GRANDSTREAM NETWORKS

Firmware Release Notes
Firmware Version 1.2.1.13
Product Name: GXV3000
Date: September 4, 2009

SUMMARY OF UPDATES

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

NOTE: Once upgraded to 1.1.3.x, unit will not be able to downgrade to any previous releases.

CHANGES SINCE FIRMWARE RELEASE 1.2.1.12

BUG FIXES

- Fixed GXV3000 does not lookup correctly on DNS SRV

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Firmware Release Notes
Firmware Version 1.2.1.12
Product Name: GXV3000
Date: July 21, 2009

CHANGES SINCE FIRMWARE RELEASE 1.1.3.50

BUG FIXES

- Fixed GXV3000 crashes after it transfers a call
- Fixed GXV buzzes when remote party picks up on GXE
- Fixed GXV does not reply 180 when using TCP
- Fixed GXV does not follow DHCP renewal time when the DHCP server changes it
- Fixed GXV does not re-query for DNS for SIP servers after DHCP restarted
- Fixed GXV goes in locked state for 30 seconds after network cable is unplugged/plugged
- Fixed GXV fails to send session refresh requests after receiving UPDATE without a Session-Expires header
- Fixed the gateway IP address becomes zero
- Fixed GXV does not display call info when "Start Browser on Boot" is enabled
- Fixed GXV LCD screen saver runs when there is an incoming call
- Fixed GXV does not send out register info when it does not receive reply from DNS server
- Fixed Send key behavior error in LCD WEB browser mode
- Fixed GXV3000 Web UI display issue
- Fixed GXV crashed when conferencing different video codec
- Fixed typo in the list of Factory Functions:Diagnostics
- Fixed out of channel issue for Keep-Alive Using SIP OPTIONS
- Fixed during firmware upgrade, GXV slider moves back and forth
- Fixed ptime values do not relate to SIP account
- Fixed GXV uses the wrong anonymous setting when user presses SEND key to redial
- Fixed when GXV is in a video conference, the Bit rate, tone remapping filter, and volume control indication text doesn't show up on LCD.
- Fixed Attended Transfer fails when incoming call is from anonymous

- Fixed GXV unable to recover Attended Transfer with 4xx REFER response
- Fixed GXV sends mal-formed NOTIFY for Presence
- Fixed the issue that it does not send the corresponding packet if the packet size is over 1500 bytes and Vlan tag is enabled
- Fixed Line can not be released under certain operation. If two calls come almost at the same time when the auto answer is enabled, the first call gets answered, the second will be ringing
- Fixed DNS related problems
- Fixed dial tone disappears if user presses speaker key while previewing ring tone through LCD menu
- Fixed if set local RTP port to 1024, it will be reset to 5004
- Fixed session timer refresher is not correct
- Fixed it takes very long time to get IP through DHCP if server response is slow
- Fixed long life of packet fragmentation causes packets to be dropped

ENHANCEMENTS

- Display UNREGISTERED icon when network is down
- Removed "LCD background color" and "Display Clock instead of Date" options and from the web basic settings
- Added Onhook Dialing Support for GXV3000
- Store the last adjusted value of the auto tone remapping filter and use it as default in future calls
- Making Call History list consistent as GXPs
- Added keypad MENU option to change time zone
- Added wallpaper support
- Reorganized GXV idle screen
- Enhanced GXV GUI widgets
- Immediate screen refresh when quitting menu and going on-hook
- Ported GUI display module from GXPs to GXV to improve the GUI on GXV
- Added "Check SIP User ID for incoming INVITE" option to GXV
- Added Publish for presence option on GXV
- Enhanced GUI color support
- Improved image quality on v4.x HW
- Added support to ring in the speakerphone as well when the headset is detected
- Not to show the firmware upgrade/provisioning window if the files remains the same in the server
- Added support to display ICON and/or Forwarded number on LCD when Call Forwarding is activated
- Added "DO NOT POWER OFF!" message in Chinese when GXV is on Chinese mode

Firmware Release Notes
Firmware Version 1.1.3.50
Product Name: GXV3000
Date: March 19, 2009

