GRANDSTREAM NETWORKS

Firmware Release Notes
Firmware Version 1.0.1.15
Product Name: GXW 40XX / HT-502
Date: July 3, 2008

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 1.0.1.8

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