shows unsupported option BLF/evenlist BLF/Presence Watcher from Key Mode under basic settings page
- Fixed GXP21xx crashes when a user keeps doing hold and unhold operations by switching lines
- Fixed phone LCD does not show date information when the P value about it set as 5
- Fixed GUI crashes when LDAP Display Name is configured with a few same valid values
- Fixed phone crashes when incoming INVITE and outgoing SUBSCRIBE at the same time, they got same transactionID
- Fixed LCD switches language, stays at "language switching" screen
- Fixed part of LCD show white screen when making a call via call history
- Fixed x-gs-screen does not work using SIP NOTIFY
- Fixed GXP21xx takes 3 seconds to send REFER when Semi-Attended transfer is performed
- Changed minimum weather update interval to 5 min, default update interval to 15 min
- Reversed softkey label 123<->ABC IPv4<->IPv6
- Added web GUI->Advanced settings, Layer 3 QoS does not have valid range of values or error message when the value is invalid
- Fixed GXP2110 web page can't save the MPK function after update
- Fixed Jitter buffer options missing in web GUI
- Fixed GXP2124/2100 PoE reboot from ringer & HF high current
- Fixed GXP21xx does not play busy tone after the last key release if dial plan does not match (e.g. 123*)
- Removed web validation for 802.1p priority value to check if VLAN Tag is greater than 0
- Fixed network activity lost when changing the 802.1p value

ENHANCEMENTS

- Added LDAP phonebook feature
- Added LLDP feature
- Added 3CX feature request: Click to dial
- Added support to dial out in MPK with alphanumeric character set
- Added option to allow phone to go back to idle status after the configured time on off-hook state
- Added HTTP/HTTPS Proxy options in basic configuration page
- Added web option to disable Telnet
- Added web login wrong password counter
- Add more strict check for "request line" to avoid ring after received invalid INVITE
- Added support to program TRANSFER button to do in call DTMF
- Added support for "Match Incoming Caller ID" pattern in distinctive ringtone
- Added a softkey "RejectCall" and "AnswerCall" for second incoming call

- Added more methods for input in XML application
- Added support for drop down list in XML application
- Added login session timeout window to allow extending or terminating web session
- Added phone and line status display to the web with basic line operations
- Added web direct call dialing
- Added new telnet command "phone_status" to get current phone status of line, screen and hookswitch.
- Added download via HTTPS for XML phonebook and XML idle screen
- Added web page favicon
- Added an option for the phone book key to do LDAP search
- Added support to append variables to URL in xmp application
- Added support for HTTPS for XML application
- Added User-Agent in HTTP Get request when downloading: Phonebook, Idle screen, language file.
- Added Turkish language support
- Added support for generic XML config file
- Added support so phone book can display eight Chinese characters on first name and last name
- Allowed using certain values in xml idle screen/xml application
- Extended the length of Firmware Server Path, Config Path and XML fields

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Firmware Version 1.0.3.30
Name of Product GXP2100/GXP2110/GXP2120/GXP1450/GXP1400/GXP1405/GXP1100/GXP1105
Date: April 25, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.1.110

BUG FIXES

- Fixed phone does not get provisioned/upgraded when using domain name, new base file version 1.0.3.19
- Fixed a few crash/freezing issues
- Fixed possible memory leak when BLF key is cleared
- Fixed pressing EHS button twice needed to get a dial tone
- Fixed potential memory leak when NOTIFY for BLF or presence has invalid body
- Fixed phone getting slower and unusable in idle caused by memory leak
- Fixed TRAN+MPK cannot do Blind Transfer when MPK is configured as eventlist BLF
- Fixed display issue when using built-in Czech language
- Fixed display issue when using German
- Fixed phone entry display issue when using Russian
- Fixed cannot dial via voicemail key when phone is offhook
- Fixed device crash when download P value set as Special characters
- Fixed modify basic settings account option doesn’t take effect on GXP140X web UI
- Fixed SCA mode: Device have an abnormal result when do Fwdall operation
- Removed irrelevant Web UI options on GXP110x/14xx
- Fixed phone automatically changes speaker mode to handset mode under certain conditions
- Fixed phone crashes right after booting up when using TCP or TLS
- Fixed IM alerts and incorrect display for IM on GXP1400
- Fixed handset hissing noise in GXP110x/14xx
- Fixed cycle reboot when provision on 3CX

