GRANDSTREAM NETWORKS
Firmware Release Notes
Firmware Version 1.0.8.4
Product Name: GXW40XX / HT50X
Date: December 11, 2012

SUMMARY OF UPDATES

The main purpose of this release is implementing additional features and addressing stability issues observed in previous releases.

CHANGES SINCE FIRMWARE RELEASE 1.0.7.6

BUG FIXES

- Fixed device crashes after firmware upgrade/downgrade under certain conditions. Upgraded boot loader version to 1.0.0.16
- Fixed device does not check for binary configuration file after GAPS redirection under certain conditions
- Fixed if P8 is set to 01, IVR does not announce the correct IP address
- Fixed call hold fails with MetaSwitch
- Fixed device does not increment nc on the first re-register
- Fixed device gets in a reboot loop if XML configuration file contains same P numbers but with different values
- Fixed device does not play Ring Back Tone when it off hook auto dials another device
- Fixed when iLBC Payload Type is 101 call has no audio
- Fixed when iLBC payload type is 120 call has noise
- Fixed TR-069: Firmware upgrade using HTTPS with authentication freezes device
- Fixed IVR plays incorrect info for IP settings under Static IP
- Fixed device crashes under long term testing
- Fixed device sends malformed DHCP Packet when DHCP host name is longer than 14 characters
- Fixed Time Zone is delayed by one hour
- Fixed Circular Hunting Group ringing error
- Fixed Blind Transfer fails under certain condition
- Fixed Direct IP Call fails when use # as re-dial key
- Fixed Broadsoft Interop: no reINVITE for Fax re-negotiation to PCMU causing Fax Pass-through fail under BS special feature
- Fixed device does not respond 401 challenge on FAX pass-through reINVITE for codec re-negotiation
- Fixed BroadSoft Interop: device does not play ring back tone under BS special feature
- Fixed when caller is not in transfer or hold or waiting status, Current Disconnect does not work when caller's call setup failed
- Fixed crash caused when call is cancelled before SIP stack is ready
- Fixed Current Disconnect does not work when device is in single call and receives BYE

ENHANCEMENTS

- Added support to display Connected Line ID
- Added option “Reregister before Expiration”
- Added option "DNS Mode: Use Configured IP”.

- Added option “Disable SIP NOTIFY Authentication”
- Added support for SIP NOTIFY “resync” event
- Updated gs_cpe release to 1.0.1.28

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Firmware Version 1.0.7.6
Product Name: GXW40XX / HT50X
Date: August 21, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.6.13

BUG FIXES

- Fixed device locks up after provisioning under certain conditions
- Fixed it takes too long to respond to 407 after sending a BYE
- Fixed Voice Band Transmission delay issue
- Fixed outbound calls do not work using ports members in Hunting Group under certain condition
- Fixed SIP stack stops responding on an outbound call
- Fixed HT503 Life-line feature does not work without account being registered first
- Fixed a bug in syslog printing
- Fixed 301 Redirect handling error
- Fixed disabling FXS1 will cause the other FXS Ports unavailable
- Fixed device didn't use the Keep-alive Interval setting to re-send Binding Request
- Fixed not updating USA Session timer when processing dialog replace
- Fixed if Alert-Info is used in a format different from supported device will drop the INVITE
- Fixed device does not ring if INVITE contains: Call-Info: answer-after=0
- Fixed when enable forward in Hunting Group Call Transfer fails
- Fixed call fails when device gets challenged by 407 followed by 401
- Fixed Authenticate Conf File option does not work properly
- Fixed device will crash if it keeps receiving 486 Busy Here
- Fixed error when transferee off hook before semi-attended-transfer completes
- Fixed device crashes under certain dial plan setting
- Fixed dialing Busy Forward is invalid
- Fixed Off-hook Auto-dial doesn't accept star key
- Fixed the problem that illegal value can be saved in webUI "off hook auto dial"
- Removed "enable ring transfer" from web UI
- Removed "DHCP Domain" from Web UI
- Fixed cannot deliver CID when ring cadence is changed
- Fixed internal communications can be hang up by the third party
- Fixed BroadSoft interop: fax pass-through fails with re-INVITE
- Fixed BroadSoft interop: HT7xx fails to handle INVITE with Diversion Inhibitor
- Fixed BroadSoft interop: REGISTER Fail back does not send to the primary server
- Fixed BroadSoft interop:INVITE Fail back does not send to the primary server
- Fixed device reboots during a call if the web UI Reboot button is pressed
- Fixed with TLS incoming calls will stop working and go straight to voice-mail
- Fixed IVR response issue under certain conditions
- Fixed default Dial Plan not consistent with other products
- Fixed device as callee has noise on G726-32
- Fixed “Use # as dial key” set to “No”, for Direct IP call the # key can also be used as the send key
- Fixed memory leak issue when STUN and "Validate Incoming Messages" are enabled
- Fixed Syslog for NTP does not include MAC address
- Fixed semi-attended or attended transfer sometimes appeared one way or two-way no audio.
- Fixed "SSL.." related web UI description not clear
- Fixed device can dial out while busy tone is played
- Fixed SRTP issues

