Release Note for Firmware Version 1.1.0.2

Mar. 23, 2007

Note: This release includes a new boot loader program. Once upgraded to 1.1.0.2, the firmware can NOT be downgraded to any previous releases.

1.1.0.2

- Fixed no reorder tone when SIP INVITE hits 484 but early-dial is not configured
- Fixed lose registration after PPPoE renewal with an new IP
- Fixed device freezes if the domain string in WWW-Authenticate contains semicolon
- Fixed Attended Transfer will fail (as transferee) if Contact header come after Refer header in the REFER request
- Changed when we receive an re-INVITE without SDP, default to sendrecv in offer (200/SDP) regardless of current RTP state
- Syslog changes as well as some driver changes for polarity reversal

HandyTone specific (HT-286 Rev. 3/HT486 Rev. 2):

- Fixed 2s fxs turn-around after PSTN pass-through call
- Fixed fax pass-through doesn't work well under LBR
- Fixed DTMF range of 0-11 is offered in SDP instead of 0-15
- Fixed hear no vice after attempt to blind transfer by using *87 + no number

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- Fixed VLAN does not work with DHCP
- Fixed a SIP Contact header parser bug
- New boot.bin (1.1.0.1) with WDT support

- Added conference moderator can drop 3rd party via hook flash
- Added support for PPPoE service name
- Added support for configurable powerline ring tone frequency and cadences
- Added we offer both PCMU/PCMA in pass-through mode fax reINVITE
- Fixed if the second party hang up in 3WC, bad audio is heard
- Fixed conference moderator sending INVITE when pressing key after 1 party leave 3WC

- Changed when no CallerID is available, we display Private instead of IP address
- Fixed HT487 *00 works as PSTN access code when not configured
- Fixed we do not encode "#" in outgoing INVITE To URI

NOTE:

1.0.8.33 firmware has major changes compare with previous firmware (1.0.6.x or 1.0.7.x), so downloading time may take up to 8 minutes.

REMOVE ALL the files in the TFTP/HTTP server, including the configuration files. (Or use different directory for Firmware versus configuration/provisioning file)

Make sure all the files that come with Release_ BT100-HT486-HT286_1.0.8.33.zip are unzipped into the TFTP or HTTP server.

1.0.8.33 10/18/2006

- Fixed we send BYE before final NOTIFY arrives as transferor
- Fixed we keep sending CANCEL when 487 for INVITE arrives before 200 for CANCEL
- Fixed we send 481 for early INFO

1.0.8.32 9/18/2006

- New T.38 code to fix some FAX failures
- Moved "PSTN Access Code" under BASIC SETTINGS page
- Moved “WAN Side HTTP Access” under BASIC SETTINGS page
- Moved “Reply to ICMP on WAN Port” under BASIC SETTINGS page

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- Fixed provision/sip_registration timing issue causing registration to send to wrong address when connected to cable modem directly
- Fixed router models responds ICMP unreachable with ICMP unreachable and uses incorrect target MAC address
- Fixed WEB UI label on Call Progress Tones for HT287/487
- Fixed SIP stack incorrectly parsed "CT" header
- Changed SIP stack implementation for From/To headers from static to heap, increased heap size
- Tears down a call when 404 is received for in-dialog reINVITE and drops incoming RTP frames that does not come from matching SDP host under CBCOM mode to temporarily go around the platform's misbehavior
- Sends 400 BAD REQUEST for incoming CANCEL requests that are not matched
- Decoupled Call-Waiting from Call Features

- Disable interrupt during the LEC initialization, pertaining to the CBCOM crashing issue
- Implements Provider Lock (P9997/P9998/P9999) feature
- Fixed BT100 "Basic Settings" page cannot update

- Fixed we will perform firmware upgrade even if configured not to when DNS query for config server (0.0.0.0) failed
- Fixed "Allow outgoing call without Registration" fails
- Fixed Bug in Via header, DNS form instead of IP is used in the header
- Fixed BT-100 does not clear out screen after reminder-ring times out, and allows indefinite onhook-after-call-hold
- Improved router performance for HT-487
- Disable LEC when fax is detected
- Disable G.723 VAD to solve crash problem in silent mode
- Fixed IVR bug in 1.0.8.29
- Removed Lucent FS5000 SE Mode