ENHANCEMENTS

- Improved efficiency of handling escape characters

- Added support for updating display after transfer
- Added a softkey "RejectCall" and "AnswerCall" for second incoming call
- Changed minimum weather update interval to 5 min, default update interval to 15 min
- Changed phonebook download interval to 5 min, 0 for disable automatic downloading
- Changed call screen/onhook screen GUI design
- Added support for XML application
- Added support for IPV6
- Added support to DHCP option 120
- Switched to use new gs_cpe for TR-069 support
- Added voicemail softkey on the LCD
- Added support for Paging
- Added option in MPK to do Intercom
- Added support for option "Transfer to" in BLF key mode for GXP21xx
- Added support Czech language
- Added call mode display on GXP2100 and GXP1450

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Firmware Version 1.0.1.110
Name of Product GXP2100/GXP2110/GXP2120/GXP1450/GXP1400/GXP1405/GXP1100/GXP1105
Date: January 5, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.1.108

BUG FIXES

- Fixed GXP2120 freezing with extension board when receiving multiple incoming calls continuously
- Fixed 3-Way Conference cannot disconnect conversation under certain conditions
- Fixed phone cannot hold/resume after another incoming call if "SIP Server" and "Secondary SIP Server" are configured the same
- Fixed phonebook entries not sorted after downloading phonebook XML file
- Fixed phone book does not display last name
- Fixed phone does not display registration icon when SIP registration option is disabled
- Fixed dial tone or DTMF tone can be heard in calling status
- Fixed phone crashes when pressing DTMF at the time of another call coming in
- Fixed phone would slow down and eventually crash with certain 301/302s
- Fixed GXP2110 freezes when idle for some time
- Fixed not displaying holding line's information when line2 is on hold and if there is incoming call on line1 on GXP1400

ENHANCEMENTS

- Added Handset TX gain web control
- Added configuration file save/download from web GUI
- Added password confirmation on web
- Added an option to ring both the Speaker and Headset when the phone is set to Headset mode
- Added account name on display for GXP140x

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Firmware Version 1.0.1.108
Name of Product GXP2100/GXP2110/GXP2120/GXP1450/GXP1400/GXP1405/GXP1100/GXP1105
Date: November 12, 2011

CHANGES SINCE FIRMWARE RELEASE 1.0.1.83

BUG FIXES

- Fixed phone crashes after receiving BS eventlist Notify message
- Fixed GXP1400 cannot do factory reset correctly
- Fixed BroadSoft interop: Directed Call Pickup fails when MPK User ID is not configured as extension number
- Fixed GXP2100 failed to lookup Caller Name from Phonebook
- Fixed Reconf softkey is invalid
- Fixed device requests a wrong name when download a phonebook xml or screen xml
- Fixed WEB page diaplays lack a sentence after updating
- Fixed message softkey displays "HANDSET" in Diagnostic Mode
- Fixed Monitor mode choosing Eventlist BLF and BLF causes LED show abnormal
- Fixed BLF related performance issue
- Fixed cannot remove existing/duplicate LDAP directory entries
- Fixed BroadSoft interop: GXP140x fails to display both Caller ID name and number
- Fixed BroadSoft interop: GXP140x Phone crashes after updating "Line Key" mode
- Fixed SRTP keylength32 not setting correctly, leading to interop failure with Snom
- Fixed GXP1450: Phone stops playing audio if CODEC is changed
- Fixed high pitch noise when receiving invalid CN packets
- Fixed [TR-069] SetParameterValues problem Device.VoiceService.1.VoiceProfile.1.SIP.RegistrarServer
- Fixed phone crashes when receiving inactive in early media
- Fixed phone sends wrong register request after network recovery
- Fixed Session-timer not working as expected (caller doesn't go back to idle)
- Fixed phone web UI shows HTML code when input "<>"
- Fixed cannot send the number via "Send" button
- Fixed IP address isn't come back to 0.0.0.0 when pull up phone's cable
- Fixed Transfer fails when setting Dial Plan Prefix
- Fixed audio quality issue under certain conditions

