GRANDSTREAM NETWORKS

Firmware Release Notes
Firmware Version: 1.3.1.6
Name of Product: GXW 410X
Date: December 8, 2009

NOTE: Once upgraded to 1.0.1.8 and above, you will not be able to downgrade to any previous releases.

SUMMARY OF UPDATES

The main purpose of this release is improving voice quality and addressing stability issues observed in previous releases.

CHANGES SINCE FIRMWARE RELEASE 1.2.1.5

BUG FIXES

- Fixed packet contains extra data after being IP fragmented
- Fixe device doesn't send out CID of Australia to VOIP side
- Fixed 1 stage incoming call does not show CID during VOIP call setup period
- Fixed device sends out DNS A record query first when configured with DNS SRV
- Fixed SIP display info of FROM header is empty after call transfer
- Fixed device crashes when syslog is filled with domain name and syslog level is set to DEBUG under Static IP mode
- Fixed CID detection issue with Verizon PSTN line
- Fixed Caller ID missing in South Africa
- Fixed device fails to recognize CID from Ericsson MD110 under Turkish Telekom

ENHANCEMENTS

- Set the default "Min Delay Before Dial PSTN Number" to 500ms
- Added more PSTN line status display on Status page
- Add pause to Dial Plan when dialing
- Moved DMTF related setting under Dial Plan page
- Added support to bridge 2 RTP streams between FXO
- Added support for CID auto detection under FXO Line Test page
- Improved CID detection stability

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Firmware Release Notes
Firmware Version 1.2.1.5
Name of Product GXW 410X
Date: July 2, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.25

BUG FIXES

- Fixed calling out causes losing second dial tone of the incoming call

- Fixed one-way audio issue after call transfer failure
- Fixed device does not follow DHCP renewal time when the DHCP server changes it
- Fixed two RTP stream cause poor voice quality
- Fixed GXW4108 port 2-7 issue after call transfer fails due to wrong number
- Fixed no audio issue if VoIP call receiver sets up unconditional forward to another VoIP number
- Fixed GXW410x web page display error
- Fixed out of channel issue(SIP REGISTER encounter 302 Moved response)
- Fixed device does not send out register when it does not receive reply from DNS server
- Fixed device loses gateway IP address if DHCP ACK does not contain the Gateway option
- Fixed GXW4108 transfer fails under certain scenarios
- Fixed using To-header to get PSTN dial number
- Fixed ptime values is not right for port 2 and 3
- Fixed GXW410x not passing FLASH to FXO due to SIP INFO contains signal=hf
- Fixed DNS related problems
- Fixed session timer refresher is not correct
- Fixed device crashes when set "Unconditional Call Forward to VOIP" to a profile that uses domain name as SIP server
- Fixed device fails to send session refresher at half expiration time
- Fixed no INVITE sent when packet is over 1500 bytes
- Fixed setting minimum RTP port to 1024 doesn’t take effect.

ENHANCEMENTS

- Added support for SIP over TCP

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Firmware Release Notes
Firmware Version 1.0.1.25
Name of Product   GXW 410X
Date: May 20, 2009

CHANGES SINCE FIRMWARE RELEASE 1.0.1.10

BUG FIXES

- Fixed GXW4108 would hang up if transferee picks up within 7s after PSTN call is transferred
- Fixed CID not transported correctly, Hong Kong CID displays unknown
- Fixed GXW410x can not hang up FXO port
- Fixed GXW4108 FXO port off-hook failures
- Fixed GXW410x crash issue when FXO hang up events are redundant
- Fixed GXW4108 sends redundant DTMF number
- Fixed GXW4108 replies 481 No Such Call to UPDATE INFO
- Fixed gxw4104 uses rr-18 in Web UI comments
- Fixed GXW4104 could not reboot from WEB UI
- Fixed the session timer refresher is not correct
- Fixed ACK to GXE 200OK of T38 re-INV is not correct
- Fixed GXW410x does not hang up if VOIP side Ring No Answer (SIP 408) for 2 stage dialing PSTN to VOIP calls
- Fixed it takes long time to get IP through DHCP if server response is slow
- Fixed GXW410x does not follow DHCP renew time when server changes
- Fixed for 2 stage dialing, GXW40x does not play busy tone if a wrong number is dialed
- Fixed GXW410x AC Impedance default value is not "1" after factory reset
- Fixed T38 relay issue on GXW410x
- Fixed One way audio issue in GXE call scheme