1.0.8.29 7/21/2006

- Fixed G.723+VAD crash problem
- Fixed HT487 port-forwarding failed problem
- Fixed Broken HTML page issue
- Fixed in Switch/Bridge Mode will NOT pass PPPoE packets

- Fixed BT-110/120 ring3 will not be played
- Fixed BT-110 web UI access will crash
- Fixed BT/HT rings even the held party hangs up

- Support for Call-progress tones on HT-287/487[1]
- New voice prompts for HT-287/487

- Fixed we do not play busy tone when 487 is received

- Fixed BT-110/120 CallerID ring tone is not correct if number is shorter then previous caller of the entry
- Fixed we do not follow the maddr attribute in the 301/302 for REGISTER
- Fixed HT-287/487 you cannot use VP options 01-05 when menu is locked
- Fixed BT-110/120 ends a call when pressing SPEAKER button during speakerphone mode even if handset if offhook

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- Fixed we still play dial tone when WAN cable is pulled out

[1] HT287 is HandyTone 286 Rev 2.0 and HT487 is HandyTone 486 Rev 2.0

- Fixed we crash on attended transfer on platforms that use To/From headers without square brackets
- Fixed lost of registrations when multiple bindings exists
- Support Home NPA on HT-287
- Support reject ringing call by DEL key on BT-110/120
- Support configurable time to reject call if not answered
  Option is called "Time to ring" on the WEB UI, provision variable P185, valid values 30/60/90/120, invalid values ignored.
- Support for multiple DTMF schemes
- Support for HT-287/487 gain control
- Support for auto daylight saving feature
- Support for configurable WEB UI port
- Support for DHCP options 12, 15, and 60 (DHCP hostname, domain, and Vendor ID)

1.0.8.23

Build 1.0.8.23 5/30/2006

- Fixed a crashing issue when processing fragmented large packets
- Fixed no keep-alive packet when detected NAT type is symmetric NAT
- Added the configuration option to disable firmware upgrade
- Disabled LEC when Fax communication is detected
- Temporary disabled VAD with G.723

- Fixed typo in VLAN handling that would break VLAN TCP
- Added the support to enter TFTP address for both firmware and config server via keypad or voice prompt
- Improved the TCP routing performance
- Fixed the 3-way conferencing issue when the re-INVITE to bring the first party out of hold status was challenged

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- Fixed HT-487 cannot PING from WAN side even when configured to allow it (the problem only happens after reboot--when you change it from No to Yes it works until it gets reboot)
- Fixed we still send RTP (a=sendrecv) even when we are been placed on hold when used with Spherecom proxy
- Fixed RTPSend bug
- Fixed early-dial and IVR problem
- Fixed PPPoE issue

- Fixed HT (N/A for BT) voice prompt problem introduced in 1.0.8.18
- Fixed we will retry 5 times if only config server is configured and there is config file
- Fixed an internal timer error which occasionally cause unit not sending re-registration when network is down.

- Fixed we do not clear out Call Forward settings when user configure to disable call features
- Added HT (N/A for BT) Bell-style 3WC option (P108, possible values 0/1, default 0)

- Fixed VLAN problem
- Fixed ARP problem
- Fixed web UI still display as REGISTERED even though we lost IP address (PPPoE)
- Added Timer protection. This eliminate some lost registration issue in certain network down scenario
- Changed registration back-off interval from 1 hour to 20 minutes
- Added option to allow out-going calls (dial tone) even when not registered (Configuration Parameter P109, allowed value 0/1, default 0)
- Fixed we do not handle 423 correctly
- Extend the wait time between receiving RTCP BYE to terminating call via SIP from 500ms to 5 seconds