- Fixed Attended Transfer appears one way audio and hanging up has no IVR
- Fixed enable but not forced SRTP setting causing Call Transfer failure
- Fixed device can set up error 3-way conference without pressing FLASH under SRTP mode
- Fixed device cannot resume a held call under SRTP mode
- Fixed device crashes if both caller and callee set the SRTP MODE to enable but not force.
- Fixed CANCEL was sent without UAS provision response
- Fixed Semi-attended transferfer failed on Asterisk
- Fixed when No Key Entry Timeout value is large then 10,the real timeout value is inconsistent
- Fixed GXW40XX chose random RTP port on second incoming call of hunting group
- Fixed offhook issue for hotline calls
- Fixed callee did not use the audio coder negotiated in the 200OK
- Fixed update fail for web UI options XML Config File Password and HTTP/HTTPS Password
- Fixed HT503 FXO port in the INVITE didn't carry name when set user=phone
- Fixed TR-069: CPE doesn't send autonomous inform with 4 VALUE CHANGE on WAN address change
- Fixed TR-069: CPE does not does not verify the validity of SSL certificate
- Fixed crash issue caused by setting the LAN IP Address as active notification too early

ENHANCEMENTS

- Added option to disable the Hook Flash function
- Added "Connection Request Port" to "Advanced Settings" web page
- Added "Telkom SA" Special Feature
- Added a patch to support RFC2617(TR-069 requirement)
- Added ability to configure delay for off hook auto dial
- Added support for generic XML configuration file
- Added display of gs_cpe version in status page
- Added CPE SSL Certificate and CPE SSL Private Key in "Advanced Settings" page

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Firmware Version 1.0.6.13
Product Name: GXW40XX / HT-50X
Date: May 15, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.6.8

BUG FIXES

- Fixed internal call doesn't have audio

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NOTE: Once upgraded to 1.0.5.10, downgrading to any previous releases is not supported.

Firmware Version 1.0.6.8
Product Name: GXW40XX / HT-50X
Date: April 10, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.5.10

BUG FIXES

- Fixed Direct IP Call does not have Call Waiting tone
- Fixed setting Caller ID Scheme to ETSI-FSK prior to ringing with RP rings automatically

- Fixed device does not ring in some DTMF schemes
- Fixed HT503 converts unreserved characters "." and "-" to hexadecimal format
- Fixed device doesn't use maddr from the Route Record Header
- Fixed device gets in a loop of sending bye messages
- Fixed a memory leak causing GXW4024 crash after a few days of usage
- Fixed DTMF detect at a high speed crash when using SIP-INFO
- Fixed GXW4024 Subscribe For MWI crash The Unit
- Fixed Dial Plan set to a single digit dialing is abnormal
- Fixed device does not accept Re-INVITE with "t38+other codec" with pass-thru
- Fixed SIP T2 Interval setting does not take effect except for 4 seconds
- Fixed Dial Plan Prefix can be set to space. Limit allowed characters in P values 66, 619, 666, 766 (Dial Plan Prefix)
- Fixed when set UAS Specify Refresher to UAC DUT device doesn't use UAC to send update
- Fixed Prack CSeq always incremented upon receiving a 18X message retransmission
- Fixed after Attended Transfer the third party hold back the call appears one way audio
- Fixed setting iLBC Payload Type set to 101 have no audio
- Fixed caller request timer causes one way audio
- Fixed Disable Visual MWI doesn't work
- Fixed HT503 FXO port Chinese and English mode configuration not consistent
- Fixed firmware check mode get from IVR does not match the webUI display
- Modified DHCP client to ensure pipe to process is closed after use

ENHANCEMENTS

- Added support to apply configuration changes without reboot
- Added support to display Connected Line ID
- Updated TR-069 CPE
- Added NTP update interval option. P value 5005
- Added an option to change the Voice Frames per TX
- Added a configuration option to override User-Agent header
- Added an option to Enable/Disable each FXS Port for GXW40xx
- Added support of dual frequency for dial tone. New P values 4041 for Prompt Tone and 4042 for Prompt Tone Access Code
- Added support to send SIP log in syslog
- Changed the P-Asserted-Identity (PAI) header to the P-Preferred-Identity (PPI) header in SIP INVITE
- Added support for single image upgrade

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Firmware Version 1.0.5.10
Product Name: GXW40XX / HT-50X
Date: November 3, 2011

