Release Note
Firmware Version 1.1.4.16
July 8, 2007

Firmware 1.1.4.16 has major changes (compare with 1.1.1.X or 1.1.2.X and 1.1.3.X Releases) which may requires firmware to be downloaded 3 Times(and reboot itself) and may take more than 8 minutes, and it *CAN NOT BE DOWNGRADED* to previous version.

Make sure all the files that come with Release_BT200_GXP2000_GXP2020_1.1.4.16.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 1.1.1.16 first.

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Product:	BT200	
Date:	2007-6-15	
Release items:	boot55d.bin	1.1.4.5
	bt200d.bin	1.1.4.16
Previous release:	boot55c.bin	1.1.4.5
	bt200c.bin	1.1.4.14

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Product:	GXP2000	
Date:	2007-6-15	
Release items:	boot55d.bin	1.1.4.5
	gxp2000d.bin	1.1.4.16
Previous release:	boot55c.bin	1.1.4.5
	gxp2000c.bin	1.1.4.14

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Product:	GXP2020	
Date:	2007-6-15	
Release items:	boot55d.bin	1.1.4.5
	gxp2020d.bin	1.1.4.16
Previous Release:	boot55d.bin	1.1.4.5
	gxp2020d.bin	1.1.4.14

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Build 1.1.4.16 (07/06/2007)

- Fixed GXP2000 noise on call hang up
- Fixed GXP does not allow change network settings when "Lock Keypad Update" is set to Yes
- Fixed GXP2020 screen does not clear cleanly when going offhook (some pixels left over from the date string at idle screen)
- Fixed GXP2020 does not show SRTP icon when making an SRTP call
- Fixed GXP2000 increased handset background noise in 1.1.4.14
- Fixed BT200 does not play ring tone after call-waiting came in and onhook
- Added BT200 onhook call-on-hold reminder message (port from BT100)
- Fixed BT200 HOLD issue: hanging up the handset while having a call on HOLD will terminate the call
- Adjusted GXP2020 GUI line info display for multiple call scenarios
- Fixed handset ringer level too loud for GXP2000
- Disabled silence suppression for G723/G729 to bypass the crash the problem
- Fixed we will default to PCMU regardless of configured codec when incoming SDP contains video
- Added option to disable use of multiple media attribute in SDP to workaround some platforms not supporting it. Provisioning parameters P137/487/587/687/1787/1887, takes immediate effect without reboot.
- Fixed GXP2020 does not show "Sync Phonebook XML..."
- Fixed we use ";user=phone" in BLF SUBSCRIBEs even when "User is phone" is not selected
- Fixed we will only update the first BLF/Presence status
- Fixed GXP2000 high pitch sound when offhook using the handset
- Fixed we do not honor session-timer refresher party when other the parameter contains a space between semicolon and "refresher=" parameter
- Added support for Nortel MCS server-side conference
- Added provisioning status on BT200
- Added GXP offhook auto dial
- Added support for BT200 provisioning display (Bugzilla #748)
- Added support for new factory parameter section (to be removed for external release note)

Build 1.1.4.14 (06/14/2007)

Bug Fixes:

- Fixed SRTP broken in 1.1.2.1
- Fixed SRTP sequence number wrap around bug
- Fixed we include "application/xpidf+xml" in Accept header for SUBSCRIBE
- Fixed GXP-2000 Accounts 1 and 4 "SIP T1 Timeout" last option incorrectly labeled as "1 sec" instead of "2 sec"
- Fixed we stuck at provisioning when receives a 200 OK with 0 content-length
- Fixed DoS issue by WWW-Authenticate header
- Fixed a crash issue with 489 Bad Request caused by Proxy-Authenticate header
- Fixed we cannot parse DST string via configuration file correctly
- Fixed ring tone download goes to incorrect TFTP server
- Fixed we still claim an IP address after receiving DHCP NAK
- Fixed description for "Enable Call Features" in web UI
- Fixed we incorrectly formed the qop param without the quotation
- Fixed we incorrectly parsed the nonce param
- Fixed we incorrectly handled presence NOTIFY
- Fixed we incorrectly handled dialog NOTIFY
- Fixed we incorrectly responds 481 to refer NOTIFY when BYE arrives first
- Fixed we accept broadcast SIP messages
- Fixed phonebook download account index incorrect
- Fixed GXP-2000 crash with EXT board
- Fixed GXP-2020 always sends signal=2 in DTMF via SIP INFO regardless of the actual DTMF
- Fixed the in-call timer does not tick when a call is on MUTE
- Fixed the offhook status line disappears between DTMF digits and in-call
- Turn MWI off when phone is offhook
- Fixed phonebook download account index incorrect
- Fixed CBCOM mode results in no audio
- Fixed we display "HV: 0.4.255" on some old GXP-2000 hardware
- Fixed Sylantro interop issue
- Fixed we do not honor maddr parameter in SIP Contact header
- Fixed we used the cached "realm", "nonce", or "opaque" parameter if they are 0 length
- Fixed we do not display correct IP address after DHCP NAK
- Fixed GXP2000 crash if you add entry to phonebook from call log and edit name/number field
- Fixed we cannot authenticate using auth-int
- Fixed a TCP interface bug and a HTTP server bug which should speed up web UI and fixes some display issues