1.0.8.16

Build 1.0.8.16 2/10/2006

- Added we will attempt to start the INITIAL(booting up) 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)
- Fixed Brazil and Denmark CID schemes not working
- Fixed we still do attended transfer and conference with call-features are disabled
- Fixed we stops playing dial tone when a hold call is disconnected remotely (A call B, A put B on hold by using FLASH, and B hang up before A dial any digits)
- Added model number and firmware version (app) in every syslog message
- Fixed we do not process the IP packets if the first fragment did not arrive first

1.0.8.12

Build 1.0.8.12 1/23/2005

- Fixed if config server failed (no ARP response in local network for both HTTP/TFTP or no TCP response to SYN if HTTP) we do not proceed to firmware server
- Fixed syslog does not log second digit of the error code correctly
- Shortened the wait-time between downloading files from 1 second to 100ms
- Fixed TCP/HTTP download problems
- Added specific Syslog messages during download to differentiate different scenarios
- Added protection so we will reject older bootloader (any firmware older than 1.0.7.11 including 1.0.7.11)
- Added under CBCOM mode, the User-Agent field will be appended with " en" at the end so the SIP server will know that the mode is active
- Added DTMF-Brazil caller ID scheme
- Fixed we do not send CANCEL to end the call when onhook during ring-back stage

- Fixed you cannot pick up an call-waiting incoming call if you first put the phone on-hook to end the original conversation
- Added Onhook-Threshold option. New parameter P245: valid options are 2/4/6/8/10/12, actual interval is x100 ms, default is 800ms. Invalid entries ignored.
- Fixed we cannot handle the /1 after the port number in the SDP m-line (G729r8 problem reported)
- Changed we accept both Message-Waiting and Messages-Waiting in MWI NOTIFY (415 problem reported)
- Added we will do TFTP/HTTP provision upon DHCP/PPPoE completion if no IP address was available initially
- Fixed BT100 when on hook on a call placed on hold using the HOLD button is disconnected which should remain on hold
- 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 will drop NTP response if the Leap Indicator is non-zero
- Fixed BT100 plays ringback tone occasionally when switching between calls using hook-flash (using FLASH button will not have problem)
- Added RNK mode
- Fixed under static IP mode we register to the wrong IP address first under certain scenario

1.0.7.18

Build 1.0.7.18 11/18/2005

- Fixed DHCPd fails to work if a DHCP client's MAC address is same as the cloned MAC address.
- Fixed DHCPd always send DHCP NAK when client attempt to renew.
- Fixed DHCP fails to work after receiving DHCP NAK.
- Reduce the DHCP polling interval to 30 seconds from 1 minute (this is the effective minimum DHCP renewal time)
- Fixed we do not attempt firmware download if config server in FQDN form and DNS failed (or DNS server not configured), impacts HT-487 and BT100.
- Fixed restriction from web UI: if LSB of first octet of cloned MAC address is 1, it is not accepted and default MAC address will apply
- Added the cloned MAC address will only kick in when it is in router mode (in bridge mode we will not do)
- Added RNK mode for BT100 (plays local ring back on 18x with SDP)
  Design: usually when we receive 18x with SDP, we will start RTP, but when configured as RNK, the incoming RTP frames are dropped and a local ring back tone is played.
- Changed: under SONUS and LEVEL3 modes Call-ID for registration stays the same through out power-up session.
- Added Anonymous Method to allow configuring to use Privacy header (P268 = 0 for From_Header or 1 for Privacy_Header)

- Added restriction from web UI: if LSB of first octet of cloned MAC address is 1, it is not accepted and default MAC address will apply.
- Added 30 second auto cut-off after on-hook "call-on-hold reminder ring" per AOL's requirement.
- Fixed G723_SID typo.
- Fixed use SIP server URL instead of IP in Referred-By header (Fixed Nortel 3WC)
- Added we offer DHCP lease term of 60 seconds when there is no DNS server (simulate this by not plugging in WAN port of HT-487).
- Fixed we always use first codec offered in ACK's SDP when given.
- Fixed we do not follow the "sendrecv" attribute in ACK's SDP.
- Added CBCOM mode.
- Fixed BT does not take certain ring tone file (ported from GXP-2000)
- Changed- we will not start sending T.38 media until we receive the ACK for the 200 OK send
- Fixed early dial does not work.

