Release Note
Firmware Version 1.1.1.14
October 2, 2006

Firmware 1.1.1.14 has major changes (compare with 1.1.0.x Releases) which may requires firmware to be downloaded twice(and reboot itself), and it can not be downgraded to previous version.

Make sure all the files that come with Release_GXP2000-BT200_1.1.1.14.zip is unzipped into the TFTP or HTTP server.

For any firmware upgrade from 1.0.1.x or 1.0.2.x, please refer to previous release note and firmware and upgrade them to 1.1.0.16 first.

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Product:	BT200	
Date:	2006-10-2	
Release items:	boot55b.bin	1.1.1.3
	bt200b.bin	1.1.1.14
Previous release:	boot55b.bin	1.1.1.2
	bt200b.bin	1.1.1.13

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Product:	GXP2000	
Date:	2006-09-28	
Release items:	boot55b.bin	1.1.1.3
	gxp2000b.bin	1.1.1.14
Previous release:	boot55b.bin	1.1.1.3
	gxp2000b.bin	1.1.1.13

Build 1.1.1.14 10/2/2006
- Fix Missing Extension Module issue.

Build 1.1.1.13 9/28/2006
- Audio adjustments
- Fixed some v 0.3 and 0.4 blank LCD issue
- Added "application/dialog-info+xml" in the Accept header of 415 response
- Added Syslog for SIP dialog matching result

New boot55b.bin 1.1.1.3 9/28/2006
- Fixed init problem causing the GXP-2000 v1.1 bootup problem

Build 1.1.1.12 9/15/2006
- No change for GXP-2000 from 1.1.1.11
- Disable the EMIF optimization which caused some BT-200 to fail to upgrade.

Build 1.1.1.11 9/14/2006
- Fixed GXP-2000 crashes when a very large phonebook file is downloaded under some certain scenarios

Build 1.1.1.10 9/12/2006
- Fixed GXP-2000 crashes when a very large phonebook file is downloaded
- Fixed GXP-2000 crashes when "Remove Manually-edited entries on Download" is set to Yes
- Fixed GXP-2000 Name is not displayed for multi-functional keys on EXT
- Fixed SIP stack incorrectly parsed "CT" header
- Support for displaying Account Name in lieu of date for BT-200 idle state
- Turn on this option by provision parameter P339 (1: use Account Name, 0: use date). First 12 digits are displayed, aligned to center (odd length 1 slot to the right), undisplayable characters will be blank
- Support for 5 provision attempts
- Fixed GXP-2000 take account 1 information when replying the missed call for account 2
- Fixed GXP-2000 Speaker mode is triggered on 2nd call when the first call ended
- Fixed BT-200 turns speaker on when MSG key is pressed even when no voicemail user ID is configured
- Fixed BT200 does not send “cancel” after being pressed “FLASH” ,when callee is ringing
- Fixed audio quality degraded when call-waiting tone is been played
- Fixed we cannot correctly parse incoming SIP messages with Contact headers that come without a username part causing some Broadsoft test cases to fail
- Support for Broadsoft click-to-hold (Allow-Events: hold)
- Fixed G.723 6.3kbps decoder does not work, web UI enabled this option
- Fixed BT-200 volume control via arrow keys fails
- Increased speakerphone mic gain by 7.5db