- Fixed SDP negotiation issue
- Fixed crash issue with autodial performance test
- Fixed Tone Disconnect issue with multiple ports
- Fixed Current Disconnect issue with multiple ports
- Fixed PSTN to VOIP call issue when GXW410x disables SRTP and the other party enables SRTP but not enforced
- Fixed memory leak that created syslog of "Out of Mem"
- Fixed GXW410x does not regenerate * and translate escape # in INVITE (1 stage dialing)
- Fixed Web UI does not highlight selected TAB color
- Fixed SIP INFO sent to wrong places after Verso server uses our own WAN IP
- Fixed wrong setting of NAT in account 2 affects all accounts
- Fixed mismatch between number of frame/TX between GXW410x and Siemens Gateway
- Fixed load64 upgrade cannot auto-reboot and requires a power cycle (fix in new load 1.1.3.4)

ENHANCEMENTS

- Added additional boundary condition for cadence resetting when detecting CPT
- Increased string size under Unconditional Call FWD to VoIP
- Enhancement -- if Unconditional Call Forward is configured with a user ID, UCF have higher priority over stage dialing for PSTN incoming calls
- Make FXO port hook flash configurable in Web UI
- Enhanced Web UI comments: example 4 under DIAL PLAN and Unconditional Call Forward under CHANNEL
- Detailed the critical data logging in application

Firmware Release Notes
Firmware Version 1.0.1.10
Name of Product GXW 410X
Date: October 10, 2008

CHANGES SINCE FIRMWARE RELEASE 1.0.1.8

BUG FIXES

- Fixed flooding DNS queries if STUN server is invalid
- Fixed device keeping trying firmware download if firmware upgrade server address is invalid
- Fixed Peer System not working unless User ID is configured
- Fixed VoIP caller gets 403 if no User ID is set in channel table even though caller IP is in SIP profile
- Fixed typo on ring-no-answer in previous build
- Fixed READY LED light doesn't light up after system boots up successfully
- Fixed GXW4108 Local SIP Listen Port option setting ch1:5061;ch2:5062;ch3:5063;ch4:5064;ch5:5065;ch6:5066;ch7:5067;ch8:5068 will cause system unable to boot up (issue on build 1.0.1.2, 1.0.1.8 but GXW4104 is OK)

ENHANCEMENTS

- New boot and loader to better handle system recovering if needed
- Improved regional PSTN incoming CID detection reliability
- Enhanced web UI comments on current disconnect threshold
- Added detail usage instructions on CPT tones(units, etc) in web UI
- Enhanced LED indication with sequence or pattern for provisioning
- New tone detector (DTMF, Call Progress Tone)

- Added support for regeneration of hook flash event upon receiving event from VoIP side via SIN INFO or RFC2833
- Changed web UI status page: Interface to Part Number (Under HW Revision)
- Added configurable DTMF payload type, default 101
- Enhanced Dial Plan to allow replacement block in the middle of segment, such as {<0=00549>[2-9]xx<15=>x+}
- Changed H.264 default packetization mode to 0;H.264 level to 2.0; and H.264 bogus packets to use SEI
- Reduced jitter buffer delay by 50ms

CHANGES SINCE FIRMWARE RELEASE 1.0.1.2

- Add support of non-numeric character in user id of channel table and unconditional call forwarding
- Enhance dial-plan to allow + as a leading prefix in dial plan
- Fix system CBCOM mode doesn't encrypted SIP if profile 2 is active
- Support Venezuela time zone
- Fix system cannot save Proxy-Requires for SIP profile 1
- Fix SRTP issue when receiving 183 response before 407
- Fixed a CID bug under some regional environment
- Improved to allow FXO pick up upon CID acquisition instead of waiting till the 2nd ring
- Fixed Unconditional Call Forward uses default SIP profile in From header if profile 2 is configured for the port
- Allow Unconditional Call Forward server Port to be less than 1024
- Support up to 32 characters allowed in user-id, authen-id, and password fields
- Support send SIP REGISTER even if authentication password is not required
- Added configurable Use Outband DTMF Params under FXO/Lines pages
- Support Web UI accept non-numeric character for UserID and Authenticate ID
- Fixed a SIP session timer bug to search ";refresher="
- Improved security handling when the caller sent 180/183
- Support DTMF type Caller-ID (DTMF CID support)
- enable DTMF detection on outbound calls from VOIP to PSTN (PSTN side input DTMF digit after call is established)
- Fixed sometime lose SIP registration when local sip port change