New Features/Changes:

- Improved audio quality
- Changed description for "Enable Call Features" in web UI
- Changed we use 408 instead of 487 for ringing-no-answer
- Added wrap-around support for MENU UI
- Added speed searching phonebook and other list box items

- Added option to support mute speaker ring in headset mode (P336)
  --Note that this option does not exist for GXP-2000 HW 1.0
- Added we will clear new missed calls display message after viewing Missed Calls menu without visiting details of each entry
- Support for concurrent multiple DTMF schemes
- Changed: We will bypass unregister and move forward to register if we received a non 2xx final response for the the (un)-REGISTER
- Added we will clear new missed calls display message after viewing Missed Calls menu without visiting details of each entry
- In web UI, the "if set to Yes, "#" will be function as the "(Re-)Dial" Key" is removed
- Add display "Preparing to write" and "This may take a while" between the period of firmware download complete and firmware flashing begins
- Disabled iLBC (iLBC is not working well and will be fixed later)
- Added support for XML encoded in UTF-16
- New dialing string display-old scheme will only show the last 11 digits dialed; new scheme will automatically scale down to a smaller font to display up to the last 42 dialed digits. Old scheme does not show line and account info while dialing digits, new scheme will.
- New centralized GUI control allowing showing multiple call information simultaneously including:
    - Still show account information while dialing (previous version only show dialed digits)
    - When 2 calls are present, we will split screen in half vertically (3 lines per call) and display both calls
    - When 3 calls are present, we will split screen in 3 sections vertically (2 lines per call)
    - When 4 or more calls are present, we will split screen in multiple sections (1 line per call) and display as many calls as we can. If there is an active call, that call will occupy 2 lines.
    - Select scenarios will occupy the entire screen for the active call (such as when TRANSFER key is pressed, or when CONFERENCE key is pressed)
    - When TRANSFER key is pressed you will still see the current calling info along with the prompt or the transfer target number you are dialing
    - When CONFERENCE key is pressed you will still see the current calling info along with the prompt
    - When call forward are requested (such as *73), you will still see the account information along with a prompt and the forward target number as you enter

Build 1.1.3.2 (03/15/2007)

Bug Fixes:

- Fixed attended transfer will fail (as transferee) if Contact header come after Refer header in the REFER request

New Features/Changes:

- Default DST rule changed from "4,1,7,2,0;10,-1,7,2,0;60" to "3,2,7,2,0:11,1,7,2,0;60" in compliance with the U.S. Federal Law passed in Aug 2005.

GXP-2000 Multilanguage New Strings:
Strings changed/Typos Fixed:

- The 403 message incorrectly spelled "alloweded", change to "allowed"
- In MENU-Preference: "Do NOT Disturb", changed to "Do Not Disturb"
- Copyright string at the bottom of HTML pages changed from (2004-2006) to (2004-2007)

Notes:

- The DST rule change above will ONLY occur after a factory reset
- If the GXP-2000 EXT does not start correctly immediately after the upgrade-completion bootup, please power cycle the GXP-2000

Bug Fixes:

- Fixed under VLAN mode, we send malformed ARP responses
- Fixed we crash when we do DHCP renew during a call under certain scenarios
- Fixed we increment UDP version for session-timer refresh reINVITE where session information did not change

New Features:

- Changed when we receive an reINVITE without SDP, default to sendrecv in offer (200/SDP) regardless of current RTP state
- Added support for changing local SIP port on registration failure
- Added configurable registration backoff interval, P138/471/571/671 in seconds (1-3600, default 20)
- Changed iLBC default payload type back to 97 (changed to 99 in 1.1.2.27); note that the actual payload type is not changed

