GXE-5000 Release Notes

Build 1.0.1.63 (7/27/2010)

- Fixed GXE5000 will not show the DHCP assign clients.
- Fixed GXE5000 DHCP server assigns wrong IP addresses to clients.
- New UPnP module that supports more router implementations.
- Fixed GXE5000 becomes unstable when a voice prompt image is upload with a different file name.
- Fixed GXE5000 HTTP(s) wording in Chinese administrator web page.
- Fixed GXE5000 is unable to detect UK CallerID.
- Fixed GXE5000 DHCP server will assign a different IP address every time the client is rebooted.

Build 1.0.1.61 (6/10/2010)

- Fixed GXE5000 cannot set the DOS attack counter number to 100.
- Fixed GXE5000 drop call when a extension wants to pickup a FXO inbound call using BLF feature.
- Fixed GXE5000 crashed when using CTR5000 router because an error parsing the URL.
- Fixed GXE5000 system crashed when using some upnp capable routers because it doesn't close the TCP socket.
- Fixed GXE5000 UPNP client issue when the router doesn't use the default port maps.
- Fixed GXE5000 assigns an invalid extension when two terminals auto provision at the same time.

Build 1.0.1.60 (05/12/2010)

- Fixed GXE5000 RTP sent from Vinetic DSP with a wrong change in the RTP timestamp.
- Fixed GXE5000 process for FXS user to pick a park call using flash hook.
- Add return to main menu option by pressing star when user does a wrong selection in dial by name menu.
- Support “user=” parameter in SIP URI using sip trunk.
- Support automatic detection and protection against SIP DOS attacks .
- Fixed GXE5000 cannot show fax-mail messages in the personal web page.

Build 1.0.1.59 (04/06/2010)

- Modify the maximum Authentication ID length of extension from 24 to 40.

- Adjust default music-on-hold volume to prevent audio distortion.
- Modify the fax email notification subject and body text.
- GXE5000 call queue now supports a new agent login mode (direct login mode).
- GXE5000 implements a new filesystem to handle devices with bad NAND flash blocks.
- GXE5000 enhancement to prevent call drops under high network/GXE activity.
- Fixed GXE5000 would drop some calls when the IPterminal respond ACK after 5 seconds.
- Fixed GXE5000 crashed when conference room is full.
- Fixed GXE5000 will drop a call when there have too many calls(about 30) need to be transfered or forwarded conditions.
- Fixed GXE5000 cannot reset the IP address after a factory reset when the voice mail files and fax mail files use more than 70% of the flash memory space.
- Fixed GXE5000 in some situations cannot handle the DHCP offer lease time correctly.
- Fixed GXE5000 sometimes the voice may be choppy because it cannot resume sleep switch.
- Fixed GXE5000 lost CDR record because an error in the memory filesystem.
- Fixed GXE5000 crash because it receives a RFC2833 packet with RTP timestamp 0.
- Fixed GXE5000 sometimes cannot config the FXO channel 2 with right parameters. Therefore, this channel to not detect busy tone successfully.
- Fixed GXE5000 sometimes the CDR record call time length is 0.
- Fixed GXE5000 sometimes will not allow to upload a backup file if the length is more than 3MB.
- Fixed GXE5000 crashed because an invalid FXO configuration parameter.
- Fixed GXE5000 cannot show Faxmail messages in personal web page.
- Fixed GXE5000 crashed while doing a FXO auto detect.

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Build 1.0.1.54 (12/14/2009)

- GXE5000 support new driver.

- GXE5000 support DOD feature.
- GXE5000 IVR would replay the Dial by name result prompt 3 times if user do not input any select.
- Fixed GXE5000 SIPTRUNK device do not work when the URL would be set a more than 64 byte string.
- Fixed GXE5000 do not release the SIPTRUNK outbound call at once when the remote user is not online.
- GXE5000 would prompt manager “modify success” when manager just modify the call route profile name.
- GXE5000 do not submit the call route profile when manager input enter in the call route profile name field.
- Modify the “,” help document for call route profile.
- GXE5000 would prompt manager “the quoted call path would be deleted” when manger modify/delete/disable the SIPDID in SIPTRUNK.

Build 1.0.1.52 (11/20/2009)

- GXE5000 support Dial-by-Name feature.
- Fixed GXE5000 forward call failed when the terminal use simple SIP header.
- Fixed GXE5000 allow manager input special text in extension user name and department name in personal web page.
- Fixed GXE5000 modify the call-route profile file name sometimes.
- When manage input a wrong condition for call-route profile GXE5000 would input red fond warning prompt.
- Add a help link for auto-detect web page.
- Fixed GXE5000 allow manager input '.' for FXS/Group/Conference/group extension.
- Fixed GXE5000 drop call because manage input space in the DIGITMAP of call-route profile, now GXE5000 does not allow input space in DIGITMAP.
- When open the voice mail for extension GXE5000 set the entry option of voice mail to "Extension and password" instead of "password only".
- Fixed GXE5000 drop call when the terminal use re-invite session timer.
- Fixed GXE5000 would prompt attendee "conference password wrong" when attendee want to join a none password conference and input some digit in the joining flow.
- Fixed user cannot open GXE5000 web page by HTTPS sometimes.
- Fixed GXE5000 lost statistic information after reboot sometimes.
- Fixed GXE5000 send Email error by TLS mode sometimes.