ENHANCEMENTS

- Enhanced Blind Transfer operation, no SEND key is required at the last step
- Added support on configuring feature code on Multi-Purpose Keys
- Changed date format to day-month-year
- Added flash key support for 3-way conference on GXP11xx
- Added support for two accounts for GXP1400/1405
- Added fixed Jitter Buffer support
- Added support for "Check SIP User ID for incoming INVITE"
- Added 802.1 setting validation on the web

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Firmware Version 1.0.1.83
Name of Product GXP2100/GXP2110/GXP2120/GXP1450/GXP1400/GXP1405
Date: July 26, 2011

CHANGES SINCE FIRMWARE RELEASE 1.0.1.66

BUG FIXES

- Fixed Attended Transfer sometimes fails with OpenSER
- Fixed GUI crash when input last name or first name or phone number until user cannot add
- Fixed all Call History will disappear when Call Log is set as Disable
- Fixed cannot unlock the unit after setting provider lock option

- Fixed phone freezes when getting a call from gxp280 with Chinese SIP display name
- Fixed phone freezes when input non-UTF characters
- Fixed 3CX Interop : GXP21xx TLS issues
- Fixed phone crash with Call Park
- Fixed no IP shown under Status page when using VLAN
- Fixed soft key "SignIn" changed to a black rectangle (occurs on gxp2120/2110/2100)
- Fixed cannot return to idle status via speaker button or EndCall soft key
- Fixed ACS Connection Request Password field shows its value
- Fixed phone displays registration true when account registration is not enabled
- Fixed LCD shows speaker icon after pressing *30 under onhook mode
- Fixed phone does not recognize 96 as a valid dynamic payload type number
- Fixed two transfer records after doing a transfer
- Fixed GUI crash during language switching
- Fixed phone might force to reboot when GUI is too busy on handling other operations
- Fixed GUI crash during web UI update while on a call
- Fixed memory leak due to the SIP Account configuration reload after updating from web
- Fixed white screen on reboot (core)
- Fixed GXP21xx : Last digit in star codes not displayed on screen
- Fixed SIP/TLS, SIP/TCP SIP parser failed to respond to incoming SIP messages
- Fixed on hook dialing * function display unusual
- Fixed unconditional transfer cause one-way radio
- Fixed press an idle line when conversation used SRTP cause GUI crash
- Fixed missed call soft key displays after deleting the record
- Fixed phone response slowly when there are mass of phonebook records or call records (lowered records to 500 for GXP1400)
- Fixed * and # cann't be used as user ID in MPK
- Fixed noise when used SRTP
- Fixed Polish language display is bold and with missing characters after reboot
- Fixed Vol=0 passes small voice and small tone
- Fixed LCD freeze when conversation used iLBC

- Fixed Hold and Unhold cause one-way radio
- Fixed GUI crashed when pressed Backspace(when there exist some numbers) in Factory reset option, also fixed GUI crashed when pressing backspace+cancel softkey in Config Server input
- Fixed phone uses speaker instead of handset when 2 accounts members of same ring group registered in same phone
- Fixed we need press flash two times to change lines
- Fixed Dial DTMF does not work
- Fixed GXP1450: Phone stops playing audio if CODEC is changed
- Fixed GXP2100 (possibly others) crashes on call-waiting call when the caller exists in the phonebook (only specific case)
- Fixed Phone keeps playing Ring Back Tone even after receiving 200 OK
- Fixed Idle screen softkeys disappear when configuring ACS setting
- Fixed Handset/Speaker icon wrongly displayed while picking up with AnswerCall key
- Fixed callreturn can't be saved and doesn't work after reboot
- Fixed Recalibrate sidetone gain
- Fixed still appear missed call softkey after removed all call records
- Fixed language doesn't display fully and cannot change language from English to others
- Fixed dimly lit LED LINE2 during boot on GXP1450
- Fixed cancel softkey is not working and LED does not turn off
- Fixed re-set SRTP option only when the phone has ReferDialogId
- Increased maximum allowed SIP message size from 10k to 64k (allowing handling large eventlist NOTIFY)
- Fixed uninitialized variable in SRTP receiver thread。 Add warning on decrypt error
- Fixed we send BYE to target when transferee hang up a call before target answers
- Fixed phone stuck and cannot reboot when provision happened in dialing mode

