Release Note
Firmware Version 1.0.0.44
May 29, 2007

==

Product:	GXW4004	
Date:	05 - 29 - 2007	
Release Items:	gxw4004base.bin	1.0.0.31
	gxw4004boot.bin	1.0.0.7
	gxw4004core.bin	1.0.0.14
	gxw4004prog.bin	1.0.0.44
Previous Release:	gxw4004base.bin	1.0.0.31
	gxw4004boot.bin	1.0.0.7
	gxw4004core.bin	1.0.0.14
	gxw4004prog.bin	1.0.0.43

==

==

Product:	GXW4008	
Date:	05 - 29 - 2007	
Release Items:	gxw4008base.bin	1.0.0.31
	gxw4008boot.bin	1.0.0.7
	gxw4008core.bin	1.0.0.14
	gxw4008prog.bin	1.0.0.44
Previous Release:	gxw4008base.bin	1.0.0.31
	gxw4008boot.bin	1.0.0.7
	gxw4008core.bin	1.0.0.14
	gxw4008prog.bin	1.0.0.43

==

==

Product:	HT502	
Date:	05 - 29 - 2007	
Release Items:	ht502base.bin	1.0.0.31
	ht502boot.bin	1.0.0.7
	ht502core.bin	1.0.0.14
	ht502prog.bin	1.0.0.44
Previous Release:	ht502base.bin	1.0.0.31
	ht502boot.bin	1.0.0.7
	ht502core.bin	1.0.0.14
	ht502prog.bin	1.0.0.43

==

Build 1.0.0.44   5/29/07

- Fixed layer 3 QoS issue for SIP. Both RTP and SIP packets will all contain this layer 3 QoS (Diff-Serv) setting.
- Fixed tone detection for modem and fax. DIS tone is now used instead of CED tone for callee side Fax tone detection.
- Moved per account/profile CPT setting to advanced setting. System wide ring cadence is also added in advanced setting (P-value 4040).
- Added the support for SRTP. The number of concurrent SRTP sessions is limited to 4. The P-values are the same as other products. There are also 4 new star codes to manipulate SRTP setting:
  *16 – Set force SRTP mode
  *17 – Disable SRTP mode
  *18 – Force SRTP per call
  *19 – Disable SRTP per call
- Added initial hunting group implementation.

Build 1.0.0.43   5/10/07

- Removed 101 provisional response that might cause trouble to some non-conforming proxies/client.
- Fixed missing “sip:” scheme in Referred-By header.
- Fixed space character in config file is not decoded properly.
- Fixed VLAN and Layer 2 QoS support.
- Fixed loop current disconnect on-hook detection problem.
- Fixed direct IP calling problem using voice prompt.
- Fixed GXW400X will respond with busy after make blind transfer under certain conditions.
- Fixed GXW400X does not insert route entry for INVITEs after 302 response.
- Fixed BYE is not supported to terminate early dialog.
- Fixed product ID read problem.
- Fixed configuration file authentication problem.
- Added a configuration option to disable visual message waiting indicator.
- Added the support for NAT options and bandwidth control.
- Added loop current disconnect period.
- Added configurable delayed call forward timeout.
- Added customizable SSL private key and certificate.

Build 1.0.0.39 3/15/07

- Fixed incorrect timer value and the device sleeps forever.
- Added redundant packet in T.38

- Fixed GXW400x does not detect FXS hangup around same time as loop disconnect.
- Fixed GXW400x will crash for direct IP call .
- Fixed errors in time zone setting in web page .
- Fixed GXW4008 will send a bad INVITE to sever after call waiting.
- Fixed GXW400x does not handle special characters in provision file.
- Fixed GXW400x system clock slow down.
- Added LAN MAC address in syslog message.
- Fixed incorrect response in 200 OK for inactive media.
- Changed enable call feature description in web page.
- Fixed incorrect handling of unsupported media type in SDP.
- Added Australia/New Zealand SLIC mode.
- Added the support for symmetric RTP.

Build 1.0.0.37 2/20/07

- Fixed voice prompt cannot be accessed when enable call feature is set to No.
- Removed G.728 from web UI.
- Fixed call waiting tone will mute local audio.
- Fixed DEVICE busy call forwarding does not work if call waiting is disabled.
- Added the support for current disconnect.
- Added the support for T.38.
- Added do-not-disturb service.
- Added call-return service.

- Fixed DEVICE returns 486 BUSY as transferor after blind transfer.
- Fixed DEVICE port cannot be used during blind transfer.
- Fixed DEVICE does not play dial tone after blind transfer.

- Added the feature to disable voice prompt.
- Fixed incorrect time zone setting for Newfoundland.
- Fixed DEVICE does not retry on 401/407 for requests other than INVITE and REGISTER.
- Fixed no ringback tone when a call is blind transferred.
- Fixed DEVICE returns 486 BUSY when dials an invalid number on second channel.

- Fixed incorrect web page display when SIP transport is set to TCP.
- Fixed a timing issue where DEVICE may not retry after 401/407 is received.

- Fixed direct IP call did not work without ‘#’.
- Fixed DEVICE did not reset status to idle after call waiting failure.
- Fixed howler tone being played in the middle of the conversation.

- Fixed crash problem and memory leak when handling 302 response.
- Fixed crash problem when processing call records.
- Fixed DEVICE did not send CANCEL when the transferee of blind transfer hung up.

- Added UPnP support.
- Changed default dial plan to “{ x+ }” .
- Fixed direct IP calling problem via voice prompt introduced in previous version.
- Fixed no reminder ring when the callee hangs up in call waiting case.
- Fixed no SIP CANCEL when canceling blind transfer.
- Changed default G.726-16 payload type to 100.
- Fixed no from-tag when sending INVITE after 302 response.
- Fixed a resource deadlock that results in attended transfer to fail.
- Fixed device crash problem when dials a number and WAN is disconnected.

- Changed DMZ behavior not to route Web and telnet access in WAN port to DMZ host.
- Blocked packets to access LAN IP from WAN port.
- Fixed no IP issue when WAN cable plugged in after system fully boot up.
- Added reboot after config file change and blink LED to indicate in provision state.
- Improved LEC NLP performance to make the transition between silence and voice less noticeable.
- Added the support for dial plan.