CHANGES SINCE FIRMWARE RELEASE 1.0.0.55

BUG FIXES

- Configurable RNA (request not answered) timeout is not working correctly
- Active dialog was not matched correctly if Call-ID length is longer than 64 bytes
- Random call drops in during active conversation
- Poor audio quality on certain calls

ENHANCEMENTS

- Added support to process RTP Event of # toward PSTN network
- Added support to process SIP INFO digit of * and # to regenerate those digits towards PSTN
- Add support of application/dtmf-relay for SIP Content-Type in sip_messages.
- Added preventive check on memory boundary
- Added preventive timer for “Wait for Dial tone” feature to prevent port hangs up in case of some mis-configuration conditions.

- Enhanced Echo cancellation mechanism

Release Notes of 1.0.0.55

Changes since release 1.0.0.41

- Added support for SRTP thorough SIP message key exchange
- Added support for configurable Local SIP Listen Port
- Added support for Fix Port or Round-robin port scheduling (VoIP to PSTN)
- Added support for configurable SIP URI for Offhook Auto Dial (PSTN to VoIP, 1 stage dialing)
- Fixed GXW uses 503 as system unavailable if system is busy
- Fixed no BYE or CANCEL if PSTN hangup during ringing for 1 stage dialing
- Added configuration support of Enable Disconnect Tone Detection
- Added configuration support of min delay before dialing after off-hook
- Added configuration support of PSTN CID relay to use SIP From or SIP P-Asserted-Identity.
- Removed duplicate Silence Suppression under ADVANCD web UI
- Added enhancement on Status Page showing more info if line is busy
- Added support for configurable item of PSTN Current Disconnect threshold under FXO/Lines
- Added support for space character in dial-plan grammar input.
- Fixed PSTN incoming call gets rings forever if sip profile is set to SIP Register No
- Fixed software reboot doesn't work after a PSTN to VoIP call if 1 stage dialing w/o off-hook auto dial is configured
- Fixed port hang after a previous off-hook auto dialed INV failed because server is down and 1 stage dialing is configured without off-hook auto dial.
- Added support of dial plan for VoIP to PSTN 1 stage dialing (see notes and web UI for grammar)
- Corrected typo of "FXO Lines" if Status Page is active in previous build
- Removed extra "Inter-digit Timeout" under ADVANCED (duplicate with this under FXO Lines tab)
- Added support of T38 ECM mode and configuration
- Support No STUN message but send keep alive packet "No, but send keep-alive"
- Corrected telnet interface shows GXV-3000 as command shell prompt, instead of GXW-4100
- Support FXO voip to PSTN call supervision via polarity reversal service on the PSTN line(web UI under FXO/Lines, see following notes for usage
- Added support for Attended Transfer using "Refer-To Uses Target Contact"
- Added support for configurable Silence Suppression and Voice Frames per TX UI (Advanced Settings)
- Improved T38 success transmission rate for concurrent faxing
- Added support for challenging remote-reboot NOTIFY (replies "401 Unauthorized" with WWW-Authenticate header).
- Fixed we do not use the same Authorization credential in ACK as in INVITE
- Fixed we do not follow Retry-After as indicated in 500/503 for REGISTER
- Fixed we respond to incoming non-INVITE requests with incorrect account when talking on a different account
- Added under Broadsoft mode, register delay after 403 changed to 20 minutes (otherwise 60 minutes)
- Support of DHCP option 61 (client identifier) and removed DHCP option 57 (maximum DHCP message size)
- Added option to Disable Call-Waiting Tone