- Fixed RTP reconfiguration bug
- Fixed no audio after a failed transfer if BYE came before end of new call
- Fixed LED for line 1 is in green, even though we press line 2
- Fixed GXP2100 displays duplicate CallerID when a matching number entry exists in phonebook.
- Fixed 3CX PnP feature
- Fixed the multicast PnP SUBSCRIBE does not go out if first account is set to use non-UDP transport (TCP or TLS)
- Fixed the multicast PnP SUBSCRIBE does not go to the multicast address if first account is configured with an outbound proxy server
- Fixed Removed Manually Edited Entries on Reboot
- Fixed couldn't show 2nd DNS server address on GUI
- Fixed phone crash when receiving 423 for BLF/Presence SUBSCRIBE
- Fixed when two or more keys are pressed at the same time might result the phone unable to handle the correct key release event
- Fixed long digit tone issue after release of key press.
- Fixed wrong dial plan input error from web might erase the value without reset it to default
- Fixed occur one-way audio and caller give up the call itself
- Fixed web double colons showed under first voice decode selection issue
- Fixed IVR notification for network address issue
- Fixed can hear custom tone after call established
- Fixed during call, after "Downloading" warning window, call screen is not responsible.
- Fixed cannot switch to PCMU when RFC2833 event payload type is not configured
- Fixed LCD config menu sip account page always display account in English
- Fixed account name does not translate after switching language if account name is empty
- Fixed gxp21xx to display the missing strings in translated language file to display in English string

ENHANCEMENTS

- Added easy conference feature
- Add configuration option to choose SIP/SIPS when using SIP/TLS
- Extended the length of Firmware Server Path field
- Added support to protect ACS and SSL fields when provider lock is set
- Added volume indicator on the screen when adjust volume using volume up/down keys
- Added Keypad Lock feature on GXP1400
- Added "Call Features" menu option to display Call Forward status for all accounts
- Added 3rd line display to GXP1450 when phone is ringing and has only one line in use
- Added support for LED displaying fast blinking green for eventList BLF initiator/proceeding state under BroadSoft mode
- Added support for displaying Message Waiting Count in LCD
- Added functionality to delete all call history at once
- Added IPv4 network address validations to web configuration page
- Added blinking MSG LED during provision
- Added RTL support in idle screen XML
- Added RTL support in web configuration
- Allowed the use of Redial key as Send key when there is already any input in the dial buffer
- Added syslogs for displaying IP failure after getting IP using PPPoE
- Added support for Chinese City name in weather screen
- Added support to override the re-registration time
- Added support for Broadsoft Feature Key Synchronization
- Added support for BroadSoft Network Conference
- Added support for Arabic language
- Added support for Hebrew language
- Added support for x-gs-screen event from SIP NOTIFY message in GXP21xx/1450
- Added scaling to reduce Handsfree 'PuPu' sound
- Added support for PIN Lock feature
- Added support for "Search" function in phonebook

KNOWN ISSUES

- XML Application is not working

CHANGES SINCE FIRMWARE RELEASE 1.0.1.64

- Fixed implied transfer when switch lines (Fixed sometimes will show IdleScreen and line1 is green)
- Fixed cannot input a right alphanumeric as user ID in MPK after received 'Enter alphanumeric'
- Fixed problem for AutoCF-2, AutoCF-10, AutoCF-11
- Added Hungarian, Polish and Slovenian support
- Fixed no audio on 5th call for GXP2120.
- Fixed session timer is not set correctly when receiving re-INVITE with replaces
- Added Syslog for dialog XML entity URI parsing error.