Build 1.1.1.9 8/9/2006

- Improved audio quality
- Fixed BT-200 does not play busy tone to signal call fail after it terminates a call when SRTP is enforced
- Modified memory management for iXML parser. This should resolve the freeze on downloading 50-record phonebook XML problem
- Fixed several GUI menu bugs
- Fixed GXP-2000 does not save after more than 30 extension entries
- Fixed GXP-2000 does not store UserID for KEY36
- Fixed GXP-2000 cannot answer incoming call when in the SIP proxy edit screen
- Fixed the screen XML '$d' variable does not display correctly
- Fixed BLF does not activate speed dial when BLF party is in use
- Fixed GXP-2000 continue to ring when BYE is received for early dialog
- Fixed we do not clean out the call properly when terminating a call due to SRTP not enforced
- Fixed we will perform firmware upgrade even if configured not to when DNS query for config server failed/we query "0.0.0.0" when configured such in firmware/config servers
- Fixed a memory-leak issue that is only exposed by how GXP-2000 handles attended transfer (does not apply to other products)
- Fixed GXP-2000 does not place transferee on hold when attempting to transfer
- Extend the original "Disable missed-calls" feature to allow a new mode to disable all call-logs on a per-account basis. P182/442/542/642: old values (0/1), new values (0/1/2) where 2 means disable call-log.
- Fixed both GXP-2000 and BT-200 turns speaker on when MSG key is pressed even when no voicemail user ID is configured
- Fixed a potential crash if a NOTIFY with bad dialog XML
- Added a memory debug feature: on right-top corner current memory status is displayed in lieu of time (or date, if reversed) in the format of x/y where x is the current usage and y is the peak usage

Build 1.1.1.7 7/13/2006

- Fixed BT-200 LCD blink when G.723 is in use
- Fixed BT-200 rings even the held party hangs up
- Fixed custom ring tone by Alert-Info fails
- Fixed offhook auto-dial is not enabled on BT-200
- Added option to check incoming INVITE sip user ID
- Fixed DTMF buffer not cleared when switching lines for unestablished dialogs
- Support disable call-waiting tone for GXP-2000
- Add UCF (Unconditional Call Forward) icon on status line for GXP-2000
- Fixed high pitch done played when Call Forwards are enabled and disabled
- Fixed BT-200 does not ignore CONFERENCE and FLASH key during conference
- Fixed user cannot enter * and # in phonebook entries. In addition, user can enter @ by using HOLD key in phonebook submenu

- Fixed we crash on attended transfer on platforms that use To/From headers without square brackets
- Fixed BT-200 keypad UI for TFTP server not working
- Fixed we still responds "recvonly" on un-hold SDP message
- Fixed GXP-2000 ring tone change via keypad menu not effective after reboot
- Fixed BT-200 does not save handset/speaker volume change
- Fixed BT-200 does not save speaker volume over reboots
- Added volume control is stored after reboot
- Added Support for GXP2K-EXT keys in diagnostic mode
- Disabled headset side tone
- Fixed IP Fragmentation bug
- Add Support for IM and screen XML feature (saving to flash)
- Fixed we send NTP to wrong IP address
- Added force LCD update on hook status change (this makes LCD GUI look more responsive when onhook)
- Added customizable idle screen via downloading XML by HTTP/TFTP
- Added support for SIP MESSAGE method (RFC 3428); stores up to 100 incoming IM messages, after that new messages are dropped
- Added support for SIP PUBLISH method (RFC 3903)
- Added support for SIP Presence package (RFC 3856, 3863) for use of 7 MFKs and GXP-2000EXT
- Added support for SIP Dialog package (RFC 4235)
- Added support for SRTP by SDES
- Fixed GXP-2000 crashes when speed dial user ID contains '@'
- Fixed the clock on the right top corner displays incorrectly if switches from 12hour display to 24hour display.
- Added support for G.726 codec
- Added support for GXP-2000 Extension console.
- Added support for anonymous call using privacy header
- Added support for downloadable phonebook

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Build 1.1.0.16

- Fixed IP Fragmentation bug
- Fixed GXP-2000/ ring tone change via keypad menu not effective after reboot
- Improved audio quality with some audio parameter changes
- Fixed we crash on attended transfer on platforms that use To/From headers without square brackets
- Fixed we reject cfg files smaller than 512 bytes
- Fixed BT-200 keypad UI for TFTP server not working
- Fixed a typo in LCP (PPPoE related)
- Fixed GXP2000 provides RING only for first incoming call; when first caller hangs up, the ringing stops.
- Fixed BT-200 does not play short beep on auto-answer
- Fixed Bug in Via header when DNS name is used instead of IP address
- Fixed we still responds "recvonly" on un-hold SDP message