- GXE5000 support new extension **Auto Provision Feature**.
- Fixed GXE5000 drop call when Polycom terminal want to forward call by refer signal.
- Fixed GXE5000 drop call when user transfer call to sip trunk and the sip trunk server respond multi 183 and 180 to GXE5000.
- Fixed GXE5000 do not detect CallerID in UK sin77-BT mode.
- Fixed GXE5000 do not add sip-trunk device name to from field in SIP DID mode.
- Fixed GXE5000 cannot send fax more than 60 pages by virtual printer mode.
- Fixed GXE5000 cannot build video connect by re-Invite between two video phone when they have build audio connect firstly.
- Fixed GXE5000 drop the FXO inbound call when user forward to the other user and both terminals only support G729.
- Fixed GXE5000 cannot check the DTMF signal sometimes when user leave voice message.
- Fixed GXE5000 generate a MAC00000000 terminal config file to the Non-Grandstaream terminal in the auto-provison process.
- GXE5000 support invite attendee into a conference in personal web page.
- GXE5000 expand the port forwarding number from 8 to 16.

Build 1.0.1.50 (9/25/2009)

- Fixed GXE5000 can not save config data because the cvs file too huge consume file system space.

- GXE5000 recognize the sip-info by content type when it conflict with X-grandstream header.
- GXE5000 support PAI(P-Asserted-Identity Header) config field in SIP Trunk.
- Fixed GXE5000 does not release memory when manager record voice menu by terminal.
- Fixed GXE5000 lost the statistics for wait time for CQ when the call forward to CQ voice mail.
- Fixed GXE5000 allow user input special character "&*%$" when he edit extension password field by FireFox.
- Fixed GXE5000 show 500 error when manager input date string that length more then 50 for PlayRule.
- Remove the Entry voice mail option for CQ and Group.

- GXE5000 support DiffServ QOS feature.
- Fixed GXE5000 forward call to CQ voicemail when conference call to this CQ.
- Fixed GXE5000 does not prompt manager when he add one existed IP into BlackList.
- Fixed GXE5000 crash because the CDR files space more than memory free space.
- After express setup GXE5000 build a default voicemenu and its NoEntry option point to extension 6000 voicemail.
- Remove the Cancel button in personal web page.
- GXE5000 will use default port 3478 when manager does not configure this port for STUN server.
- Add 5/10 seconds option for Group extension parallel ring time.
- Add 0 times option for voice menu's NoEntry repeat field.
- GXE5000 will check all Email address input in the same format as group extension Email input.
- Add parallel ring time config field for CQ.
- GXE5000 support three Entry voice mail modes—Extension&password/only password/directly.
- After manager enter the PSTN AUTODETECT web page, GXE5000 will force a reboot afterwards.
- Fixed GXE5000 does not release park room sometimes, and this issue cause all park room is busy.

- Fixed GXE5000 can add announcement for more than 30 entries.
- Fixed GXE5000 drop call when extension call back to voice message caller who come from TG trunk.
- Group Email address support "-".
- Fixed GXE5000 drop fax that HT502 call to GXE5000 FXS extension.
- Fixed manager cannot review CallQueue Greeting when manager upload this greeting in zip format.
- Fixed GXE5000 lost group voice mail password.
- Fixed GXE5000 drop call when conference invites an extension and this extension forward call to FXO trunk.

- Support new CallQueue feature.
- Support new CallPark feature.
- Fixed GXE5000 sometimes crashed when user want to call back to voice mail caller.
- Fixed GXE5000 cannot build Greeting when user records his/her voice by terminal without G.723 codec.
- Fixed GXE5000 drop call when it interop with "iNext" SIP trunk server, add config field for Reinvite delay time in SIP TRUNK web page.
- Fixed GXE5000 drop inbound call when it interop with Broadvoice sip trunk server.
- Fixed GXE5000 send huge SYSLOG to SYSLOG server and causes SYSLOG server crash when manager config a wrong information to SMTP server and send mode in SSL.

- Fixed GXE5000 does not delete temporary CDR files.
- Fixed GXE5000 will allow extension user enter administrator web page when the extension user input admin web page URL.
- Fixed GXE5000 lost BLF information when upgrade to version 1.0.1.47.
- GXE5000 support terminal hold Call by C = 0.0.0.0 in the SDP mode.
- Fixed GXE5000 cannot pickup Inbound Call from internal PSTN trunk, because GXE5000 does not release SUBSCRIBE dialog resource for sip terminal.
- GXE5000 support config the 2833 event Payload type in system web page.
- GXE5000 support config sip trunk support video or not in sip trunk web page.
- GXE5000 support config JitterBuffer type in FXO parameter set.
- Fixed GXE5000 does not allow manager modify the TimeZone sometimes.