1.0.7.11

- Fixed long UDP vulnerability issue, can be verified with a customer provided Perl script
- Added override MTU option
- Added we ARP for SIP server/proxy if it is in same LAN prior to registration.
- Fixed we always use the firmware server's host name in the "Host:" header in HTTP GET when downloading configuration file via HTTP (applies to BT100/HT-487 only).
- Added TTY support (HT-487 ONLY)
- 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).
- Added new time zone Newfoundland GMT-3:30
- Added Syslog for registered message to indicate the actual registered time and the re-registration time
- Added syslog message for failed configuration file authentication
- Fixed BT100 first codec changed from MENU UI not updated until reboot
- Fixed if caller ID block (permanent) is selected we still make calls without hiding caller ID on second call of 3WC
- Fixed AOL dial plan not kick in for second call
- Fixed we always send REGISTER to SIP server although outbound proxy is configured when both are in IP address form
- Add TA to give second dial tone for *72, *66, *70 and *67, when configured as Sonus ASX or Level3 mode and enable call features is set to No
- Add allow configure to accept SIP from proxy only (P243, default is No). This can be tested using direct IP call

- Add allow configure to reply to ICMP on WAN for NAT models (P189, default is No--this means it does not reply to PING on WAN port)
- Changed we do not start RTP when restoring a call placed on hold until we received the 200 OK for reINVITE.
- Changed auto upgrade interval to minutes on BT100 for consistency with HT-487
- Added support for allowing SIP account cfg in basic setting page (configurable via P241)
- Ported authenticating config file to HT-286, HT-487, and BT-100
- All web UI synchronized- Login page auto focus on password field, Save_ok page shows REBOOT button and other adjustments, Register Interval in seconds, … etc.
- Added Level3 Mode:
  -Level3 style 3WC
  Added Configuration version in status page
- Fixed unreliable CWCID
- Fixed we do not CANCEL a call if FLASH is used before a call is established
- Fixed no busy tone played when remote disconnect and there is a call on hold (A talk to B and have C on hold, B hang up and A does not play busy tone, BT only)
- Change Web UI display name from "HT487" to "HT486 REV 2.0" (all other places stay the same), also adjusted some drop down items in Advanced Page.
- Change the upgrade period unit from day to minute. Minimum is 60 minutes.
- Added 3 Way Conference for HT286/HT486(Rev2.0)/BT100
- For BT100 and HT486 Rev 2.0 only
  - Enable TFTP server with FQDN and merge with HTTP server.
  - Allow Firmware Upgrade Server and Configuration Server to be configured from different server.
  - Allow Prefix and Postfix to Firmware or Configuration file, this allow different firmware version and different configuration file(for the same device MAC) in the same file directory.
  - Allow firmware to be accepted only when the prefix and postfix matches what's defined in the downloaded configuration file.
  - Allow firmware to be encrypted so that only the encrypted firmware with right AES key to be accepted.
  - Add logic parameter P88 so that when it is set to 1, it will prevent user from resetting the client to factory default.
  - Added support for allowing SIP account configuration info in basic setting page (configurable via P241), they includes SIP User ID, Authentication ID and Authenticate Password and Display Name.
  - Added Configuration File Authentication (Configurable via P240).
  - Add configuration file version number in status page (When configured in Level3 Mode)
  - Add Configuration file version number, MAC address, and IP address of the client into SIP Registration message(When Configured in Level3 Mode)
- Fixed we do not CANCEL a call if FLASH is used before a call is established
- Fixed NTP does not work when NTP server is in local subnet