CHANGES SINCE FIRMWARE RELEASE 1.0.1.62

- Fixed GUI will crash if set name to more than 30 special character in web page
- Fixed GUI crash when doing transfer
- Fixed use special Chinese characters as DTMF cause transferor and transferee's GUI crash
- Fixed phone will reboot if fresh web page when use Firefox
- Fixed cannot save account configuration via MENU
- Fixed potential memory leak when downloading phonebooks with very long strings

CHANGES SINCE FIRMWARE RELEASE 1.0.1.61

BUG FIXES

Changes:

- Add 489 handler for subscriptions. We will cease retrying SUBSCRIBE if we receive 489 for that event.
- Fixed MPK Dial DTMF with special keys cause LCD lock up
- Protect against calling string functions with NULL strings.
- Added syslog to verify reboots are indeed triggered
- Fixed GXP1450 has display issue on idle screen and call screen
- Fixed LCD doesn't display error message when offhook using handleset, if "Offhook Auto Dial" number doesn't match dial plan.
- Fixed bad voice after switch from G.726-32 IETF packing to ITU and back to IETF—changing the packing mode will now work without a reboot (listed as a known issue in 1.0.1.61) which would have been a problem if you have one account using IETF and another using ITU.
- Fixed displaying overlap (add options under call log menu)
- Added output keypress syslog to capture the actual keypad event
- Fixed iLBC crash issue with large ptime
- Fixed poor audio quality with ptime larger than 60
- Fixed LCD temporarily freezes on 4-way conference hang up until an incoming call comes in
- Fixed BLF keys does not work properly
- Fixed cannot set time

ENHANCEMENTS

- None

Changes Since Firmware Release 1.0.1.59

Bug Fixes

- Fixed iLBC crash issue with large ptime
- Fixed poor audio quality with ptime larger than 60
- Fixed LCD temporarily freezes on 4-way conference hang up until an incoming call comes in
- Fixed BLF keys does not work properly
- Fixed cannot set time

Enhancements

- Added support for G.726-32 Packing Mode

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Changes Since Firmware Release 1.0.1.56

Bug Fixes

- Fixed Italian translation issue reported by customer
- Fixed LCD freeze issue under certain conditions
- Fixed freeze issue when download or clear Custom SCR
- Fixed GXP2120 no audio after multiple lines on HOLD
- Fixed time zone default setting problem—after factory reset it should be set to “Automatic” with fail-over default set to Eastern Time

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Changes Since Firmware Release 1.0.1.26

Bug Fixes

- Fixed GXP21xx Phone doesn't display weather in Celsius
- Fixed phone crash when disconnection network link under weather page
- Fixed cannot see the numbers when dialing to establish a second conversation
- Fixed Transferred Calls do not display name from phonebook
- Fixed DHCP Option 66 not working
- Fixed HTTPS mode cannot save
- Fixed BroadSoft interop: add support for displaying Advice of Charge in LCD
- Fixed BroadSoft interop: phone fails to auto-answer a click-to-dial second call from Call Manager
- Fixed XML Phonebook with 10+ digits causes phone to freeze and not boot-up
- Fixed icon display error under certain conditions
- Fixed after reboot two times, phone changed getting IP form PPPoE to DHCP
- Fixed GUI crashed when press "fwd all" softkey and onhook dialing
- Fix provision starts while call in progress. Wait until all calls finish before performing provisioning
- Fixed onhook dialing caused GUI crash
- Fixed GXP1450 cannot unlock keypad under NULL password
- Fixed update configuration many times caused GUI crash
- Fixed GXP1450 dial phone number caused it disappear
- Fixed XML phonebook previous entries not deleted
- Fixed cannot download Ring Tone by HTTP
- Fixed SUBSCRIBE expiration was different from Register Expiration
- Fixed HTTP upgrade reboot loop
- Fixed phone crashed on pressing menu key
- Fixed unable to display remote number for incoming/outgoing calls (under onhook dialing mode)
- Fixed phone crashed on receiving SUBSCRIBE request
- Fixed no option of configuration on LCD menu
- Fixed phone did not display Caller ID on LCD