CHANGES SINCE FIRMWARE RELEASE 1.0.5.5

BUG FIXES

- Fixed device crashes when Subscribe for MWI is enabled
- Fixed problem with LAN IP-conflict-detection feature under PPPoE mode. Modified IP configuration request to not ask for default IP
- Fixed device doesn't switch to G711 when modem signal detected under Pass-through mode
- Fixed device doesn't switch to T.38 or PCMU when Fax tone receive. Added a web UI configuration option "Enable Silence Detection for Fax Disconnect" to enable/disable the 7-sec silence detection. (P value 4406--4409)
- Fixed device doesn't reply with PRACK message to the 2nd RSeq. Fixed the case when PRACK is challenged by 401 Unauthorized or 407.

- Fixed device does not failover to second server when DNS mode is set to SRV
- Fixed device in TCP mode cannot establish the call
- Fixed TCP/TLS connection shutdown due to keep-alive packet from SIP server
- Fixed SIP Registration Failure Retry Wait Time error
- Fixed GXW-40xx does not handle the RFC-3891 correctly
- Fixed HT503 does not free the FXO port
- Fixed IVR cannot announce the IP address correctly when it changes
- Fixed device cannot make Direct IP call when the SIP server configured does not exist
- Fixed problem that direct IP call does not go through when "Use # as a dial key" is set to No
- Fixed device do not reset the nonce count after getting challenged with a new Nonce
- Fixed GXW4024 Offhook Auto-Dial choice still in profile 1 (Chinese)
- Fixed when doing ring-transfer, after C rejects B(or rings time out), B works abnormal
- Fixed DNS process issue
- Fixed device cannot register when SIP transport set to TCP(or TLS) and both Primary and Failover have SIP server configured

ENHANCEMENTS

- Added the option to change the Voice Frames per TX
- Enabled DHCP option 125
- New TR-069 binary and dataModel
- Removed SIP password from configuration file
- Reset the Hunt Group Ringing Port to the Active one for every new incoming call. (Added new P values 4395, 4396, 4397, 4398 for Hunt Group Type)
- Use Option 66 for Provisioning Using HTTP the same way as for GXP21XX

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Firmware Version 1.0.5.5
Product Name: GXW40XX / HT-50X
Date: July 19, 2011

CHANGES SINCE FIRMWARE RELEASE 1.0.4.2

BUG FIXES

- Fixed the problem that application software keeps writing to the flash under certain conditions
- Fixed user cannot set DNS mode to NAPTR/SRV on HT502 FXS PORT2
- Fixed ALL2-G726-16,32,40 codec has bad voice quality
- Fixed HT503 NTT CID scheme not working
- Fixed in call waiting, obvious echo in one caller after the other caller hangs up
- Fixed device crashes because incorrect order of data in 183 SDP causes the media to be NULL
- Fixed the replace/prepend syntax < = > in HT503 FXO Dial Plan does not take effect
- Fixed device does not keep track of multiple IPs when looking up Proxy FQDN
- Fixed GXW400X Failover to FXO not working
- Fixed device cannot get an incoming call while C did not answer when doing Ring Transfer
- Fixed device does not respond properly to 408 during an Attended Transfer with Nortel
- Fixed GXW400X Life Line mode issue
- Fixed SRTP issues
- Fixed ROH tone issue when using the IVR for the first time after device boots up
- Fixed GXW4008 Life Line mode setting misses a default value
- Fixed problem that Direct IP call using *47 does not work
- Fixed NEON: the message led will stop flash on inter-port call
- Fixed GXW40XX without setting Primary Sip Server cannot map to device
- Fixed inter-port calls cannot hear the busy tone after one of them goes on hook
- Fixed device does not honor the prefix on Unconditional Call Forward
- Fixed HT503 FXO cannot handle transfer in two stage mode (PSTN to VOIP)
- Fixed a problem that if both 180 and 200 OK after 183 does not contain SDP, there is no audio

path in the call
- Fixed incorrect DNS handling in static IP mode

ENHANCEMENTS

- Extended the length of Firmware Server Path and the Config Server Path fields
- Minor edit in GUI for Ring tone cadence syntax. Removed [...] in the ring tone syntax in web UI
- Added support for defining Target sip peer under different conditions for GXW40XX
- Added Special Feature ZTE IMS. If special feature is ZTE IMS, and select user=phone, REGISTER request does not contain user=phone. Other signaling will contains user=phone. Also for ZTE IMS Special Feature, extract CID from the FROM header first.
- Added support for 100rel (PRACK)
- Added support for tel URI

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Firmware Release Notes
Firmware Version 1.0.4.2
Product Name: GXW40XX / HT-50X
Date: March 24, 2011

NOTE: Firmware 1.0.3.x has major changes and upgrade takes longer time than usual. Please be patient and let the device finish the firmware upgrade before doing anything. Once upgraded to 1.0.3.x, downgrading to any previous releases is not supported.