GXP-2000 Multilanguage New Strings:
{655,255, "Enable Shared Call Apperance:"},
{656,255, "iLBC frame size: "},
{657,255, "20ms"},
{658,255, "30ms"},
{659,255, "iLBC payload type: "},
{660,255, "(between 96 and 127, default is 97)"},
{661,255, "SIP Registration Failure Retry Wait Time: "},
{662,255, "(in seconds. Between 1-3600, default is 20)"},

Notes:

- Support of new firmware format ("c" suffix)
- The "b" suffix files in the package are upgrade transitional files (1.1.2.99)

Build 1.1.2.27 1/30/2007

- Fixed iLBC default payload type incorrectly labeled as 97 (should be 99)
- Fixed we freeze if the domain string in WWW-Authenticate contains semicolon
- Fixed we incorrectly use To-tag in REGISTER
- Fixed GXP-2000 WEB UI EXT1 page translation incomplete (half of the page has "User ID" in English regardless of language pack)

- Fixed under VLAN mode, we send malformed ARP responses, shifted by 4 bytes

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- Fixed we do not untag incoming VLAN packets correctly
- Fixed iLBC frame size/payload type options missing in WEB UI
- Fixed DHCP options 2, 42, 66 not working
- Fixed DHCP option 66 does not handle path correctly
- Fixed NTP does not work when NTP server is in IP address form

Build 1.1.2.25 1/9/2007

- Fixed the "hissing" noise coming from the other parties handset
- Fixed VLAN not working
- Fixed display phonebook entry name as caller ID not working correctly
- Fixed We always use the firmware server in the HTTP host header

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- Fixed iLBC bad audio quality
- Fixed GXP-2000 incorrectly performed consultative transfer when you switch line during a blind transfer
- Fixed GXP-2000 results in one way audio when a second incoming call is not answered while the call is on hold
- Fixed we will not register any account if STUN is down or misconfigured
- Fixed if network is down-then-up STUN IP checking gets fired multiple series causing many STUN queries
- Restructured STUN/Registration to simplify account registration management
- Fixed echo in 3WC problem reported in 1.1.2.23
- Fixed GXP-2000 under 3WC, second call info not displayed correctly
- Added support for BT-200 onhook-threshold.
- Added customizable delayed call forward wait time. Provision parameter P139/P470/570/670, default is 20 seconds (as is previously), allowed value 1-120; invalid values ignored.
- Added support for BT-200 delete called/caller entries via MUTE/DEL key
- Changed NTP will retry 3 times if it receives no response from NTP server; after that it will retry it after 1 minute. This also fixed the NTP problem reported on the wiki site.
- Fixed we do not encode "#" in outgoing INVITE To URI

Build 1.1.2.23 12/15/2006

- Fixed a short buzz is heard when TRANSFER completes
- Fixed you can still enter GXP-2000 MENU when the phone is ringing

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- Optimize the speakerphone performance of the GXP2000 and BT200 to a 6 ft range. Implemented with a fixed AGC and a rough enhanced VAD based on a 5/16 frame buffering
- Added simple noise suppression for non speaking parties
- Fixed under Broadsoft mode we will not send INVITE if SIP Server/Proxy is in IP address form
- Implemented resuming call when CONFERENCE key is pressed again

- Implemented resuming call when TRANSFER key is pressed again
- Fixed GXP-2000 line key LED will become inaccessible
- Changed syslog or web UI status page for MAC address: separated by colons and in uppercase
- Added call establishment STUN queries to use event callback when response arrives, this reduces the brief delay when making and receiving calls when STUN is configured thus improving call experience

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- Added comments on WebUI for Account Name display support for BT-200
- Fixed GXP-2000 WebUI EXT1/EXT2 pages does now contain "eventlist BLF" as option, EXT2 page Key 65 display as "EXT Key 65: 65:", EXT2 page wording "UserID:" missing for Key 71-112
- Take Ring Tone out of Call Progress Tones section and added syntax description