- Fixed GXE5000 drop the call when the user inbound call from PSTN trunk and want to forward to peer's GXE5000 PSTN trunk.
- Fixed GXE5000 does not release socket when it send Email failed by a faulty SMTP server.
- Fixed GXE5000 does not accept the user dial DTMF code when GXE5000 auto-attendant check the user authority and the user dial DTMF by SIP INFO signal.
- Fixed GXE5000 accept the "." in batch add extension web page for extension number and batch number field.
- Update the help text web page for PSTN AUTO-DETECT feature.
- Update the help text web page for CallRoute feature.
- Fixed GXE5000 does not show the mouse-over text in authorization profile web page.
- Fixed GXE5000 drop the call when SIP extension call forward to external PSTN trunk.
- Add Remote Access feature for troubleshooting.
- Fixed manager cannot invite attendee from peer external PSTN trunk or internal PSTN trunk in conference web page.
- Fixed GXE5000 drop the call when the extension register expire time is over.
- GXE5000 would send bye with a reason string to the Hunting Group member, so the terminal does not show a miss call.
- Fixed GXE5000 will drop the inbound call that forward to Group extension by Auto-Attendant when all member of this group extension are not online.
- Fixed GXE5000 show long prefix digit in express setup web page sometimes.

- Fixed GXE5000 crashed cause by UPnP sent IP packet failed.
- GXE5000 would send bye with a reason string to the Hunting Group member, so the terminal does not show a miss call.
- Fixed GXE5000 conference does not accept the room number and password after user press # when the terminal send it in SIP INFO.
- Fixed GXE5000 would report wrong number when the terminal send DTMF by RFC2833 packet and the timestamp is changed in the end packet.
- Fixed GXE5000 cannot provision extension number for terminal when the terminal connects to GXE5000 WAN port in different network segment.
- Fixed GXE5000 sometimes cannot send CDR record to CDR server.

- Fixed the Wav tool generate the prompt image with noise in the end of prompt.
- Fixed GXE5000 cannot add CQ in 1.0.1.44.
- Fixed GXE5000 sometimes cannot backup system data when the last time backup failed.
- GXE5000 would reInvite SIP trunk server when the server respond GXE5000 with multiple audio codec in SDP.
- Fixed GXE5000 cannot submit or switch language in CQ web page when the language mode is French.
- Fixed GXE5000 allow manager input invalid letter for BLF name in BLF web page.

- Fixed Call fail when unconditional/busy/no answer call forward to peer user through SIPTrunk
- Fixed Call fail when FXO inbound terminated by IVR and then call peer user through SIPTrunk
- Fixed Call fail when FXO inbound terminated by IVR and then call local user that registered by another GXE5000 through SIPTrunk
- Fixed GXE web return 500 error when filled in over 50Byte Date/Time string in PlayRule Configuration
- Fixed SYSLOG show error info when Send VM email to user according to Email list
- Fixed link to fault web page when Press “Return” link in help page for CallRouting
- Support auto release call for user when manual VFAX successfully through FXO/FXS
- Fixed No Hold on music when one phone change hold to unhold and the another phone still hold
- Fixed Not fellow the Offer/Answer(RFC3264) when hold on music, Offer SDP is “a=SendOnly”, Answer SDP should be “a=RecvOnly”

- Support Bad Block management for Flash.

- GXE5000 support receive RFC2833 packet with Payload Type is 100.
- Fixed GXE5000 send request URL to SIP Trunk server with a default port 5060 when manager does not configure SIP signal port in the SIP trunk server in web page.
- Fixed GXE5000 cannot find SIP response for one session timer update request when the SIP DID trunk server IP is different from configure IP in web page.
- Fixed GXE5000 cannot build video media connection between two video terminal after one terminal transferred the call.
- GXE5000 support volume control in FXO trunk line.
- Fixed GXE5000 detect BusyTone error(sometimes) in PSTN auto detect flow.
- GXE5000 support input “+” in condition field for DigitMap profile.
- Fxied GXE5000 cut the call at one when user does not input any DTMF code for voicemail box at the third time.
- Fixed GXE5000 does not change the authority profile name for extension when the authority profile name has been changed.
- GXE5000 check the IP address format in network port forward web page.
- Fixed GXE5000 show the maximum concurrent number is 50 in SIPTRUNK and PSTN external trunk mouse over web page, now it change to 20.
- GXE5000 does not support Session timer when the terminal does not support “time” in invite request.
- GXE5000 support Remote Access feature.

- FXS support send vFAX When set Manual Selection of Fax to NO
- GXE support H264 interop with Eyebeam soft phone
- GXE support report “offline” state for BLF when extension lost its register info.
- Web Support French

Build 1.0.1.41 (5/15/2009)—Official Rleased

- Fixed GXE5000 does not allow user record Voicemenu when user use an extension to record the voicemenu but this extension is set to call-forward to other extension.
- Fixed GXE5000 drop the sip trunk(Callcentric/BroadSoft) call when the sip trunk server does not allow GXE5000 send update for session timer as a UAC.
- GXE5000 will not allow administrator to select group extension itself for group “no answer forward to”.
- GXE5000 will check the condition field logic in call route profile.
- GXE5000 will prompt administrator that system memory is not enough for system backup, when the NAND flash space is less than backup file size.
- Fixed GXE5000 drop the second call when user does not pick the first call and this terminal share one extension number with the other terminal.
- Fixed GXE5000 prompt error in system upgrade web page.
- Fixed GXE5000 IVR call to short key 0 when remote user just input # in voice menu flow.
- Fixed GXE5000 can not add call route profile in Spanish mode.
- Fixed GXE5000 spell “illgeal” for “illegal” in some web pages.
- Fixed GXE5000 does not accept call request when the phone connect to GXE5000 by TCP mode and system administrator modified its parameter via web.
- GXE5000 add GXV3140 template for extension auto provision.
- GXE5000 will prompt administrator the extension privilege need be valid after reboot.
- Fixed GXE5000 FXS extension can not resume call when FXS extension switch to FAX by flash hook and FAX flow is over.
- GXE5000 support sniffer capture Ethernet packet feature under Advance web page.
- GXE5000 support the maximum concurrent calls from 20 to 50 for peer system.
- GXE5000 use a 3 minutes wave file for music-on-hold in Chinese and Spanish mode.
- GXE5000 will prompt administrator the upload wav file length cannot more than 3 minutes.