CHANGES SINCE FIRMWARE RELEASE 1.0.3.10

BUG FIXES

- Fixed device does not send INVITE to “Failover SIP Server” when “Prefer Primary SIP Server” is set to “Yes”
- Fixed device keeps sending SUBSCRIBE to “Primary SIP Server” even when the server is unavailable
- Fixed a scenario where device stops sending Registration requests after the SIP server is down for a long period of time
- Fixed device not follow RFC3261 on PRACK
- Fixed DTMF negotiation error with early media
- Fixed HT503 FXO cannot handle transfer in two stage mode
- Fixed device displayed leading character D in Caller ID
- Fixed Call Waiting tone did not play continuously
- Fixed device lost registration when using STUN under public IP
- Fixed some GXW40xx ports did not send RTP
- Fixed device did not issue provisioning request at boot up while the uplink was not connected
- Fixed GXW40xx could not make Telematrix NEON phones to ring
- Fixed device web UI display issue in Chinese
- Fixed TCP/TLS closed connection caused the SIP stack shutting down

ENHANCEMENTS

- Added support to disable Receiver Offhook Tone
- Added support to switch between the 2 parties for Attended Transfer
- Added support for provisioning P value validation
- Changed default Diff-Serv value to 12
- Added support for Port-to-Port mapping with GXW410x
- Enhanced Failover Server feature to use Primary Server whenever device re-registers
- Added protection to prevent user from rebooting device without logging in

- Added special feature “PhonePower”
- Added support for MWI with NEON lamp on GXW4004 HW Rev. 1.3+ and GXW4008 HW Rev. 1.5+
- Added NTP client transaction information in syslog

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Firmware Release Notes
Firmware Version 1.0.3.10
Product Name: GXW40XX / HT-50X
Date: October 28, 2010

CHANGES SINCE FIRMWARE RELEASE 1.0.1.63

BUG FIXES

- Fixed device did not add “Dial Plan Prefix” for Call Forward
- Fixed the display name of account did not show in REGISTER request
- Fixed UK CID did not work
- Fixed device failed to play Bellcore DR5 based on Alert Info
- Fixed device did not fail back to primary server under certain conditions
- Fixed the issue with switching from T.38 to Audio Call (G729) after fax is done
- Fixed GXW4024 dropped packets after DHCP REQUEST
- Fixed device did not show CWCID when CW tone is disabled
- Fixed FXS did not generate DTMF tones
- Fixed device crashed after taking wrong P value in configuration file
- Fixed device did not keep same SSRC in same RTP
- Fixed HT503 did not close TCP/TLS connection when the DSL modem changed its WAN IP
- Fixed on HT503: if the FXO used in-audio or RFC2833 DTMF mode, the PIN for VoIP-to-PSTN Calls was invalid
- Fixed device could not register with some servers due to Digest user name
- Fixed HT503 failed to detect Hong Kong CID
- Fixed device did not send BYE when using OBP
- Fixed DNS feature did not work if the Preferred DNS server was an out of range IP
- Fixed DTMF RTPEVENT packet encapsulation error under PPPoE
- Fixed HT503 replied “503 INFO” with out-going calls
- Fixed Dial Plan { ^xxxxxxxxxxxxxxx+# | 0x+# | [#9]xxx | [123]xx | *x+ } did not work
- Fixed device as a callee did not use the audio codec negotiated in the 200OK
- Fixed device did not increment sess-version when modifying session data in SDP
- Fixed on HT503: incorrect FXO port treated a FXS

ENHANCEMENTS

- Added support for MWI with NEON lamp on GXW4024 HW Rev. 0.4
- Added support for Special Feature “Ring Transfer”
- Added support for TR-069
- Added support for XML provisioning
- Added support to respond with only the first matched vocoder in SDP in 200 OK
- Added support for port-to-port mapping with GXW410X
- Added support for tel-uri
- Added support to prepend CLI before the SIP URI in the Contact header

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Firmware Release Notes
Firmware Version 1.0.1.63
Product Name: GXW40XX / HT-50X
Date: May 20, 2010

CHANGES SINCE FIRMWARE RELEASE 1.0.1.57

BUG FIXES

- Fixed GXW4024 crash issue related to early media
- Fixed Dial Plan issue with ‘#’ right after ‘+’
- Fixed device sends DNS query to wrong server before sending CANCEL
- Fixed device can dial out while Busy Tone is played
- Fixed GXW40xx IVR "Device Not Registered" is played when Hunting Group is Active and Main Account is Registered
- Fixed device drops calls when 'Loop Current Disconnect' is set to Yes
- Fixed device Config/FirmwareTFTP path parser returns error if one of the octets of the IP address is 255