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- Fixed some factory blank-LCD problem caused by GUI library not initialized correctly
- Redesigned mic and AGC/VAD changes
- Fixed handset/headset echo issues
- Fixed dial tone click and garbled dial tone in Broadvoice test
- Changed ringer volume gain from analog to digital scaling and increased max ringer volume
- Fixed G.723 on BT-200
- Fixed we allow HOLD to a call with early media
- Fixed under Broadsoft mode we will not register if SIP Server/Proxy is in IP address form
- Fixed we display SRTP error messages when 488 is received
- Added "P-Asserted-Identity" header for anonymous calls by Privacy header under Broadsoft mode
- Fixed BLF does not work with GXP-2000 EXT broken during eventlist implementation
- Fixed GXP-2000 renders bitmaps incorrectly when the encoded bmp string contains CRLF in it

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- Fixed volume cannot be adjusted for HW 0.4
- Fixed 3WC audio degrades when G723 6.3k is used
- Changed: we will NOT challenge reboot NOTIFY with 401 and accept it with 200 when SIP Authenticate ID is not configured.
- Fixed we do not use anonymous URI when making anonymous call using Privacy header as per RFC3325
- Fixed we sent RTP under MUTE in PCMU (regardless of the actual payload type in use) causing a short audible sound on remote end. An invalid RTP is sent instead which will be dropped but still keeping the NAT binding alive
- Fixed we do not sent RTP keep alive under HOLD
- Added tone analysis (disabled)

- Added support for save call history entries to phonebook
- Added support for displaying phonebook stored name instead of "From" header or "P-Asserted-ID"

- Added support for challenging Broadsoft remote-reboot NOTIFY (replies "401 Unauthorized" with WWW-Authenticate header).
- Fixed system ring tone does not play with accounts 2/3/4 when account 1 is set to use a ring tone that does not exist
- Fixed we do not use the same Authorization credential in ACK as in INVITE
- Fixed we do not display the number of messages (MWI) correctly when the number exceeds 99 (reported on Wiki). We now displays up to 999, messages over 999 will be displayed as 999.
- Changed behavior to "No Key Entry Timeout": imposed a minimum of 1 second and extended maximum to 30 seconds. Setting it to < 1 results to 1 and setting it > 30 will result in 30. Default is still 4 seconds. This is changed per frequent user requests (also reported on Wiki).
- Added under Broadsoft mode, register delay after 403 changed to 20 minutes
- Added support for reorder tone, played in lieu of busy tone when 403/480/484 is received for INVITE, Provisioning parameter P349
- Support G.726
- Fixed configuration download causes factory reset when EXT board is connected in certain scenario
- Changed behavior to "Automatic Upgrade": provisioning is delayed whenever a line is in-use, this include an offhook-idle line.

- Support BroadSoft Redudency.

- Fixed GXP-2000 HW1.1 cannot switch LED color in diagnostic mode
- Fixed iLBC will not work properly if switching from 20ms to 30ms without reboot
- Fixed 3WC with G.722 and G.711 bad audio
- Fixed digital volume gain management
- Fixed save volume audible by doing it in the background
- Added the function: In phone book->new entry, after input name and number, name and number can be displayed on “Add Phone Book Entry” menu
- Fixed the GUI MENU title incorrect in phonebook menu.
- Fixed name and number cannot be modified after an entry is added
- Fixed a potential DNS SRV priority handling bug
- Added under Broadsoft mode, keep-alive packets are sent to ALL DNS SRV resolved hosts
- Added DHCP option 61 (client identifier), Removed DHCP option 57 (maximum DHCP message size)
- Added support for Broadsoft Redundancy package, tested Broadsoft R14 test plan (113-123)

- Fixed BT-200 UI MENU option 7 (codec select) cannot choose G722/G726-32/iLBC
- Fixed HTTPd returns 404 when login as user (not admin) and saves any changes instead of displaying the page "Your configuration changes have been saved"

- Fixed BT-200 stops mute-indication when switching from speakerphone mode to handset mode and back to speakerphone mode
- Added "Hd" item to the BT-200 UI MENU option 8 (code rel) to indicate HW revision
- Fixed we do not follow the "a=fmtp:18 annexb=no" line in SDP (disable VAD), port from c64
- Fixed GXP-2000 phonebook GUI interface not initialized properly (empty title)