Build 1.1.0.13 5/16/2005

- Add Quick IP Calling mode
- Fixed GXP-2000 Speed Dial/Asterisk BLF pick up broken in 1.1.0.12
- Fixed GXP-2000 crashes when a very long DTMF string is dialed
- Fixed SIP NOTIFY to event REFER violating RFC 3515
- Fixed we do not affix To-tag for PRACK request
- Fixed we do not use new branch for PRACK request
- Fixed we do not include Contact header in 180
- Fixed we do use random port for RTP even if random port is set to yes
- Fixed the ping problem when the device is in router mode
- Enabled the broadcast drop mode (this should improve the switch performance for multicast and broadcast)
- Fixed the 3-way conferencing issue when the re-invite to bring the 1st party out of hold status gets challenged.
  This is difficult to verify. Basically our 3-way conferencing can be broken into the following steps:
  - A invites B
  - A re-invites B to put B on hold
  - A invites C
  - A re-invites B to put B out of hold status

  If at step d) the proxy challenges the request with 401/407, we couldn't complete the conferencing
- Fixed the PPPoE TCP problem
- Added idle timers to fix more idle screen blackout cases
- Fixes for blank LCD or corrupted GUI issues
- A-Tick and DC filter changes
- Fixed crash on incoming call when all channels are in use
- Fixed lost registration problem
- Fixed we will never switch DNS server even if primary DNS server failed to respond and there is a secondary DNS server
- Changed for GXP-2000: once entering direct IP calling mode, the cursor focus is in the text field instead of CANCEL button
- Reduce the GXP2000 handset earpiece audio level by 4.5 dB
- Fixed GXP-2000 crashes when using MISSED CALL GUI to dial out
- Added use of MUTE/DEL key during incoming call ringing state will reject call using 486
- Added MUTE/DEL key will act as toggle key to turn DND on and off during idle
- Fixed GXP-2000 direct IP call cannot specify port.
  Note that a new input method is specified here: you will use * to enter dots (separator between octets) and use # to enter colon (separator for port). So you can enter "10.10.12.135:5068" using "10*10*12*135#5068". This is probably more intuitive
  Note 2: Direct-IP calling feature is further cleaned out so that STUN mapped info is not used when we detect direct IP calling destination is in local subnet (From and Contact headers are also cleaned to use IP address only, not including the configured SIP URI).
- Fixed AGC setup change

- Fixed RTPSend bug
- Fixed we do not use the previous SSRC, timestamp, and sequence number after restoring a previously hold call
- Fixed static IP problem in 1.1.0.2
- Fixed we start sending RTP when restoring a call before we receive 200 OK Fixed we do not clear out CallFwd settings when user configure to disable call features
- Added special factory workaround mode-when configured to use static IP 192.168.0.160 and no gateway IP is configured, provisioning is skipped
- Added support for Broadsoft Click-to-Answer feature using "talk" event
- Fixed GXP-2000 cannot make direct IP calls
- Fixed GXP-2000 factory MAC-Edit function cannot change last digit to A-F
- Fixed redial does not append the dial-plan prefix
- Fixed we will retry 5 times if only config server is configured and there is config file
- More LCD fix for GXP-2000
- Fixed some crashes Issues