ENHANCEMENTS

- Added support for showing display name before SIP URI in Contact header
- Added star code *03 to disable LEC on a per call base
- Added web UI option "Disable Line Echo Canceller (LEC)" for each port/profile

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Firmware Release Notes
Firmware Version 1.0.1.57
Product Name: GXW40XX / HT-50X
Date: February 9, 2010

CHANGES SINCE FIRMWARE RELEASE 1.0.1.56

BUG FIXES

- Fixed device WAN side responds to DNS queries
- Fixed under PPPoE mode, RFC2833 DTMF event does not work
- Fixed realm parameter does not contain quoted-string when the challenge has empty realm

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Firmware Release Notes
Firmware Version 1.0.1.56
Product Name: GXW40XX / HT-50X
Date: January 28, 2010

CHANGES SINCE FIRMWARE RELEASE 1.0.1.54

BUG FIXES

- Fixed device does not regenerate DTMF with RFC 2833
- Fixed device does not support GSSP with FQDN
- Fixed SSL Key and Certificate get truncated
- Fixed GXW4024 application lost under certain conditions
- Fixed GXW4024 memory leak issue
- Fixed FAX fails after receiving reINVITE that contains T38MaxBitRate:33600
- Fixed application crash issue on GXW4024

ENHANCEMENTS

- Added syslog for Call Waiting Caller ID
- Enhanced management on LAN DHCP Base IP -- When the device detects WAN IP is conflicting with LAN IP, the LAN Base IP address will be changed based on the network mask -- the effective subnet will be increased by 1. For example, 192.168.2.1 will be changed to 192.168.3.1 if net mask is 255.255.255.0. Then the device will reboot.

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Firmware Release Notes
Firmware Version 1.0.1.54
Product Name: GXW 40XX / HT-50X
Date: December 10, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.42

BUG FIXES

- Fixed when device calls out, it doesn't respond ACK after receiving 200OK from remote side
- Fixed HT503 doesn't reply ACK for PSTN incoming call under certain conditions
- Fixed GXW4024 crashes under certain conditions
- Fixed issues with Anatel Certification Tests
- Fixed GXW4024 profile 3 page error. The option value on web page does not change if using IVR to change Disable Call Waiting, SRTP Mode, Send Anonymous, or preferred voice codec
- Fixed HT503 crashes for VOIP-to-PSTN call (1-stage) if FXO port User ID is not configured
- Fixed for Blind Transfer, device doesn't response in time after receiving "Notify"
- Fixed device rejects all packets coming from backup server when "Use DNS SRV" and "Allow incoming SIP messages from SIP proxy only" are set to "Yes"
- Fixed device does not reset nonce count
- Fixed when Syslog Level is set as "NONE", it still sends some syslog information
- Fixed device does not send CANCEL when REFER gets 403
- Fixed HT503 crashes with too many pass-thru
- Fixed one way audio in *00 case when FXS and FXO use different codecs
- Fixed device freezes after "iperf" test
- Fixed device responses 400 Bad Request after receiving Refer with Compact header

ENHANCEMENTS

- Added support for traffic shaping feature

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Firmware Release Notes
Firmware Version 1.0.1.42
Product Name: GXW 40XX / HT-50X
Date: September 29, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.41

BUG FIXES

- Fixed HT503: In Unconditional Forward to VoIP case, FXO does not cancel the call after the PSTN side hangs up
- Fixed FXS ports get blocked on GXW4004 w/Hunting Group
- Fixed device still has Call Waiting Tone when "Disable Call-Waiting Tone" is set to YES
- Fixed GXW4024 Attended Transfer failed on GXE
- Fixed Dial plan sent as name in SIP From Header
- Fixed account can not register if the 200 OK from the SIP server does not include Expires header or "expires" parameter in the Contact header

- Fixed device reboots too quickly not giving time to send 200 OK
- Fixed device responds 481 (subscription does not exist) for incoming NOTIFY
- Fixed 202 Accepted and Notify out of order when responding to the REFER request
- Fixed for device two ports inner call, after the call timeout, it can still establish communication
- Fixed for SRTP mode, 200 OK does not contain SRTP media attribute
- Fixed an issue with Transfer on Conference hang up when FXS port hangs up to transfer the conference to the remaining two PSTN parties
- Fixed device can not send T.38 fax under certain situation