- Fixed DNS SRV entries not sorted correctly
- Fixed if SIP/NTP/STUN/Syslog/Upgrade server all configured in FQDN (not IP address), registration may fail
- Fixed we *unnecessarily* encode certain non-reserved characters in the Replaces parameter of Refer-To as Nortel DMS-10 team reported.
- Changed Web UI Register Expiration from minutes to seconds, HTTP auto-upgrade from days to minutes
- Fixed we included the Replaces parameter portion in the request line and To header of the INVITE as transferee in an attended transfer
- Fixed we cannot handle Call-ID over 80 bytes long correctly (Alcatel's report)
- HT486 adjusted Wed UI, removed polarity reversal setting
- Added capability to handle multiple 18x with SDP redirection (for Nortel DMS10)
- Fixed comma-in-"Contact" bug reported
- Added support for Authenticating configuration file before accepting changes
- Fixed we always send REGISTER to SIP server although outbound proxy is configured when both are in IP address form
- Add allow configure to accept SIP from proxy only (P243, default is No). This can be tested using direct IP call
- Add allow configure to reply to ICMP on WAN (P189, default is No-- this means it does not reply to PING on WAN port)
- Changed we do not start RTP when restoring a call placed on hold until we received the 200 OK for reINVITE.

1.0.6.7

- ***This release includes a new boot loader program. Customer is not encouraged to downgrade to the older firmware. Customer is recommended to contact support should they need to downgrade to the older firmware.***

- Fix Refer-By header issue
- Fix the issue where TFTP server is wiped out if upgrade from below 1.0.5.21 using TFTP with config file
- Fix crash issue when changing static IP to DHCP and reboot through Web UI
- Fix ill-formatted contact header in 302 response
- Fix the issue where INVITE after 302 response bypasses outbound proxy
- Fix device lock-up issue when multiple RTCP packets are embedded in a single UDP
- Fix the issue with MD5-sess authentication
- Add the Web configuration for time zone on HT 486 Rev 1.0
- Improve the reliability of reading from Flash
- Fix the PSTN ringing issue with HT486 Rev 2.0 in Demark
- Fix the issue where interruption table may get corrupted under certain conditions
- **Fix the no dial tone issue with certain European analog phones when on-hook voltage is set other than 18V**
- **Fix the European PSTN ringing issue with HT486 Rev 1.0 and 2.0**

- **Fix the issue when Grandstream phone does not upgrade ARP table on gratuitous ARP**
- **Fix the header parsing issue when expires header appears before contact header**
- **Fix the issue where Voice Frames Per TX can't be updated**
- **Fix the frequent registration issue due to Ethernet link loss**
- **Fix the SDP negotiation issue when there is multiple m= lines with both audio and T.3 8**
- **Add work around for lost-registration issue with VOCAL**
- Add the support for bridging mode on HT486 rev. 2.0
- Change the default iLBC payload type to 97
- Change the jitter buffer delay to 2 for G.723 and 3 for other codecs
- Add the support for VLAN coloring for VoIP and PC on HT486 Rev2.0
- Fix the hook flash problem with LBR codecs
- Fix the issue where internal web server is not accessible from LAN port when HT486 rev1 or rev2 is reset and WAN link is down
- Fix the issue where branch ID is not upgrade during registration
- Fix the issue where replaces header cannot match an early dialog
- Internal web server now returns "access denied" rather than a blank error page
- Support T.38 real-time fax on HT-286, HT-486 Rev 1 and HT-486 Rev 2. Fax Pass-Through mode under PCMU/PCMA is still supported
- Support Syslog on BT101/102, HT-286 and HT486 Rev 2
- Fix the FXS port frozen issue under certain circumstances
- Add call statistics on status web page on BT101/102, HT-286 and HT486 Rev2
- Enable caller ID scheme and call waiting caller ID support on HT-486 Rev 1
- The device now returns 481 instead of 200 if SIP INFO is received on a unknown call
- Add the support for Nortel x-nt-GUID header
- Add the support for P-Asserted-Identity header
- Add the support for DTMF tone A-D

Release 1.0.5.23 3/10/2005

- Fix the problem (from code merge) causing tftp of 0 for HT486 rev 1.0 under factory reset or HTTP upgrade circumstances
- Fix the issue related to syslog implementation (use of broadcast address and a typo)
- Do not tear down a call immediately upon hanging up but instead wait for 3 seconds before clearing a call
- Fix the stack overflow and memory leak issue that cause devices to lock up and crash
- Fix the file download corruption issue related to PPPoE
- Fix the issue where PPPoE-Relay-Session-Id is not sent under certain conditions
- Fix the TCP MTU negotiation issue
- Fix the Layer 2 QoS issue associated with ARP packet
- Add support for Canadian Call Waiting Caller ID (HT286/487 only)
- Fix invalid RTP SSRC after SIP RE-INVITE
- Fix the issues related to Record-Route