- Fixed onhook dialing on non-idle screen caused LCD display error
- Fixed call timer of long time conversation didn't display fully
- Fixed cannot see DTMF digits
- Fixed dialog box of “applying new configuration” didn't display fully
- Fixed EXT speed dial was not valid
- Fixed noise issue with busy call back
- Fixed Index value error in re-invite when use intercom to establish a second call
- Fixed GXP2120 Phone crashed when selecting “status” from keypad
- Fixed there are two different results when setting “Enable TR-069” to “Yes”
- Fixed lose registration issue under certain conditions
- Fixed no-audio with SRTP
- Fixed entering password more than 12+ digits caused phone crash
- Fixed date didn't display fully in call history
- Fixed during Attended Transfer with SRTP, transferee heard noise
- Fixed during Attended Transfer with SRTP, transfer target cannot hear transferee
- Fixed pressing MPK with “Dial DTMF” mode or “Enable MPK sending DTMF” did not send complete DTMF events via SIP INFO
- Fixed call logs could reappear after reboot
- Fixed phone responsed 488 when choosing SRTP mode as “Optional” on caller and “Enabled and forced” on callee
- Fixed phone cannot use Multi Purpose Key to send DTMF when “Disable in-call DTMF display” is set to “Yes”
- Fixed cannot redial after Blind Transfer
- Fixed Call Pickup failed on GXE
- Fixed Blind Transfer failed on GXE
- Fixed 3CX interop : GXP21xx displays missed call when call queue answered from elsewhere
- Fixed 3CX Interop: UPnP provisioning
- Fixed 3CX Interop : BLF pickup not working
- Fixed 3CX Interop : TLS issues
- Fixed 3CX Interop: BLF treats caller (direction=initiator, state=early) as ringing (flash red) instead of busy (solid red)
- Fixed 3CX Interop: lost registration problem (related to BLF)
- Fixed LCD display update had 2-5s delay when a call come in
- Fixed transferor cannot recover the conversation with transferee when use "SRTP" on Blind Transfer
- Fixed phone crashed with Call Park after web configuration update from GXE registration to OpenSER registration
- Fixed PPPoE account name with some special characters caused the phone not to boot up
- Fixed phone cannot save phonebook after reboot
- Fixed cannot add phonebook after reboot
- Fixed incorrect value used for sendrecv configuration
- Fixed phone cannot hear prompt with onhook dialing *30
- Fixed choppy audio during start of conversation
- Fixed noise issue when press 0 to send DTMF
- Fixed phone did not send DTMF info in early media
- Fixed cannot change in web GUI, set P64 to other values caused reboot loop
- Fixed GXP phone responded with "486 Busy Here" when number of dialog added up to 15+
- Fix progress bar displayed after reboot due to configuration change
- Fixed the first account can affect the other accounts when choosing Special Feature "HUWEI IMS"
- Fixed repeatedly use MPK to send DTMF via SIP INFO caused phone crash
- Fixed crash issue cause by music ring tone
- Fixed LCD lockup issue on answering multiple calls
- Fixed BLF Call Pickup prefix issue
- Fixed phone as transferee doesn't go back to idle after transfer target termite the call if transferor doesn't send BYE to transferee after received final NOTIFY