Build 1.0.2.13 2/21/2006

- Added support for DHCP option 2 (Time Offset). Provision variable P143, possible values 0/1. When set to 1, it will override the configured Time Zone setting if available. Default is No (0).
- Solved the LCD problem.
- Fixed keypad not responding if boot up without network cable
- Fixed when LINE keys are pressed when in GUI MENU, the usual "Dial Using" prompt is not displayed
- Added support for DHCP option 42 (NTP server). Provision variable P144, possible values 0/1. When set to 1, it will override the configured NTP server. Default is No (0).
- Added support for DHCP option 66 (TFTP server). Provision variable P145, possible values 0/1. When set to 1, it will override the configured provision path and method. Default is No (0).
- Added configurable DHCP options 12 (host name/P146), 15 (domain name/P147), and 60 (Vendor Class ID/P148). Max length allowed is 32 bytes each.
- Fixed we do not follow the port in the Record-Route URI if there is maddr attribute
- Added we will attempt to start the initial provisioning routine every minute for 5 times unless we received any response from the server (any type of HTTP error or OK response, and TFTP error or data response will stop further trying)
- Changed we will restart STUN accounts when STUN port mapping changed (previously we only check IP address change). Also changed: STUN checking interval is now the same as the configured keep-alive interval (used to be 1 minute fixed)
- Changed the RTP keep-alive interval (when a call is placed on hold) to the same as the configured keep-alive interval (used to be 25 seconds fixed)
- Fixed we do not send RTP keep-alive interval during MUTE
- Fixed when early-dial is in use, the phone times out 4 seconds after the last incomplete digit is dialed instead of the configured no-key time out value configured. Also fixed deletion of an incorrect timer due to this action.

- Fixed we still attempt to start conference with 2x GSM channels with Asterisk due to its misbehavior on codec negotiation
- Fixed a string matching function which made our password checking case-insensitive (it should be case sensitive)
- Fixed VLAN TCP issue
- Added model number and firmware version (app) in every Syslog message
- Changed- In GUI Status menu item, "N/A" is displayed instead of "NO" for accounts configured to be not-in-use
- Fixed (temporarily) the 3WC related problems by application restricting 1PCM+1LBR conference rule
- Fixed memory error Syslog not including right source
- Fixed under "Ring Volume" the volume is decreased by 1 each time you enter that menu item even if you did not make any change
- Fixed we do not detect duplicate firmware (and fall into reboot loop) if the provisioning server responds first data frame < 512 bytes
- Fixed GUI sometimes send additional key events (cause of the SIP configuration problem). Now the focus is default at the "CANCEL" button in each SIP edit dialogs. This also resolves the Upgrade menu inaccessible problem
- Fixed GXP displays "TFTP Provisioning" briefly before correcting it to "HTTP Provisioning" if HTTP is in use
- Fixed ring tone file problems
- AGC change
- Changed- FUNC keys would act as speed dials even when BLF status is BUSY
- Changed- All keys are blocked when provisioning is in progress (including LINE keys and SPEAKER key)
- Changed- Factory reset will now clear phone book as well as custom ring tones
- Changed- GUI->Config->Upgrade changed to allow edit firmware and config server (the original interface was implemented before the application TFTP), Note that the old 12 digit IP address format you use for direct IP calling is no longer valid here. You will need to type in the dots (* key) to separate the octets.

Fixed we fall into reboot loop when there exists a ring3.bin in the server and ring2.bin spans over 64k

- Fixed we do not reboot immediately upon receiving a cfg file (that caused a change)
- Changed- we no longer reboot if only ringx.bin are downloaded
- Fixed we display line status as MUTE if the previous call is in MUTE and disconnected by remote party
- Fixed when account name length > 16 characters, GXP-2000 freezes after a call
- Added allow user to use the Speed Dial keys to do blind transfer
- Fixed we do not process the IP packets if the first fragment did not arrive first
- Fixed we still display AM/PM even when 24 hour display mode is selected
- Fixed our LCD backlight does not light up immediately when a call comes in
- Added- LCD backlight stays light up when there are unviewed missed calls to alert user, the LCD backlight also stays up whenever the MENU operation is in progress
- Fixed GUI SIP configurations were not accessible
- Changed- GUI menu item sequence rearranged for user-accessibility, Status page items rearranged for a quick glance at registration status
- Changed- when display mode is set to DDMMYYYY and reverse date/time is set to YES use period as separator instead of hyphen (I have seen many Europeans request

for this and this does make sense so that when you see a date 03.04.2006 you can tell if 03 is the month or 04)