ENHANCEMENTS

- Added support for "Disable Reminder Ring"
- Added support for two VLAN priority configuration
- Made ABCD alphabetic characters (in Dial Plan) case sensitive
- Enhanced device to load dial plan first before configuration
- Added support for annexb attribute in SDP content. For G729, added "annexb=no" in SDP attribute if VAD is disabled
- Added support to select codec for a single call via star code. (This feature is only applicable to FXS port)
- Added double-quote to the caller name in INVITE
- Added support for "Allow incoming SIP messages from SIP proxy only"
- Added support for Semi-Attended Transfer (Transfer while ringing)

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Firmware Release Notes
Firmware Version 1.0.1.41
Product Name: GXW 40XX / HT-50X
Date: August 6, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.35

BUG FIXES

- Fixed device does not send re-Registration after receiving inconsistent time stamp from NTP
- Fixed device fails to process signaling locally when using DMZ

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Firmware Release Notes
Firmware Version 1.0.1.35
Product Name: GXW 40XX / HT-50X
Date: April 24, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.21

BUG FIXES

- Fixed problem when "*Validate Incoming SIP Message*" is set to Yes, device responds 415 to incoming NOTIFY
- Fixed IP Address obtained via PPPoE is invalid after network disconnection
- Fixed under PPPoE mode, the remote side can't hear GXW4024
- Fixed GXW4024 *"Send Hook Flash Event"* does not work
- Removed Contact header from 404
- Removed HT503 unsupported function *"Disable Call-Waiting"* and *"Disable Call-Waiting Tone"* from FXO port web page
- Fixed HT502 fails to load RTP payload table
- Fixed after *87 Blind Transfer, device does not send out CANCEL under certain conditions

- Fixed no audio issue after *87 Blind Transfer under certain conditions
- Fixed GXW40xx Chinese web page Radius Timeout/Retry default value is not correct
- Fixed GXW4024 Profile 3 Chinese web page Local SIP/RTP port default value is not correct
- Fixed HT503 can't call-out from FXO when FXS and FXO port uses different DTMF method
- Fixed GXW4024 1-way audio problem
- Fixed HT502 unhold has wrong SDP when handling INVITE with (a) = inactive
- Fixed firmware download fails with postfix added
- Fixed NAT traversal on port 2 cannot be set to *"No, but keep alive"*
- Fixed miss-handling of SDP "inactive" attribute
- Fixed fax issue with Avaya
- Fixed T38 fax from GXW4024 (registered on GXE5000) to PSTN crash issue
- Fixed HT503 makes DTMF short

ENHANCEMENTS

- Added HT503 syslog to display caller ID number/name detected by FXO port
- Added an additional registration notification via IVR
- Added support for DTMF capability preferences
- Enhanced "*Check SIP User ID for incoming INVITE*" to check SIP User ID in request URI, instead of To-header
- Enhanced "*Firmware File Postfix*" implementation to be consistent with other products

CONFIGURATION UPDATES

N/A

NOTES

N/A

Firmware Release Notes
Firmware Version 1.0.1.21
Product Name: GXW 40XX / HT-502
Date: November
 20, 2008

CHANGES SINCE FIRMWARE RELEASE 1.0.1.15

BUG FIXES

- Fixed DMZ issue
- Fixed one way audio issue with Call Park/Pickup
- Fixed a timing issue with Attended Transfer
- Fixed T38FaxVersion:0/1 issue
- Fixed device does not play incoming DTMF tones
- Fixed device reboots during a call after taking a new configuration file
- Fixed fax issue CRM:00280219

ENHANCEMENTS

- Added support for Registration Retry Wait Time
- Added support for 3rd SIP Profiles for GXW4024
- Added support for Net Connectivity Detection via STUN
- Added support for Offhook Auto Dial
- Added support for Failover SIP proxy
- Added support for Spanish IVR

- Added support for SIP User ID checking
- Added support for T.38 fall back to pass through

CHANGES SINCE FIRMWARE RELEASE 1.0.1.8

BUG FIXES

- Fixed device would cut off live call during provision and firmware upgrade.
- Fixed forward-on-busy is not enabled when the phone is off-hook.
- Fixed call forward on busy not working under 3-way conferencing.
- Fixed no reminder ringing in certain call waiting cases.
- Removed LAN device on GXW4024 that might cause problems if GXW4024 is connected under 192.168.2.X network.
- Fixed crash issues under certain conditions.
- Fixed infinite loop when invalid configuration parameters are in the configuration file.
- Fixed broken Bellcore style distinctive rings.
- Fixed RTP port might be an odd number if random port selection was chosen.
- Fixed redial when dial plan prefix was used.
- Set caller ID to anonymous if the number field is not numeric.
- Fixed alignment problem with Chinese web pages in the firmware upgrade section.