- Fixed blind transfer one-way audio issue and display/store wrong remote number issue
- Fixed doing factory reset from LCD MENU may not clear all configuration contents
- Fixed when disabled Call Logging in web UI, call logging did not turn off. Also, when disable call logs, previous call logs were erased
- Fixed a scenario where we stopped registration after the SIP server is down for a long period of time
- Fix dialog matching problem causing auto answer fail
- Fixed echo issue when RTP is sent to itself
- Fixed transferee didn't send INVITE to transfer target after receiving REFER using TLS
- Fixed phone accepted BLF NOTIFY which had Dialog-Info element with version older than the latest processed
- Fixed SIP Registration failed when no Authenticate ID is configured (should use SIP User ID implicitly instead)
- Fixed phone did not play DTMF tone from remote when in conference
- Fixed softkey state mismatch issue
- Fixed only check for firmware upgrade if pre/postfix changes
- Fixed GUI crashed during downloading phonebook when using an incorrect domain name
- Fixed we allow user to reboot the phone from the web UI even while in call. We will now wait until the call ends.
- Fixed directs IP calling crashed when press # and *
- Fixed # dialed out during Blind Transfer even though # is not used as dial key
- Fixed time display on LCD not updated via NTP
- Fixed Spanish date display issue (caused by bad translation string)
- Fixed when using "configured IP", can't make call after TCP connect gets [RST, ACK] response

ENHANCEMENTS

- Added call history writing enhancement
- Added feature to allow disabling weather, stocks, and/or currencies
- Added support to display forwarded call
- Added CBCOM mode
- Added support for TR-069
- Added support for AKAv1-MD5 authentication
- Added support to Polycom-style Call-Info header
- Added new feature to allow manually input time/date
- Added sorting when displaying call logs to prevent any miss-ordering
- Added three Mexican time zones with each DST
- Added support for Asterisk's out-of-box paging/intercom
- Added "Dial DTMF" mode to multi-purpose keys (GXP21xx only)
- Changed GXP2110 default LCD contrast from 10 to 7 (factory reset only)
- Changed Russian font and translation file per customer request
- Enabled Star key keypad locking on all models (only available to GXP1450 prior to this)

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Firmware Release Notes
Firmware Version 1.0.1.26
Name of Product GXP2100/GXP2110/GXP2120
Date: November 16, 2010

CHANGES SINCE FIRMWARE RELEASE 1.0.0.44

BUG FIXES

- Fixed GXP21xx Presence watcher does not work with Nortel server

- Fixed BroadSoft interop issue: Barge-in is not possible if SIP user ID contain '_' character
- Fixed BroadSoft interop issue: GXP2120 offers SRTP in SDP even when it is disabled after receiving INVITE without SDP
- Fixed phone doesn't display name info if meet tel format in PAI
- Fixed GXP21xx phones do not play voice in conversation when update web page information
- Fixed number via on-hook dialing will not be sent after SPEAKER button is pressed
- Fixed PPPoE does not work
- Fixed key presses gets queued during bootup
- Fixed if no mac item in XML provision file it will not write in
- Fixed phones do not send DTMF during early media
- Fixed phone cannot use speaker button to hang up under onhook dial mode
- Fixed GXP21xx crashes after configuration update and callqueue sign-in
- Fixed GXP2100 second incoming caller ID display covered by soft key
- Fixed phone sends DNS query for "0" when Config Server Path is empty
- Fixed INVITE for INTERCOM on shared line always use appearance-index=1
- Fixed gxp21xx cannot configure "Firmware Sever" with IP add port via LCD
- Fixed ForwardAll invalid when Authenticate ID is blank
- Fixed extension board issues

ENHANCEMENTS

- Added indication for registered, unregistered account status
- Added support for Russian and Croatian
- Added support for applying changes immediately for most variables without reboot including SIP account information
- Added stock and currency tabs to idle screen
- Added web configuration for stock and currency update (Basic Settings page)
- Added restriction to only accept UTF-8 as encoding in XML provision file
- Added support for Portuguese
- Addde support for grayscale enabled idle screen GUI
- Added support for RFC5922 (SSL certificate validation)
- BroadSoft interop: Added support for displaying Advice of Charge in LCD
- Increased Display of Numbers for Call History
- Increased RFC2833 digit duration to minimum of 100ms, fixed RFC2833 DTMF not always working
- Changed "user ID is phone number" option to "Tel Uri" so that user can disable it, use existing "user=phone" option or enable "tel uri" in INVITE request