- Changed- when display mode is set to DDMMYYYY and reverse date/time is set to NO, the long date string is displayed as "dddd, dd MMMM" where "dddd" is day-of-week in English, "dd" is day-of-month in number, and "MMMM" is month in English, example: "Friday, 27 January" (standard mode is "dddd, MMMM dd")
- Added allow disable miss-call features as per-account setting, changing this setting takes immediate effect without reboot. Incoming call is still logged, only missed calls are not. It can be provisioned using P182/442/542/642, valid values 0 and 1, default is 0 (No) which WILL log all missed calls.
- Added disallow MENU actions when provisioning is in progress
- Changed-when you use the UP arrow key to view missed calls, you will return to main idle screen directly if you either use the LEFT key or delete all missed calls
- Changes to phonebook: on the main Phone Book menu, the "Add" is renamed to "New Entry", in the New Entry page the "Add" is renamed to "Confirm Add", "Back" is renamed to "Cancel & Return"
- Fixed phonebook entries remains in flash (and reappear after reboot) after delete

Build 1.0.2.3 1/24/2006

- Added GUI Interface
- Added display day-of-week, display name/extension (account 1) on idle screen (name and extension on idle screen will display in non-bold font when not registered)
- Added allow configuring TFTP provisioning using URL involving FQDN or IP address and file path (previously only IP address is allowed), previous TFTP IP address provisioning variables obsolete
- Added allow user to specify different URL for configuration file and firmware files.
- Added option to authenticate configuration file
- Added different syslog messages when firmware/config files are not downloaded/accepted instead of a single error message
- Added router mode (so GXP-2000 can act as a router) and associated
- Added option "WAN respond to ICMP", default to No, only activated if device is under router mode
- Added Address Book
- Added Call Log with timestamp for incoming calls, outgoing calls, and missed calls, stores up to 50 calls per log.
  Design: the original call log (in/out) are replaced by the new one. In the GUI MENU, there are 3 options under the "Call History" section: Received Calls, Dialed Calls, and Missed Calls.
- Added RNK mode to all accounts
- Fixed we will drop NTP response if the Leap Indicator is non-zero
- Added we will fall back to the origin endpoint of the incoming SIP message if the Via header URI is in FQDN form
- Fixed we cannot handle the /1 after the port number in the SDP m-line (G729r8 problem reported)
- Added "paging" by using "answer-after=0" parameter in Call-Info header.
  When offhook, you see "LINEx: DIAL USING" you can press the ROUND button and you will see "LINEx: PAGE USING" you can toggle between the modes by pressing the button BEFORE any DTMF digits are dialed

On the callee side you have to make sure the newly added "Allow Auto Answer by Call-Info" is set to Yes, it is also recommended to set "Turn off speaker on remote disconnect" to Yes to avoid the busy tone when remote party hang up.
Note: this does not work with Asterisk or other proxies that does not pass along the Call-Info header. There are workarounds for Asterisk (server side setup).

- Fixed GXP-2000 append ";user=phone" twice in Referred-By header when enabled
- Fixed GXP-2000 always miss out the last 4 bytes of the URI in REGISTER messages with authentication
- Shortened the wait-time between downloading files from 1 second to 100ms
- Added configurable T1 timeout interval--this is a per-account setting. Possible values are 0.5sec/1sec/2sec. Provision parameter P209 (P440/540/640 for accounts 2-4 on GXP-2000), valid values 50/100/200 (in 10ms unit). Invalid values ignored, default value 1 second.
- Added configurable T2 interval--this is a per-account setting. Possible values are 2/4/8 sec. Provision parameter P250/P441/541/641, valid values 200/400/800 (in 10ms unit). Invalid values ignored, default value 4 seconds.
- Added implementation of T2 timer (see RFC3261) and send BYE when 200 OK (for INVITE) time-out.
- Added under Broadsoft mode, DNS SRV fail-over happens after 3 retries (so if you set T1=0.5 sec, it takes 7.5 seconds to fail-over to second server).
- Fixed session-timer does not work properly before session establishes (UPDATE/481 issue reported)
- Fixed we do use increment CSeq in the INVITE to unhold when 100rel (PRACK) is enabled
- Fixed during MUTE the call timer does not gets refreshed
- Added LCD displays provisioning status and warning message when flash writing/erasing is in progress
- Added provisioning protection- During provisioning all incoming SIP packets will be dropped without processing
- Added we will do TFTP/HTTP provision upon DHCP/PPPoE completion if no IP address was available initially
- Fixed VLAN bug
- Fixed several TCP/HTTP bugs
- Fixed TFTP retry issue
- Fixed we respond 200 OK with "event: presencenoevent" for the SUBSCRIBE (event: presence) we receive
- Fixed "#" would be included in the end when transferring even when it is configured to be dial key
- Added Asterisk Busy Lamp Field support (BLF)
- Added allow auto-answer by Call-Info option per-account
- Added option to turn off speakerphone automatically on remote disconnect per-account
- Added support for Packet Loss Concealment for PCM and GSM
- Fixed speed dial will still dial even if the account is set to not "in-use"
- Fixed a bug that we incorrectly stops ring when remote CANCEL an unanswered call
- Fixed we do not send INVITE to the updates address if 302 specifies in IP address form
- Fixed G723_SID typo