CHANGES SINCE FIRMWARE RELEASE 1.1.3.29

BUG FIXES
- Fixed GXV3000 will self reboot when default screen saver is enabled
- Fixed GXV3000 failed to send session refresher at half expiration time
- Fixed GXV3000 locked up when "Keep-Alive Using SIP OPTIONS" is enabled
- Fixed Branch ID does not update for each boot up
- Fixed GXV3000 Click-to-Dial does not put first party on hold under BroadSoft mode
- Fixed GXV3000 fails to send out REGISTER to secondary Sip server on failover under BroadSoft mode
- Fixed GXV3000 fails to send BYE to the secondary sip server on failover under BroadSoft mode
- Fixed Codec Renegotiation: Blind Transfer of Call on Hold failed under BroadSoft mode
- Fixed GXV3000 unable to resume video in a conference when the last dialed number hangs up

- Fixed GXV3000 with Network Screen Saver enabled would crash after a while
- Fixed GXV goes on-hook when the transferee received BYE before it sends out INVITE
- Fixed Low frame rate of RTSP and video loopback factory function
- Fixed if any input box is blank for static IP setting on LCD menu, it will be set to 0.0.0.0
- Fixed when GXV3000 is set auto-answer, after caller called and hung up quickly before GXV can answer, GXV will go offhook in dialing stage
- Fixed GXV3000 SUBSCRIBE for MWI does not follow Cseq
- Fixed the screen does not animate the volume bar during a conference
- Fixed GXV responds to INVITE with incorrect number of media in SDP
- Fixed on GXV3000, the SIP port is always the same after reboot when using "random port" option
- Fixed Mirror button would not pickup the call if the INVITE does not contain SDP
- Fixed Voice_frames_per_TX not used correctly
- Fixed distinctive ring tone does not work for other accounts than account 1
- Fixed caller ID distinctive ring tone does not overwrite video call ring tone
- Fixed GXV camera block button causes text box input to fail
- Fixed blind transfer canceled by transferee will not terminate
- Fixed GXV unable to switch to another conference party video that has different codec
- Fixed when GXV3000 is transferee in a blind transfer, if the target doesn't exist, the phone won't go back to previous call
- Fixed on GXV3000, when RTSP server is not configured on the web, and try to open RTSP stream, the phone tries to send DNS query
- Fixed "Delete all" doesn't work in GXV3000 phonebook
- Fixed on GXV3000, the dial plan prefix doesn't work when the SEND key is pressed for redial
- Fixed on GXV3000, during blind transfer, after pressed some number, and press transfer key again to cancel it, we can hear busy tone after a short period
- Fixed when "Choose Video Codec By Local Preference" is enabled the 3rd configured video codec has no effect
- Fixed there is one way audio when local RTP port is configured as 6000
- Fixed GXV3000 has problem to make calls when GXV3000 disabled SRTP and the other party enables SRTP but not enforced
- Fixed when GXV3000 enforced on SRTP and call a phone with SRTP disabled, GXV3000 keeps playing ringback tone with screen goes back to idle. Now it plays busy tone, and the error message will be displayed
- Fixed per-call call-waiting settings do not expire if remote party hangs up
- Fixed GXV3000 does not clear dialing error message after switching back to a held audio line
- Fixed GXV3000 shows OSD under remote HOLD in video call
- Fixed GXV3000 shows arbitrary characters when switching from Chinese to downloaded languages. Now switching to downloaded languages will implicitly switch to English (a reboot is still required for the downloaded language to take effect)
- Fixed speaker icon remains when remote disconnect a locally held call
- Fixed WEB UI display errors: Status page displayed reversed boot/load
- Fixed DNS client doesn't switch DNS server when one of them is not working

ENHANCEMENTS
- Added support for video bit rate adjustment using SIP INFO
- Added support for entering "space" in phone book
- Updated DHCP ACK options when renewal, reboot
- Added support for showing some message when video request is sent
- Added H264 packetization to 1 for Broadsoft
- Changed default dial-plan to {x+|*x+}
- Added hint text displayed on GXV3000 for "call forward" Similar to GXPs
- Added support for H.263 Annex D (Unrestricted Motion Vector Mode)
- Added support for GXV3000 displaying "REMOTE HOLD" when held
- Added support for timeout symbol in dial plan grammar
- Changed once feature code (*xx) is accepted it is cleared from the screen (before it just append to it like *706, now *70 will be cleared as soon as accepted and you will see 6 only when you dial)
- Added support for SIP URI dialing