- GXE5000 support new payload type 96 for RFC2833 RTP media packet.
- GXE5000 do express setup steps in a independent web page when it run first in reset factory mode, administrator need finish all steps of express setup to make GXE5000 go to normal configuration web page.
- GXE5000 support CBcom mode in sipserver for accepting terminal register.
- GXE5000 support Gratuitous ARP protocol.
- GXE5000 would prompt administrator that all messages in voicemail box or faxmail box would be lost when administrator try to disable the voicemail box or faxmail box for an extension.
- GXE5000 would auto build the condition and call path field for one new added digitmap in a call route profile.
- GXE5000 would show the caller number for voice message or fax message to “unknown” in personal web page.
- When administrator delete a call route profile, GXE5000 would delete the profile from all extension/conference/trunk…… call route list.
- Adjust the web page for siptrunk/external PSTN trunk/Peer list for showing all fields in fixed size.

- The size of Event list string change from 100 to 150.
- Fixed GXE5000 cannot allow user recording his greeting when he want to recording again.
- GXE5000 would prompt administrator to reboot GXE5000 for FXS parameter be valid when he modify the FXS parameter.
- GXE5000 would delete old FXO device at once when administrator configure an invalid string to the FXO line field, for example "---".
- Fixed GXE5000 allow configure multiple of the same trunk device in an authorization profile.
- GXE5000 would check the time format is valid or not for play rules.
- Adjust the web page after administrator input wrong password when he want reset GXE5000 to factory mode.
- Fixed GXE5000 can not release the FXS port call waiting tone when FXS does not accept the waiting call by hook flash.
- Fixed GXE5000 show web page when administrator input <tr> in extension department field.
- Fixed GXE5000 prompt attendee "this is wrong password" when the attendee does not input any DTMF for conference, Now the prompt change to "please input password".
- Fixed GXE5000 show the DDNS status "server error" sometimes even DDNS update the dynamic IP successfully.
- Fixed GXE5000 prompt user the extension is busy when user wants to paging/intercom conference, now the prompt change to "You are not authorized to dial this number".
- Fixed administrator cannot mute/delete one attendee in conference web page when the caller number of this attendee is "unknown".
- Change the default BT event number from 2 to 1.
- GXE5000 limit the voice message size to 5 minutes, user just can leaving one voice message or expand one voice message up to 5 minutes.
- Fixed GXE5000 catch any packet by troubleshooting sniffer tools when GXE5000 WAN port work in PPPoE mode.
- Fixed GXE5000 sometimes does not send MWI to terminal.
- Fixed GXE5000 drop the call to FXS extension when the FXS extension is ringing, now this call transfer to FXS voice mail.
- GXE5000 would forbid the submit button when administrator uploading the configuration file to GXE5000 in the system restore web page.
- Fixed GXE5000 send hold NAT packet failed when the remote device connect by TCP/TLS mode.
- GXE5000 build BLACK list to prevent some remote device to connect to GXE5000.
- GXE5000 show the BusyEvent number field in PSTN trunk web page.
- Fixed GXE5000 run slowly sometimes, it makes GXE5000 process all media and signal packet delay several seconds.
- GXE5000 would check port conflict or not between HTTP port/SIP TCP port/SIP TLS port/Telnet port.
- GXE5000 change the voice mail title to "You have received a voice mail message in mailbox XXXX".
- GXE5000 would prompt administrator "System memory space is not full for backup" when administrator want to system backup but the system memory is not full to save the backup file.
- When user transfer a voice mail from his mailbox by the other extension terminal the caller number of new voice number is the extension number of the user.
- GXE5000 support NAT Traversal with UPNP for SIP TLS port.
- Use new 3 minutes long classical Music-on-Hold music.

- Fixed GXE5000 cannot record greeting prompt for voice menu by telephone terminal.
- Fixed GXE5000 prompt test Email failed sometimes when the Networks delay time is too big.

- Fixed GXE5000 respond 200 to peer Option request when the peer device have deleted, now it change to 404.
- Fixed GXE5000 External PSTN trunk cannot work when the UDP list have ":" in the end.
- Fixed GXE5000 would reject X-lite subscribe request when this message does not include a accept header line.
- Fixed Prompt tool does not support NTFS file system.
- Fixed GXE5000 just permit user record maximum to 2 minutes prompt and greeting by telephone terminal, now change to 3 minutes.
- Fixed GXE5000 voice mail drops call when the personal greeting length is more than 2 minutes.
- Fixed GXE5000 conference can not invite one attendee from web page when the remoter device respond multi 180/183 to Invite request.