- Added we do NOT accept incoming auto-answer call when phone is not in idle

Product: GXP-2000
Date: 2005-08-15
Release items: boot55.bin 1.0.1.2
gxp2000.bin 1.0.1.12
Previous Release: boot55.bin 1.0.1.2
gxp2000.bin 1.0.1.9
New features
gxp2000.bin:
• Added Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
• Added sidetone support
• Added new time zone Newfoundland GMT-3:30
• Added support for Authenticating configuration file before accepting changes
• Added capability to handle multiple 18x with SDP redirection
• Added "No STUN but send keep-alive" mode
• Added use "Anonymous" as display name when block CallerID is selected under Nortel MCS mode
• Added we ARP for SIP server/proxy if it is in same LAN prior to registration
• Added Factory reset not allowed if KEYPAD is locked
• Added more information to Syslog (primarily to identify between accounts)
• Added MENU item Ring Volume (after Ring Tone) and removed legacy ringvolume adjustment
• Changed UI MENU items 1-6 and 13 becomes accessible even if Locked Keypad is set to yes
• Added MENU item Download Mode (TFTP/HTTP)
Note: From now on setting/changing TFTP IP in MENU item does NOT turn on TFTP automatically. Use the new Download Mode to ensure the appropriate mode is chosen (TFTP/HTTP mutual exclusive)

Bugs fixed
gxp2000.bin:
• Fixed we do not re-register using updated stun response when WAN IP changes
• Fixed we use different To-tag in 180 (for INVITE) and 200 (for CANCEL) when remote party disconnect an incoming call before it is answered
• Fixed we use different call leg information for SUBSCRIBE (MWI) through out the session--we should use new From-tag and Call-ID in the first SUBSCRIBE and then reuse them plus the To-tag in future SUBSCRIBE/NOTIFY transactions (RFC 3842)
• Fixed when switching line, current call gets dropped if it is in MUTE state
• Fixed we incorrectly parses the expires value during registration
• Fixed we always send REGISTER to SIP server although outbound proxy is configured when both are in IP address form
• Fixed NTP does not work when NTP server is in local subnet
• Fixed DNS SRV entries not sorted correctly
• Fixed we unnecessarily encode certain non-reserved characters in the Replaces

parameter of Refer-To
• Fixed we do not process comma correctly in the display name portion of a Contact header
• Fixed we do not register to accounts using STUN when STUN server is down
• Fixed SIP responses may be using wrong account info when no channel has been allocated to it yet (typically 488 and 422 responses)
• Fixed PPPoE relay-session bug
• Fixed we cannot handle Call-ID over 80 bytes long correctly
• Fixed Referred-By header to use SIP domain instead of local IP
• Fixed PRACK (100rel) does not use the To-tag as given in 18x
• Fixed 100rel kicks in when called by a party that supports 100rel even if it is configured as disabled
• Fixed we use different To-tag in 180 and 200 to INVITE when 100rel is enabled
• Fixed speed dial (web UI) does not take some name strings correctly
• Fixed speed dial does not dial for accounts not registered (it will now dial either when an account is registered or if the account is configured to "not" register)
• Fixed MENU item TFTP server change does not reflect in WEB UI
• Trigger Session-Timer even if "Require: timer" is not present for maximum compatibility to certain incomplete timer implementations
• Adjusted T1 timer

