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 1.4.1.4

BUG FIXES

- Fixed device responds 484 error to re-INVITE causing incoming call fail
- Fixed web spelling and alignment issues

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FIRMWARE VERSION 1.4.1.4
NAME OF PRODUCT: GXW 410X
DATE: JUNE 18, 2013

CHANGES SINCE FIRMWARE RELEASE 1.3.4.13

BUG FIXES

- Fixed Caller ID auto detection failure
- Fixed PSTN incoming call by-pass 3CX restricted call after server returns SIP 403 if caller holds long enough
- Fixed all incoming calls are rejected if other FXO port was configured with wrong SIP credentials
- Fixed FXO crashes after receiving two or more calls
- Fixed device crashes after receiving 403 Forbidden
- Fixed 3CX SIP user account FXO scheduling cannot take PSTN incoming calls
- Fixed device cannot take PSTN incoming calls if off-hook auto-dial is configured in FQDN format
- Fixed device cannot register if SIP proxy is in FQDN format
- Fixed SIP header misses transport type
- Fixed pressing hook flash hangs up PSTN incoming calls

ENHANCEMENTS

- New Web UI
- Added support for Audio and CID buffer download via web UI
- Added support for configuration file download under ADVANCED web UI
- Added periodical DNS query to probe DNS mapping changes after system start
- Added radio button to enable use SIP user account FXO port scheduling, a system variable for 3CX (P3187)
- Added radio button to enable “Use OBP in Route” per profile in web UI (P3184-3186)

Firmware Version: 1.3.4.13
Name of Product: GXW 410X
Date: February 22, 2012

CHANGES SINCE FIRMWARE RELEASE 1.3.4.10

BUG FIXES

- Fixed ringing signal not detected with Alcatel 4400 PBX
- Fixed Canada Caller ID not working properly
- Fixed FXO cannot detect the CID of Norway and Jordan_FSK
- Fixed if the Caller ID Scheme is DTMF, the second auto detection will get FSK scheme
- Fixed cannot detect Caller ID coming from Alcatel OmniPCX Office
- Fixed audio cut off when Echo Canceller is set to Yes
- Fixed Brazil Caller ID not working
- Fixed DTMF digit ABCD not relayed to VoIP side via RFC2833 and SIP INFO, also SIP_INFO inbound ABCD not played to PSTN
- Fixed device accepts INVITE from proxy only for 2 stage dialing
- Fixed PSTN incoming call by-pass 3CX restricted call after server SIP 403 if caller hold long enough
- Fixed SIP Authentication Password does not support some symbols
- Fixed DTMF CID not used in application

Firmware Version: 1.3.4.10
Name of Product: GXW 410X
Date: December 21, 2009

CHANGES SINCE FIRMWARE RELEASE 1.3.4.9

BUG FIXES

- Fixed device status page showed wrong characters

Firmware Release Notes
Firmware Version: 1.3.4.9
Name of Product: GXW410X
Date: November 12, 2009

CHANGES SINCE FIRMWARE RELEASE 1.3.1.6

BUG FIXES

- Fixed Caller ID issue for “Out of Area” as “Reason for Absence”
- Fixed device sent out DNS A query first when configured with DNS SRV
- Fixed device didn’t reply 200OK and the port didn’t pick up
- Fixed Tone Generation issue in 1 stage dialing
- Fixed Caller ID feature unstable
- Fixed echo with Busy Tone, unit didn’t hang up timely
- Fixed two RTP stream caused bad voice quality
- Fixed device crashed when doing bridge call
- Fixed a memory leak issue

**Enhancements**

- Added support for dual cadence detection
- Added support for Polarity Reversal disconnect on outbound calls
- Added support for “Enable Call Answer Supervision”

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

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

**Changes Since Firmware Release 1.2.1.5**

**Bug Fixes**

- Fixed packet contains extra data after being IP fragmented
- Fixed 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

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 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 r-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 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

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**Firmware Release Notes**

**Firmware Version 1.0.1.10**

**Name of Product**  GXW 410X

**Date:** October 10, 2008

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**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 CCBOM 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 ADVANCED 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