- Fixed GXE5000 cannot release call for the member of group when a caller call to a group and all members do not pickup and time over after 180 seconds.
- Fixed GXE5000 does not respond ACK for peer 200 message after the first ACK have lost by the networks.
- Fixed when conference invite group in web page and all members does not pickup the call will transfer to group voice mail.
- Fixed GXE5000 just send email to the first email address in the Extension Email list.
- Fixed administrator can input letter in the line field in PSTN trunk web page.
- Fixed GXE5000 drop the call sometimes when conference resume a call, because the time over is 10 second for re-invite respond.
- Fixed GXE5000 drop call when caller call to group and all members does not pickup, later caller transfer to the other extension and one member pickup.
- Fixed administrator can input letter in extension field and submit at once when he add one extension and input letter.
- Fixed GXE5000 drop one normal call when the other call from peer heart beat failed.

- GXE5000 support upload system music on-hold file in system setting web page.
- GXE5000 support Event-list function. Administrator can build maximum to 20 Event-list URL and select maximum to 130 members for every Event-list URL.
- GXE5000 support "+" in condition field of call route profile.
- GXE5000 does not allow clear LAN IP in networking web page and does not allow LAN IP and WAN IP in same net segment.
- GXE5000 support new DDNS service "cn99".
- Fixed GXE5000 get abuse response from DDNS service server because GXE5000 update same IP address to domain name after reboot.
- Fixed GXE5000 cannot stop ring terminal, The scenarios is that FXO inbound call to auto-attendant and transfer to one extension, after call over 60 seconds GXE5000 release the FXO call but does not release terminal call.
- Fixed when GXE5000 page a group, there are only 5 members ringing because GXE5000 set the concurrent call number for paging is 5, now this field set to 31.
- Fixed GXE5000 authorize one call failed because it gets authorization information from callee, not caller.
- Fixed GXE5000 allow preview voice menu, but administrator does not upload any prompt file for this new voice menu.
- Fixed GXE5000 record voice menu failed when user record voice menu by a telephone terminal.
- Fixed GXE5000 cannot implement that transfer call when FXS terminal is busy.

- Fixed GXE5000 does not prompt user “record finished” when user record system prompt file by telephone terminal and the maximum record time is over.
- Fixed GXE5000 play “the number you dialed is not reachable” to caller when the caller call to the paging extension and the paging extension is busy. Now the prompt change to “ the number you call is busy at the moment.”
- Fixed GXE5000 does not release the call of CallQueue/Group/FXS when administrator modify their configuration data.
- Fixed administrator cannot build FXS extension when the user does not finish the express setup flow.
- Fixed GXE5000 cannot build video media for GXV3140 when it registers behind a NAT router.
- Fixed GXE5000 open the voice mail and fax mail for Agent after administrator modify the configuration data.
- GXE5000 support No-entry time out field for dialing in system setting web page.
- Add Daylight saving time in time zone select box.
- Fixed cannot make voice mail invalid or valid at once when administrator close or open voice mail for one extension.
- Fixed GXE5000 display “400 Bad request” when administrator change language mode in add BLF source list web page.
- Fixed GXE5000 display “500 internal error” when administrator change language mode in call route help web page.
- GXE5000 allows upload MoH prompt file size maximum to 3 minutes for wav format and maximum 4M for zip format.
- Fixed GXE5000 register sip trunk to sip service server failed when there are over 2 sip trunk accounts from same sip service and set to SIPDID mode.
- Fixed GXE5000 cannot build media connect for FXS terminal when administrator call from conference to group and one FXS member of this group hook off for this call.
- Fixed GXE5000 does not respond 422 to SIP terminal when the terminal request with a session time value that is less than the minimum session value in GXE5000.
- Fixed GXE5000 does not accept caller ID from FXO inbound call in Haiti because there is an invalid value in the caller ID string.
- Change the No.407 system prompt from “Message saved” to “Saved message”; Change the No.411 system prompt from “Saved message” to “Message saved”;

- GXE5000 voice mail box support configurable time that GXE5000 will automatic delete voice message when the message time is over the period of time and it is not saved by user.
- Add two separate lines over the OneButton provision web page.
- Fixed GXE5000 drop the group parallel call after administrator configure the group extensions in personal web page.
- GXE5000 supports up to 50 SIP DID in one sip trunk and support multi same digit map in one CallRoute profile, so administrator can configure maximum to 50 SIP DID condition in one profile.
- Fixed GXE5000 will lost digit map after reboot when the call route option is Call queue in this digit map.
- Fixed GXE5000 play “goodbye” two times to caller when the caller want to attend conference but the conference room is full.
- Fixed GXE5000 drop the outbound call from PSTN trunk when the switch “wait for dial tone” set to yes.
- Fixed GXE5000 does not play busy prompt to caller when the caller call to the paging extension and the paging extension is busy.
- Fixed user cannot play the voice menu greeting after user upload the greeting file in ZIP format.
- GXE5000 will prompt error information to user when user uploaded a wrong format file to voice menu.
- GXE5000 will prompt error information to user when user uploaded a wrong format file to Call Queue Announcement.