Known problems

gxp2000.bin:
• G.723 decoder only works in 5.3k mode and may produce bad voice quality when incoming G.723 stream is encoded in 6.3k

Build 1.0.1.9 6/1/2005

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Product:	GXP2000	
Date:	2005-06-01	
Release items:	boot55.bin	1.0.1.2
	gxp2000.bin	1.0.1.9
Previous release:	boot55.bin	1.0.1.2
	gxp2000.bin	1.0.1.8

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New features

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gxp2000.bin:
- Added support for Disable Call-Waiting
- Added support for system-wide do-not-disturb
- Added 3 way conference (2PCM or 1PCM+1LBR)
- Added support for GSM/FR vocoder

- Added support for G.723.1 5.3k vocoder
- Added Per-Account Codec preference setup
- Added LCD clock supports 24-hour format
- Added LCD displays remote party name/number when available
- Added LCD displays current codec in use during active call
- Added LCD displays number of messages waiting during off-hook
- Added LCD displays name and number for call logs (received and dialed)
- Added reboot button on the save_ok.htm page
- Added TRANSFER state can be cancelled by pressing the original LINE button (or TRNF key)
- Added play a brief tone before auto-answering
- Added if remote disconnect a call that is auto answered and in speakerphone mode, we do not play busy tone and go back to idle mode silently
- Added Syslog support
- Added call timer display
- Added option to keep LCD backlight always on (Basic Settings)
- Added we sent 400 BAD REQUEST to CANCELs received for an established dialog
- Added IP address display in IDLE state
- Added option to reverse clock/date display at IDLE
- Added more descriptive message when call failed

Bugs fixed

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gxp2000.bin:

- Fixed some one-way audio issues
- Fixed HTTPd/TCP not following negotiated MSS/MTU
- Fixed support for Gratuitous ARP
- Fixed registration-expires header issue
- Fixed User cannot lock/unlock keypad via web interface
- Fixed cannot get an IP address from a DHCP with a VLAN set up
- Fixed Caller-ID announcement ring tone does not work except for the first call
- Fixed dial-plan prefix: it no longer applies to VM Ext, Speed Dials, Calls made from call-logs
- Fixed GXP-2000 does not send out RTP traffic if previous call is terminated by remote party against Asterisk
- Fixed support for custom ring tone
- Fixed remote party can still hear audio after TRNF key is pressed

- Fixed phone does not ring when call came in while offhook and then onhook again
- Fixed SEND key will dial empty string when pressed before any previous numbers are dialed
- Fixed early dial with Asterisk only work for first 2 digits
- Fixed blind transfer will not be activated unless dial key is pressed (now it will kick-in after 4 seconds or whatever is configured)
- Fixed Account 1 cannot change ring tone from web interface
- Fixed ring tone file versions/date not displayed correctly in UI menu and customized ring tone triggered by caller-ID
- Fixed GXP-2000 only displays first digit of called number when early-dial is used
- Fixed GXP-2000 stores multiple entries of incomplete numbers in dialed call logs when early-dial is used (eg. if 611 is dialed, you would see '6', '61', and '611' in called log)
- Fixed GXP-2000 crashes when the "REBOOT" button is pressed in WEB UI after GXP-2000 changed to DHCP and was booted up in static IP mode
- Fixed device lock-up issue when multiple RTCP packets are embedded in a single UDP
- Fixed the issue with MD5-sess authentication

Known problems

- G.723 decoder only works in 5.3k mode and may produce bad voice quality when incoming G.723 stream is encoded in 6.3k