Firmware Version 1.1.3.29
Product Name: GXV3000
Date: October 13, 2008

CHANGES SINCE FIRMWARE RELEASE 1.0.1.27

BUG FIXES

- Fixed speaker icon remains when remote disconnect a locally held call
- Fixed screen refresh problem after saving missed call to phonebook
- Fixed RTSP client automatically quit after 60 seconds
- Fixed WEB UI display errors:
    1. Status page displayed reversed boot/load version
    2. Admin password extra colon in Chinese mode
    3. Enable Tone Disconnect missing colon in Chinese mode
- Fixed dial plan does not handle replacement blocks correctly when the dial plan segment contains more elements (like {<2=011>x+})
- Fixed WEB UI string "(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)" typo 32 instead of 30 in Chinese mode
- Fixed GXV3000 does not handle RTP correctly if ptime is different and not present in SDP
- Fixed dial plan allows any pattern to be dialed in a repetitive ('+' or '.') expression
- Fixed WEB UI returns 404 error when login as user privilege and trying to UPDATE
- Fixed Quick IP call fails after making a direct IP call with port
- Fixed some characters with accent (ì, è) to be supported by language pack (a new font is used now).
- Fixed "Enable Paging/Intercom" option is missing in the WEB UI "Auto Answer" field
- Fixed sometimes there is no audio playout (dial tone or call) after soft reboot on HW1.1/3.1
- Fixed RFC2833 & SIPINFO DTMF playout fails. If tone playout longer than 4 secs, a 100 ms tone would be played out
- Fixed when daylight saving time is set to "yes", the time saved in call history is one hour more than the real time
- Fixed RTP port number stays same through reboots. New scheme changes port number even between calls
- Fixed GXV3000 sends INVITE to 0.0.0.0 when no account is configured
- Fixed if firmware server address is invalid, the phone will still keep retrying. Now, changed it to if the domain name can't be resolved, stop retrying
- Fixed if the STUN server address is invalid, the phone floods out DNS queries. Fixed it by changing retry delay to one minute
- Fixed Direct IP call fails to work if previous direct IP call contains port number
- Fixed we respond with malformed 200 OK when we receive 2 different INVITEs with same call leg but different request line
- Fixed we allow to call a number not matching any dial plan
- Fixed HW 4.2 volume remains constant even if increased or decreased
- Fixed provisioning kicks in during incoming call
- Fixed GXV3000 send 200OK with video SDP after receive INVITE of no video supportFixed when restored call from a failed transfer as transferee, cannot terminate or hold a call
- Fixed 3WC display problem
- Fixed 408 response does not contain a human readable part "408 Request Time Out"
- Fixed attended transfer will not work as transfer target if call-waiting is disabled
- Fixed when phone is configured using static IP, it will still send DNS query even though the DNS server IP is 0.0.0.0
- Fixed GXV3000 PHONEBOOK and CAMERA buttons do not work under boot-in-browser mode
- Fixed phone cannot go back to Chinese menu after RSTP Streaming if video surveillance on remote side is disabled
- Fixed we do not follow Expires header
- Fixed we fail to resume to normal state after busy-call-forward with call-waiting disabled
- Fixed the MUTE LED will stay blinking if a call is ended while in MUTE state

- Fixed we do not handle 301 for REGISTER correctly
- Fixed no missed call indication
- Fixed SRTP fails for odd-byte RTP frames
- Fixed we respond malformed video SDP when incoming INVITE is without SDP
- Fixed GXV3000 keeps sending out ARP when SYSLOG server IP address is the same as its IP
- Fixed GXV3000 keep trying unregister when server sends 404
- Fixed GXV3000 will try to establish video when being put on hold even the session was established as audio only
- Fixed Venezuela time zone