- Added support for G.722 (now interoperable with BT100)
- Added support for iLBC (interoperable with BT100)
- Added MENU UI codec selection support for G722/G726-32/iLBC
- Added support for customizable tones
- Provision parameter P343 (dial tone), P344 (MWI dial tone), P345 (ring tone), P346 (ringback tone), P347 (call-waiting tone), P348 (busy tone)
- Added default tone strings to WEB UI
- Added option to use remote contact in Refer-To solving the SPECIAL consultative transfer problem. Provision parameter P135/469/569/669, possible values 0/1, default 0.
- After factory reset, all accounts have codec selections in this order: PCMU/PCMA/G723.1/G729AB/G726-32/iLBC/G.722/GSM (consistent with other products)
- After factory reset, default NTP server is changed to "us.pool.ntp.org"
- Fixed DHCP sent DHCP discover from non-DHCP Client port (68) when network cable is disconnected and reconnected, introduced in 1.1.2.1 with TCP/IP Stack, this is also a memory leak case which has small impact
- Fixed you can't hear the remote party during CW tones playing
- Fixed we tear down the call without sending BYE when we receive 488 for reINVITE
- Fixed we do not add CRLF for DTMF by SIP INFO in the body
- Relaxed the new TCP/IP stack about UDP connection check to allow incoming UDP from hosts other than the one we are connected to
- Audio updates
- Fixed overflow bug in digital volume scaling function
- Added 9 more dB of volume scaling gain range
- Reworked on VLAN handling
- Forced RFC2833 DTMF ending duration to be 100 if it is 0 (for compatibility with GXW when GSM is used)

- Re-architecture the audio component
- Fixed upgrade via TFTP through GAPSLITE fails
- Fixed BT-200 crash on incoming call by turning callhistory module's optimization off
- Added BT-200 support for mute indication by flashing the speaker icon
- Support for Event Notification Extension for Resource Lists (eventlist, RFC 4662) Provision parameter P134/P444/P544/P644 for accounts 1/2/3/4 respectively, allowing each account to configure 1 eventlist URI. You will have to configure each individual BLF monitored userid in the Basic Settings page and select the key mode as "eventlist BLF" (versus "Asterisk BLF") so no individual subscriptions will be sent.

- Fixed custom ring tones will not ring
- Fixed we used "dialog:id" instead of "dialoginfo:entity" in dialog XML to identify the dialog
- Behavior change: DHCP option 66 (P145) is default to 1 on factory reset

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- Fixed GXP-2000 crashes on bootup after factory reset due to illegal freeing of some uninitialized localization strings.
- Fixed we updates Record-Route set by in dialog responses
- 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 option to allow turn off display of in-call DTMF digits
- Provision parameter P338, default value 0 (digits displayed), possible value 0/1.

- Expanded syslog message to include DTMF type and ptime info in addition to payload type selected before session start
- Fixed TCP buffer-chaining bug which may be the cause of the HTTP download failure
- Support of DNS SRV query for TCP type (automatic--query for _sip._tcp.sip_proxy is sent when account is configured for TCP)
- Fixed we sent in-dialog requests to the proxy that we registered to instead of the actual proxy the dialog established
- Support Bellcore-drx (x=1-5) ring tones, supported as a general feature
- Support for BT-200 to disable call logs. Provision parameter P187 (for BT-200 only), possible values 0/1, default value 0 (calls logged as normal)
- Fixed config_e1/config_e2 page header not aligned
- Fixed GXP-2000 stalled when config_e2.htm is accessed
- Fixed GXP-2000 allows config_e1.htm access even when login using end-user password (123)
- Handset audio fixes

- Added bootloader version to HTTP header
- Reworked the SIP NOTIFY/events module to have a clean NOTIFY handler
- Added support for anonymous call rejection (per account); Provision parameter P129 (P446/546/646 for accounts 2-4 on GXP-2000), default is No
- Added support for Broadsoft remote-reboot via NOTIFY (check-sync event)
- Added support for remote reboot via NOTIFY for accounts 2/3/4 for GXP-2000
- Fixed we send 415 response for NOTIFY when no Content-Type header is present
- Fixed BT-200 fails to download configuration file (does not impact GXP-2000)
- Fixed GXP-2000 crashes when DHCP fails and link is down. Fixed when factory reset, Account 1 SIP Transport has no default value (should be UDP)
- Fixed the URI in auth header is trimmed for SUBSCRIBE
- Fixed GXP-2000 stay in speakermode after loopback call
- Fixed we will first send REGISTER to 0.0.0.0 when DNS SRV is in use for SIP server
- Fixed a bug in SIP stack which caused some mysterious crashes

- Officially named the language pack file to gxp2000.lpf. As the name suggests, this file is model specific (gxv3000.lpf, bt200.lpf … etc. in the future).

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