- Fixed GXE5000 will lost Call routing profile information after administrator modify the information in "central management" web page.
- Fixed GXE5000 can edit playvoicemenu profile after administrator upgrade image from 24 to 35.
- Fixed GXE5000 drop call sometime when the router lost the ACK respond for peer trunk call invite request.
- Fixed GXE5000 show wrong number for the call statistics of peer/sip trunk/External PSTN trunk.
- Fixed GXE5000 drop the call when user retrieves a voice mail and the caller number of this message is unknown.
- Fixed GXE5000 cannot build media connect for FXS extension. The scenario is that the FXS user wants to call to the other extension by hook flash and this extension was not off-hook, and the FXS user switch to original extension by hook flash, later FXS user switch to the other extension by hook flash, then call drop.
- Fixed Excel cannot show the call number for GXE5000 call record file(*.csv) sometimes.
- Fixed GXE5000 drop FXS extension call when the user does not dial # at the end of number.
- Fixed GXE5000 does not build GeneralOutbound profile and assign InternalCall profile to extension when 1.0.1.24 upgrade to 1.0.1.35 when this GXE5000 does not build any trunk device.
- Fixed GXE5000 displays an error prompt window when administrator switch the configuration field by Tab key sometimes.
- Fixed GXE5000 show the IP address under the "status" in extension express provision web page.
- Fixed GXE5000 show English prompt window in FXO parameter set web page when the language is Chinese.
- Fixed GXE5000 delete the other digitmap when administrator wants to delete more than 3 digitmap in a call route profile.
- Fixed GXE5000 does not set the Add button to be invalid in sip trunk web page when the SIP DID numbers are over 50.
- Add Delete(button) to ensure window for deleting FXO parameter set and deleting route record.
- Fixed GXE5000 drop the call when user want to re-record voice message.

Build 1.0.1.35 (2/24/2009)

- Add "no entry forward to" function in voice menu.
- Fixed GXE5000 send re-Invite to Polycom failed because GXE5000 does not support session version update in SDP O parameter.
- GXE5000 show voice message length up to entire minutes in voice message Email.
- Fixed GXE5000 cannot release call at once when GXE5000 does not support the caller's PT or voice codec.
- Fixed GXE5000 will drop call when Agent(for Call Queue) wants to forward this call. Now Agent does not support call forward.
- Fixed GXE5000 show 500 error when administrator want to add long except date string for one voice menu.
- Adds new prompt for release call.
- Fixed GXE5000 sometimes drops the group call when one member is peer extension.

- Fixed sometimes GXE5000 can not connect media from FXO Trunk line to a SIP Trunk line because the remote SIP server cannot respond ACK for re-invite in 5 seconds.
- Fixed GXE5000 cannot build media connect when UPNP router cannot respond in 5 second.
- Fixed GXE5000 voice menu cannot accept multi 183 respond.
- New mouse-over for extension web-page.
- Fixed GXE5000 cannot get the CallerID from FXO line in Greek, now GXE5000 support Sin227 format.
- Fixed GXE5000 shows 500 error in Play Rules web page sometimes.

- GXE5000 does not release existing session when it detect remote does not respond signal keep live “option” request.
- GXE5000 support Session Timer when calling terminal supports Session Timer.
- Fixed GXE5000 prompt user “please dial again” two times when he inbound call from voice menu.
- Fixed GXE5000 give a wrong time length for voice message Email.
- Update Spanish web text.
- Clear stun server configuration in system web page after reset to factory.
- Fixed GXE5000 voice mail box cannot recognize fax from inbound FXO call in automatic fax mode.

- Fixed GXE5000 cannot forward voicemail or convert voicemail when this user voicemail space is full.
- Fixed GXE5000 play the prompt forever when user does not input any number for authorization, now GXE5000 just play 3 times.
- Fixed GXE5000 cannot stop the prompt when user input # for authorization.
- Fixed GXE5000 cannot allow modify the GeneralOutbound/GeneralInbound/InternalCall profile and allow modify PlayVoicemenu profile after 1.0.1.24 upgrade to 1.0.1.32.
- Fixed GXE5000 assert “remove file error” forever after administrator delete one voice menu.

- Fixed the submit button in peer configuration web page is invalid.
- Fixed GXE5000 show extension always busy when this IP Phone transfer one call to the other call by different account.
- Fixed GXE5000 prompt user will drop FXO Trunk call when modify the FXO Trunk configuration data.
- Support same icon for IE and Firefox in main menu field.

- Fixed GXE5000 configuration data (including trunk/call routing profile) error after firmware upgrade from 1.0.1.24 to 1.0.1.28.
- Fixed GXE5000 release call when conference invite one attendee after 30 seconds. Now the time length set to 180 seconds.
- Fixed GXE5000 drop the call when the terminal device cannot respond the Online detect request signal-“option” in 75 seconds, now the time length change to 300 seconds.
- Fixed GXE5000 crash when the UPNP route have some special text is too long.
- Fixed GXE5000 reboot fail, the reason is some file handle in FAXproc module do not free.
- **GXE5000 supports easy configuration for call routing feature.**
- Fixed GXE5000 cannot forward call to voice mail when user call to one extension that share in multi-terminal and one member press mute button to refuse this call.
- GXE5000 offer one default template for all Grandstream terminals in auto-extension provision flow.
- Fixed GXE5000 drop call when user attend conference and want to forward to FXS extension or FXO trunk.