ENHANCEMENTS

- Changed once feature code (*xx) is accepted it is cleared from the screen (before it just append to it like *706, now *70 will be cleared as soon as accepted and you will see 6 only when you dial).
- Added support for URI in phone book
- Added configuration to disable OSD by default in video call
- Added option to support fast video update by SIP INFO (RFC5168)
- Added QCIF support for H.263
- Added support for MENU admin password feature, also added phone will exit MENU after 60 seconds of inactivity
- Added an option in menu to force the phone sends out registration request
- Enhanced the browser feature to allow text URL input using keypad
- New bootloader
- Added third option for video vocoder so it is possible to turn H.264, H.263, and H.263+ all on at the same time
- New cleaner algorithm implemented for DTMF and Call Progress tone detection.
- Changed we will give current in-use audio codec higher priority for reINVITE (so if PCMU is currently used in a call and incoming INVITE contains both G729 and PCMU, we will choose PCMU even if G729 is on top of the list). This avoids a buzz sound during the codec switch.
- Changed we will relax the profile level check in H264 fixing an interop problem with Nortel video phone causing green screen on GXV3000
- Added new option "Choose Video Codec By Local Preference" to Account Settings Page
- Added support for keep-alive by SIP OPTIONS
- Recalibrated handset & headset TX gains for v4.2 hardware
- Added "No, but allow in-call enabling" to "Enable Video"; when configured this mode all calls will default to voice calls and incoming video calls will be answered in voice only
- Changed SDP parser MIME type matching to case-insensitive
- Change delay RTP restart to restart after receiving ACK (it was restarting prior to sending 200 OK), this happens when we receive a reINVITE with video during a voice call.
- Added provisioning protection: MENU access is not allowed during provisioning and before initial provisioning completes
- Added audio volume display in OSD
- Added support for "Save to phonebook" from call history
- Added support for "Offhook Auto Dial"
- Added support to allow replace block in the middle of dial plan segment
- Reduced jitter buffer delay by 50ms
- Changed default H.264 level to 2.0
- Changed H.264 bogus packets to use SEI
- Changed default packetization mode to 0
- Added support for phonebook name lookup for call logs
- Added support for Simplified Chinese as an embedded language (no language pack download required), no reboot required if switching to Chinese or English (reboot required if switching to downloaded language)
- Added the support for tone remapping filter. The P-value is 948
- Included load64 1.1.3.2
- Changed Phonebook MENU background color to white for consistency
- Change all code filename suffix to 'b'

CONFIGURATION UPDATES

N/A

NOTES

1.1.3.x can not be downgraded to any previous releases.

CHANGES SINCE FIRMWARE RELEASE 1.0.1.20

BUG FIXES

- Fixed attended transfer issue
- Reset H.264/H.263 decoder when incoming RTP SSRC changes
- Fixed H.263+ crash problem with Eyebeam
- Fixed MUTE/DEL button not effective in the browser mode
- Fixed remote and local hold problem
- Fixed boot loop problem when default web page is set to blank
- Fixed IP address refresh problem
- Fixed DHCP failure when cable is not connected during reboot
- Fixed Call-Info header syntax when generating paging request
- Fixed H.264 encoder may generate two adjacent IDR frames
- Fixed SRTP parameter not sent when receiving 183 without SRTP
- Fixed memory corruption caused by heavy TCP traffic
- Fixed RTP padding bit support

ENHANCEMENTS

- Changed the screen capture to use BMP format instead of YUV
- Added the support for asymmetric dynamic payload type
- Added the support for VGA resolution decoding
- Added +sip.instance and reg-id parameters in SIP REGISTER request according to sip-outbound draft
- Enabled camera sharpening filter
- Added full H.263 decoder support (experimental)
- Showed boot up screen when start up
- Added STAP-A decoding support for H.264
- Added the support for RTSP streaming using H.264 (RFC3984). The following fields are added:
  Streaming RTSP Server (P953)
  Streaming RTSP User ID (P954)
  Streaming RTSP Password (P955)
  This feature can also be used to view surveillance video from another GXV-3000 or GXV-410X. After these fields are configured, users can go to LCD menu and start the RTSP client to see the video.
- Added *78 call pickup for PingTel SIPExchange
- Added the support for HTTP based network screen saver. There are two P values added: P943 to turn it on and off and P942 for server URL. The image must be JPEG base line with dimension of 320x240. Adobe Photoshop processed images are known not to work with GXV-3000.
- Added the support for weather forecast. There are two P values added: P945 to turn it on and off and P944 for server URL. A test server is available at 67.153.142.74/weather/weather.php. This server only supports US weather forecast query. For example, to query for zip code 02446, the server URL shall be configured as 67.153.142.74/weather/weather.php?z=02446. The XML schema is provided with the release.
- Added the support for RSS 2.0 feed. There are two P values added: P947 to turn it on and off and P946 for server URL.