- Fix the issue where the phone will continue to ring after a BYE instead of CANCEL is received
- Fix the issue where branch ID is not upgraded for new SIP INFO request
- Fix the issue where HT486 may send REGISTER request to its WAN port under certain conditions
- Change the default On-Hook voltage from 24V to 36V on HT286 and HT487
- Fix the issue where callee does not send DTMF when configured as SIP INFO or RFC2833 (BUG ID 4)
- Fix the caller ID/call waiting caller ID issue with GE phones
- Fix the issue where iLBC always uses 30ms mode

HTML 1.0.0.47 3/10/2005

- Add support for Canadian Caller ID and Call Waiting Caller ID (HT487 and HT286 only)
- Update the copyright information
- Fix broken HTML when the device is busy

ISSUES NOT FIXED

- Slow PPPoE connection speed reported in France
- VLAN connectivity with certain managed switches

Release 1.0.5.22 1/21/2005

- Changed polarity reversal logic per customer request and fixed the polarity reversal issue
- Add support for syslog server (HT286 only)
- Change the choice between tftp upgrade and http upgrade to mutual exclusive
- Fixed we cannot dial SIP call without using # key in HT486 Rev 2.0 when PSTN access code is non-default.
- Fixed we do not use RFC2833 to send DTMF when the incoming SDP contains iLBC and the immediate next "a" line is not the fmtp line for iLBC
- Fixed we use G.722/8000 instead of G.722/16000 (BT100 only). (Fixed compatibility problem with some other vendor)
- Fixed: we will retry http download if we received TCP RST.
- Send DHCP NAK if a DHCP request is not within our memory record (to force a restart of DHCP Discovery process).
- Fix the issue related to Record Route under some situations
- Fix the layer-3 TOS issue
- Fix the port forwarding issue

Release 1.0.5.19 12/3/2004

- Do not display SIP authentication password on the HTML page
- Release DHCP or PPPoE connection before rebooting
- Re-enable fax tone detection upon which switching to PCM if previously using low bit rate codec
- Add configuration support for European Caller ID (286 & 486 Rev 2.0 only)
- Add system up time in the "status" page of the Web interface
- Allow 96 to be used for DTMF payload type
- Fix the DHCP length issue which causes our DHCP offer to be rejected by Linksys router when it is on the LAN side of 486/487
- Add support for call waiting caller ID (HT286 and HT486 rev 2.0 only)
- Add support for polarity reversal configuration parameter (HT286 and HT486 rev 2.0 only)
- Fix the request URI and Route Set bug
- Use "terminated" as Subscription-State for termination NOTIFY as transferee
- Add register state information on the "Status" web page.

- Release 1.0.5.18 11/14/2004
- Change the factory default setting of "enable call feature" to TRUE; change the factory default setting of "NAT Traversal" to "Yes"; change the factory default setting of "Inter-key timeout" value to 4 seconds; change the factory default setting of "upgrade checking frequency" to 7 days.
- Fix the issue (resulting from recent code change) that causes bad IP header checksum in HTTP Upgrade packet
- Fix the issue that if no response is received due to bad HTTP GET packet or packet drops, HTTP upgrade retry will not happen and SIP registration will not happen or take very long time to happen
- Support attended call transfer and server-side 3-way conferencing for Nortel
- Fixed send NTP to STUN server when STUN server is in FQDN form.
- Fixed dialing bad URI when offhook auto dial is enabled.
- Fixed BT-100 dialing bad URI when using the message button.
- Fixed BT-100 show only first caller in caller-history.
- Allow lower case encoding in Replaces.
- Change the wording of "do not disturb" to "disable call waiting" in Web interface
- **Support 3-page Web configuration interface**
- Add configuration parameter to support **special feature** for Nortel's MCS, Broadsoft (except HT486 rev 1.0)
- Change the target MAC address from ff.ff.ff.ff.ff.ff to 00.00.00.00.00.00 in ARP request packet