- Fixed GXE5000 delete the voice mail media after firmware upgrade from 1.0.1.24 to 1.0.1.28, it causes user drop call when retrieving voice mail.
- GXE5000 display “Unknown” instead of blank in CDR record.
- Fixed GXE5000 cannot send caller number to Agent in the call queue flow.
- GXE5000 record the sniffer switch status in System SYSLOG file.
- GXE5000 record extension heart beat stop event in system SYSLOG when the terminal does not respond the Heard Beat “Option” request.

- Fixed GXE5000 drop call when the call forward to IVR and the terminal hold/unhold the call.
- Fixed GXE5000 cannot send out the debug info file because the debug info file name is too long.
- Fixed GXE5000 cannot send out the debug info file when the system Email mode is MTA.
- GXE5000 automatic generate a new extension number when administrator want to add an extension.
- Fixed GXE5000 displays a warning window in Chinese when the language mode is English after administrator switch the language some times.
- Fixed GXE5000 just connect one way media sometimes when Peer system send re-invite for modify media information.
- GXE5000 support SSL and the other UPD port for SMTP.

- Fixed GXE5000 reboot failed or crash after firmware upgrade from 1.0.1.24 to 1.0.1.28 because the FXO parameter is Null or fault.
- In the default FXO port parameter GXE5000 open both current disconnect and tone disconnect.
- Fixed GXE5000 cannot make some feature code real-time effect sometimes.
- Fixed GXE5000 just forward call the next call path after 180 seconds when the last SIP Trunk call path does not respond 180.
- Fixed GXE5000 cannot retrieve the voice mail when the voice mail is leaved from SIP Trunk.
- GXE5000 will prompt user “GXE5000 will drop the current calls of the Extension/SIP trunk/peer” when he want to modify/delete the Extension/SIP trunk/peer system.
- If the administrator Email list have multi address GXE5000 will send the password to the input Email address when user want to retrieve the login password.
- Fixed GXE5000 allow user input letter to FXS extension number.
- GXE5000 will prompt user check “OK” or not when he wants to delete FXO Trunk device.
- Remove the example extension that is behind the Call queue extension.
- Fixed GXE5000 show the Extension Name and department name wrapped in the extension list web page sometimes.
- Fixed GXE5000 cannot delete Trunk when the call routing profile number is more than 19.
- Change the “NONE” to “None” in the group authorization profile select item.
- Fixed GXE5000 cannot add 10 agents for call queue.

- Support configuring different parameters for every FXO port.
- Fixed GXE5000 drop the inbound call from FXO port when the inbound call is without CallerID signal.
- Fixed GXE5000 does not play the call-waiting tone to FXS when the FXS call to one group extension and this FXS is a member of this group.
- Fixed GXE5000 cannot call to the other voice menu in one voice menu flow.
- Fixed GXE5000 drop the call when the FXS user forwards this call to the other extension and this extension have set the None-condition forward function.
- SIP Trunk support 491/500 responds.
- Fixed GXE5000 cannot make one CallQueue real-time effect when there are more than 2 CallQueue in the system.
- Fixed GXE5000 drop call when user retrieves voice mail with G723.
- Fixed GXE5000 cannot make the group user real-time effect.
- Fixed GXE5000 cannot release SIP Trunk call when user modify its configuration data.
- Fixed GXE5000 the current calls number error in the SIP Trunk and peer web page.
- Fixed GXE5000 cannot accept the 180 second for group parallel ring internal time.

- Fixed GXE5000 drop the call when user wants to forward call to SIP trunk.
- Fixed GXE5000 does not forward call to voice mail when the call to FXS extension and the FXS extension have set to "Do no Disturb".
- Fixed GXE5000 crash sometime because FXO stop CPT detect error.
- Fixed GXE5000 initialize FXO port error when this port configuration parameter set is Null, GXE5000 will initialize this port by default parameter.
- Fixed GXE5000 cannot make the FXO port parameter real-time effect.
- Fixed GXE5000 still light the FXO led when user pull out the PSTN line sometimes.
- Fixed GXE5000 cannot call to next CallPath sometimes because CCM does not update the callee control information.
- Fixed GXE5000 cannot make the paging/intercom real-time effect.