ENHANCEMENTS

- Set HTTP/HTTPS download timeout to 20 seconds for each try.
- Added the support for 24-way concurrent calls on GXW4024
- Added the option for periodic upgrade based on minutes.
- Added the capability to automatic fall back to pass-through if T.38 negotiation fails.
- Added the capability to send client certificate when using HTTPS provision.
- Added support for E.164 compliant caller ID display by removing the leading '+' character.
- Added fax tone detection option for both caller and callee.
- Added device MAC address in HTTP request User-Agent field.
- Added the support for TFTP port for upgrade server URL.
- Added support for multiple DNS servers (more than 2) in DHCP response.
- Reduced DHCP discover packet size to 350 bytes.
- Added the support for replacement block in the middle of a dial plan segment, e.g. {<0=00549>[2-9]xx<15=>x+}.

CHANGES SINCE FIRMWARE RELEASE 1.0.0.86

BUG FIXES

- Fixed one way audio issue when blind transfer fails.
- Fixed bus error under some timing conditions using OpenSER.
- Fixed one way audio issue under GXE502x.
- Fixed timing problem with immediate re-INVITE after session established.
- Fixed distinctive ring problem using Alert-Info.
- Fixed incorrect handling of inactive attribute in SDP.
- Fixed broken PPTP pass-through.

ENHANCEMENTS

- Re-enabled UPnP support.
- Used status code 433 for anonymous rejection per RFC 5079.
- Added the support for Chinese HTML.
- Added options to enable/disable SIP Instance ID. The P values are 288 and 489.
- Added PPPoE CHAP support.
- Added hour and day of the week option for automatic upgrade. P285 and P286 are added.

CONFIGURATION UPDATES

N/A

NOTES

N/A

CHANGES SINCE FIRMWARE RELEASE 1.0.0.77

BUG FIXES

- Fixed incorrect session timer refresher as callee.
- Fixed T.38 incomplete DCN signal under certain conditions.
- Fixed memory leak in driver.
- Fixed RTP not stopped in call transfer timeout case.
- Fixed null pointer exception if reINVITE arrives before early media is stopped.
- Fixed a typo related to jitter buffer length setting for port/profile 2.
- Changed German ringing frequency back to 25Hz.
- Fixed ringing failure if previous CID is still in-progress.
- Fixed FXS CID TX timing that might cause CID not working reliably.
- Fixed incorrect initial SIP destination port number if DNS SRV is configured but not available.
- Fixed T.38 using FEC mode.
- Fixed incorrect Refer-To header when using transfer on conference hang-up.
- Fixed 0-value Session-Expires header.
- Fixed invalid DNS lookup when TFTP server and path are combined in firmware server URL field.

ENHANCEMENTS

- Added the support for Huawei SDP attribute “X-modem”.
- Changed the logic to play reorder tone instead of busy tone when INVITE 404 response is received.
- Changed DTMF duration in SIP INFO from 320ms to 240ms.
- Added Huawei special feature to support SoftX3000 call waiting.
- Added the option to disable call waiting caller ID. P-value 714 and P-value 823 are added for each account/profile.
- Avoided sending VMWI requests to analog phone if MWI status does not change.
- Disabled UPnP support due to excessive memory usage.
- Checked for Privacy header when displaying caller ID and applying anonymous call rejection logic.
- Added the support of ‘+’ character in dial plan replacement segment.
- Added the support to use Alert-Info to drive distinctive ring similar to other Grandstream products.
- Disabled RFC 2833 even if the other party requested but it was disabled on the device.
- Updated USA SLIC setting.
- Increased dial plan length to 1024.
- Disallowed the update of dialog remote tag.
- Added the logging of product model/firmware version during boot up.
- Changed G.726-32 logic in SDP to be compliant with RFC3551 while maintain the backward compatibility. G.726-32 vocoder will be offered using static payload type 2 (for backward compatibility) and dynamic payload type. Changed MIME type for all G.726 family to AAL2-G726-XX. P127 and P821 are added for G.726-32 dynamic payload type.
- Added P-Asserted-Identify and Remote-Party-ID support to drive caller ID display.
- Added provision using HTTPS protocol.
- Added the dynamic negotiation of Comfort Noise payload.

CONFIGURATION UPDATES

N/A

NOTES

N/A

CHANGES SINCE FIRMWARE RELEASE 1.0.0.67

BUG FIXES

- Fixed T.38 negotiation issue when SRTP is enforced.
- Fixed ringing frequency for Germany, Japan, China, Finland, Australia, Spain and Italy.
- Fixed Call-Info header syntax when making paging calls.
- Fixed multiple reINVITE requests may generate incorrect 180 response.
- Fixed crash issue when UPDATE is not in the Allow header with Session Timer.
- Fixed driver error caused by disabling fax/modem detection before call pickup
- Fixed inconsistent re-REGISTER back off interval
- Fixed crash problem when fax tone is detected in early media.
- Fixed SRTP reference count not decremented when early media also used SRTP
- Fixed one-way audio problem during attended transfer.
- Fixed ring cadence parsing that is not dividable by 400.
- Fixed distinct ring configuration not loaded properly.
- Fixed SSL key and certificate truncated using HTTP.
- Fixed crash problem caused by Broadsoft session audit UPDATE message.
- Fixed interop issues with Mitel using SIP over TLS.