- Always Unregister (not all contacts but only the binding that it registered as) and re-register when "UPDATE" is pressed in Web configuration interface. (HT486 2.0 only)
- Fixed transferee stops playing ring-back tone if transferor hang up before transfer target answers on blind-transfer.
- Fixed answering ARP for IP address of the wrong port (eg. we answer to ARP for 192.168.2.1 even though the ARP comes from WAN port).
- ALWAYS set the http upgrade URL to: fm.grandstream.com/gs and enable firmware upgrade (YES) upon reset to factory default.
- Add support for 501 not implemented response
- Fix the problem where in the Web user interface, pressing the UPDATE button will not get response if no parameter is changed
- Fix the issue of exposed password on HTML—we will not display password in the Web interface and will not take empty password.

- **HTML 1.0.0.42 11/11/2004**
- Use new graphic user interface with 3 different tabs (status, basic settings and advanced settings)
- add configuration parameter to support special feature for Nortel MCS, Howdy, etc. (HandyTone 486 Rev 2.0 only)
- do not display password on HTML pages

- **Key bug fixes and enhancements since Release 1.0.5.11**

- **Release 1.0.5.16 10/18/2004**
- Improved routing performance for HTTP traffic
- Support enable/disable of caller ID, and call waiting via keypad
- Fix the issue related to processing encrypted configuration file
- Fix the issue causing 400 bad response to be sent for NOTIFY after blind transfer
- Support Fragmented UDP frames for SIP processing
- Fix the missing Contact field for SUBSCRIBE and INFO request
- Add support for upgrading firmware or modifying configuration via http. Support file path for http url.
- Add logic to detect and decline duplicate IP during DHCP application stage.
- Add call time ticking display for callee (BudgeTone 100 only)
- Support file content authentication checking using AES during firmware upgrade
- Support for release of IP upon detecting the link is down for more than 15 seconds and re-application for IP address as soon as the link is up again
- Support attended transfer and Replace header
- Support Proxy-Require header and its configurable content

- Support pre-scheduled firmware upgrade checking frequency and add control flag to allow or prohibit auto firmware upgrade.
- Support configurable PSTN access key string
- Support 2 different Web login screens (1 for end user and the other for admin). The login interface is shared between 2 different user modes but the edit screen is different. Add port forwarding, DMZ and DHCP server related configuration options to end user configuration screen
- Fix the loss of registration issue
- Fix the issue that a HOLD initiated by 1 party can be released by the other when the other party presses HOLD and then releases the HOLD.
- Fixed the extra "@" character in "From" header when user ID is blank.
- Fix the issue related to negotiating and using the right MTU when remote end uses a smaller MTU (HT486 only)
- Fix the PPPoE link state monitoring issue if CHAP is used.
- Fix the issue where our RTP sequence ID is randomly changed when a 183 response is initially received and then a 200 OK response is received.
- Fixed layer 2 QoS (VLAN and 802.1p) issue
- Maintain the credential information for all subsequent REGISTER after the initial registration is successful, as opposed to restart challenge-authenticate cycle for each new REGISTER transaction
- Fix the "reset to factory default" which is recently broken
- Increase the timeout value for PPPoE call establishment. This will better accommodate some Chinese DSL modems' slow response. Also reset IP upon detecting the pppoe link is down for more than 15 seconds.
- Fix the issue where improperly deleting an un-initialized timer can cause timer malfunction
- Fix the issue that PPP PAP timer interferes with CHAP negotiation
- Fix the issue related to processing multiple IP addresses of DNS A record response
- Fix the issue that PCMU is always included in SDP even if it is never configured on HandyTone products
- Fix a bug to better handle very long Contact header, e.g., 500+ characters long
- Fix the ptime negotiation issue where we didn't use the default ptime when the remote end responds with a codec that is different from our first offered codec and which has no ptime in its SDP
- Fix the issue that after firmware upgrade the device should (but previously does not) reboot automatically.