- Add Venezuela time tone option in system configuration web page.
- Fixed GXE5000 crash when user delete the attendee from conference web page.
- GXE5000 show the status for the "Music on hold" in the status web page.
- Fixed GXE5000 do not update the IP address to DDNS server when the UPNP route have change the IP address because the ADSL line up and down.
- Fixed GXE5000 cannot receive the fax mail for the extension that is added before reboot.
- GXE5000 will prompt user the reason "the Trunk is use in call routing profile or authorization profile." When user cannot delete the trunk.
- Add the divide line for the "mapped IP:port" rim in the status web page.
- GXE5000 prompt the user "the maximum value must less than 20 " when user configure a value that more than 20 in the concurrent call number for SIP trunk/Peer system/External PSTN trunk.
- Add the divide line for the call record list when the record is without callee number.
- Add last page and first page link for call record web page.
- Set the daughter menu to Boldfaced when user select the daughter menu by just click the father menu.
- Check the name is valid or not for Email/Callrouting condition/callroute manupilation/voice menu name.
- GXE5000 prompt user "name is exist" when use want to add new trunk/callroute profile/authorization profile but name is exist.
- Change the color of " local" same as "online" in the extension list web page.
- Change the "call route profile"/"Authority profile" to "Call Routing profile"/"Authorization profile".
- The valid character for every name is include "space" and ".".
- Change the login web page to green color.
- Change the menu "Route Config" to "IP Route Config".
- Fixed GXE5000 does not list the extension form small to big in extension list web page.
- Fixed GXE5000 show the peer device when the option is "Trunk" in call routing profile web page.
- All option string change to capitalization for option in call routing profile web page.
- Add notes "when option is password need use extension password to authorization" in the authorization web page.
- Add 2 items--150 and 180 for group parallel ring internal time option.
- Make the title name same as the menu name for "Template Upload"/"Call routing Profile"/"Authorization Profile".
- Fixed FXS extension cannot attend conference after it have exit conference when it is the last attendee.
- Improve the Call record web page—delete redundant block and some lines wrap in Chinese mode.
- Fixed GXE5000 drop SIP Trunk inbound call sometimes when the SIP Trunk connect with server by TCP mode.
- Fixed GXE5000 cannot close the hot key in voice menu by the trigger switch.

- Fixed GXE5000 drop the inbound call from SIP Trunk/FXO Trunk when the call forwarded to intercom/paging service.
- Fixed GXE5000 conference show the attendee number error after attendee forward call to another extension.
- Fixed GXE5000 SIP Trunk register error when this Trunk register to one server by two counter concurrently.
- Fixed GXE5000 SIP Trunk does not register to server after the UPNP router IP address changed because ADSL line down and up.
- Fixed GXE5000 generate error IP for SIP contact when GXE5000 is under a UPNP router and the NAT style is symmetrical and SIP signal port does not map.

- Fixed GXE5000 drop the call that forward to External PSTN Trunk(TG).
- Fixed GXE5000 drop the call when it is inbound from External PSTN Trunk(TG) to IVR and forward to one peer system External PSTN trunk(TG).
- GXE5000 would prompt user “itself extension number for authorization please press #” when GXE5000 want to get the authorization user number.
- Administrator can configure the 4 field(switch/digit manipulation/option/value) in call path of call routing profile independently.
- Fixed GXE5000 show the voice mail name error.
- Fixed the attach wav file name in the voice Email display error.
- Fixed GXE5000 drop the call when user want to pick up a FXS call by DSS.
- Fixed GXE5000 show the attendee number error when the callee forwards call to the other terminal, for example FXO line/VM etc.
- Fixed GXE5000 HTTP server cannot work when the HTTP port is conflict with map port of the WAN network port.
- Fixed GXE5000 display the conference password in web source code.
- Fixed GXE5000 cannot point a value for “IVR” option in call path of the call routing profile when the version upgrade from 1.0.1.21 to 1.0.1.25.
- Fixed GXE5000 cannot forward to voice mail when the call is forward by condition is “No answer”.
- Fixed GXE5000 display the FXO call progress tone title in two rows in the internal PSTN Trunk web page when language is Spanish.
- Fixed GXE5000 cannot save the authorization profile for extension when the authorization profile file name length is more than 25.
- Fixed GXE5000 cannot forward call from voice mail to fax when user press 2 select to send fax to extension.
- Update the help content in the call routing profile web page in English.
- Fixed GXE5000 stop convert the voice mail to G723 format when concurrent call is more than 20.
- Fixed GXE5000 just display 4 digit number for the caller number when the call is inbound from PSTN trunk.
- Prompt-file tool will prompt user the valid wav format when it detect an invalid wav file.
- Fixed GXE5000 IVR cannot forward call to voice mail/group/fax when the short key option is point to this value.
- Fixed GXE5000 IVR cannot replay the voice menu greeting when the IVR no entry time have set to zero.
- Fixed GXE5000 IVR cannot forward to extension by point the “fax to” field when it detect the inbound call happen a fax event.
- GXE5000 support SIPDID in condition of the call path in call routing profile.
- Fixed GXE5000 would change the value of call routing profile sometimes.
- Fixed GXE5000 cannot forward call to FXO trunk/SIP trunk/External PSTN Trunk in query voice mail flow.
- Fixed GXE5000 CGI error when the condition string in the call routing profile is more than 255.

- GXE5000 FXO Trunk CallerID format support DDN parameter.

- **<u>GXE5000 support call route feature.</u>**

- Fixed GXE5000 would crash when the fax pages that virtual fax received from remote is over 20 page. In this version GXE5000 support received fax pages number is maximum to 50.
- Fixed GXE5000 does not free the UPNP port map in the router when GXE5000 reboot.
- Fixed GXE5000 does not release the call when this call want to part but parting lot is full.
- Fixed GXE5000 cannot send Email for the rest Email address in group or CallQueue extension when there is an invalid Email address in the Email list.
- Fixed GXE5000 cannot build the media for Inbound call from one Trunk and outbound call to SIP Trunk when the SIP Trunk server respond 183.In this version GXE5000 will re-Invite the SIP Trunk server and build the media connect.
- Fixed GXE5000 register to SIP Trunk server failed when the server respond 200 for the register but it is without expire field.
- Fixed GXE5000 crash because it processes some UPNP respond error.