ENHANCEMENTS

- Updated copyright notice to 2008.
- Added WAN port link status detection for FXS gateway.
- Changed the logic to choose to T.38 instead of audio when SDP attribute a=X-fax (Huawei) or a=fax (ZTE) exists.
- Added new time zone for Venezuela.
- Added the support of Radius server's request to terminate calls when run out of pre-paid credit.
- Added RADIUS attributes for billing server development.
- Modified T.38 library to send out no-signal indicator when T.38 session starts.
- Modified SDP parser to perform case insensitive matching for T.38 attributes.
- Added option to remove OBP in the route header. P4562 is added for profile/account 1 and P4563 is added for profile/account 2.
- Enhanced DNS query mode to support NAPTR (RFC 3263). "Use DNS SRV" options (P103 and P702) are renamed to "DNS Mode". Possible values are 0 – A Record 1 – SRV 2 – NAPTR/SRV. If the service provider DNS service supports NAPTR, the device will perform NAPTR lookup first and locate the proper SIP transport and service name. It will then perform a SRV lookup for the host name and port number.
- Added RAIDUS based authentication and accounting. If the primary server is configured and the authentication port is not 0, the device will get authenticated before making phone calls. If the primary server is configured and accounting port is not 0, the call record will be sent to the RADIUS server. If primary server is not reachable, secondary server will be tried. The new P values are:
  Primary RADIUS server: P4550. Default value is blank.
  Primary RADIUS Auth Port: P4551. Default value is 1812.
  Primary RADIUS Acct Port: P4552. Default value is 1813.
  Primary RADIUS server secret: P4553. Default value is blank.
  Secondary RADIUS server: P4554. Default value is blank.
  Secondary RADIUS Auth Port: P4555. Default value is 1812.
  Secondary RADIUS Acct Port: P4556. Default value is 1813.
  Secondary RADIUS server secret: P4557. Default value is blank.
  RADIUS Timeout: P4458. Default value is 2 seconds.
  RADIUS Retry: P4559. Default value is 3 retries.
- Added an option to do transfer between other parties or terminate calls when the device as a conference host hangs up. The P value is 4560 for the first profile/account and 4561 for the second profile/account. The default value is No.
- Added LCP echo every 30 seconds with PPPoE.
- Added sip-instance and reg-id parameters in REGISTER request per IETF SIP Outbound draft.

CHANGES SINCE FIRMWARE RELEASE 1.0.0.44

BUG FIXES

- Fixed some issues with provisioning of the unit using Windows based Configuration tool
- Changed layer 3 QoS value to be consistent with other Grandstream products
- Fixed device may fail to boot up if upgrade process fails under certain conditions
- Fixed an error parsing expires header
- Updated regional settings for European countries
- Fixed broken early dial feature
- Fixed the FXS configuration parameter to increase the ringing voltage. Added China as a dedicated FXS setting
- Fixed direct IP call issue when the callee did not have user ID configured.
- Fixed dialog route set update problem when provision response and final response had different route set
- Fixed hunting group (GXW only) port scheduling when the first ports were off hook
- Fixed FXS current disconnect period did not change regardless of the configuration

ENHANCEMENTS

- Enhanced Dial Plan feature to support * and #
- Added LAN DHCP enable/disable (P-value 5001), LAN DHCP starting address (P-value 5002) and LAN DHCP maximum user (P-value 5003).
- Added the configuration file download feature in Advanced Settings page.
- Added the support for session timer. This feature works the same as other Grandstream products.
- Added an option to validate incoming SIP messages and generate 4XX response accordingly. There are two new P-values added: 4340 and 4341
- Added configurable SIP T1 and T2. Their usage and P-value are identical to Grandstream products
- Relocated port 2 RX gain from P250 to P283 to avoid P value collision
- Added '*' in User-Agent header if provider lock was on.
- Added support to failover to FXO gateway
- Added DHCP option 66 support to override provision server
- Added the following TFTP private options in TFTP request:
    - grandstream_MODEL
    - grandstream_ID
    - grandstream_REV_BOOT
    - grandstream_REV_CORE
    - grandstream_REV_BASE
    - grandstream_REV_PROG
- Added paging mode using Call-Info header. New star code *74 is used to initiate a paging call.
- Added on-hook timing configuration. P-value 833 and 834 are added.

CONFIGURATION UPDATES

N/A

NOTES

N/A