Release Note

Firmware Version 1.0.3.64

Dec. 4, 2006

1.0.3.64

- Added support for "Allow incoming SIP messages from SIP Proxy only"
- Added logic to prevent user from accessing Web UI during firmware upgrade
- Added support for RED LED flashing every 0.5s if program image doesn't match HTML image version

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- Added option to use remote contact in Refer-To header for Attended Transfer
- Fixed Denmark Caller ID issue
- Added DTMF suppression for FXS port
- Fixed polarity reversal issue for CBCOM

- Changed default NTP server to us.pool.ntp.org on factory reset
- Fixed we incorrectly reverse the record-route set in certain scenarios
- Fixed no CRLF in SIP INFO packet

- Fixed anonymous calls Caller ID display issue with certain SIP servers
- Fixed we send BYE before final NOTIFY arrives as transferor
- Fixed we do not CANCEL a call if FLASH is used before a call is established
- Moved "WAN Side HTTP access" and "Reply to ICMP on WAN port" to BASIC SETTINGS configuration page
- Fixed Call-on-hold reminder ring lasts indefinitely

Note: Once upgraded to 1.0.3.57, the firmware can NOT be downgraded to any previous releases.

1.0.3.57:

- New boot.bin 1.1.0.1
- New vocht.bin 1.0.0.13
- Fixed we send 481 for early INFO

- Improved web UI to show 2nd account SIP registration status, added new page for Call Progress Tones or Ring Tones
- Added support for 3-way conference moderator press flash to drop 3rd party
- Improved HT488 dialing out to PSTN reliability via RFC2833 or SIP Info

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- New T.38 code to fix some FAX failures
- Implements Provider Lock (P9997/P9998/P9999) feature
- Fixed System Uptime begins with 1 hour if DST is enabled

- Added support for DTMF suppression
- Fixed Attended Transfer issue between FXS1 and FXS2
- Decoupled Call-Waiting from Call Features
- Fixed SIP stack incorrectly parsed "CT" header
- Disable interrupt during the LEC initialization, pertaining to the CBCOM crashing issue

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- Fixed Bug in Via header, DNS form instead of IP is used in the header
- Fixed we will perform firmware upgrade even if configured not to when DNS query for config server (0.0.0.0) failed
- Fixed “Send Anonymous” issue on HT488 FXO port
- Disable interrupt during the LEC initialization, pertaining to the CBCOM crashing issue
- 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
- Changed SIP stack implementation for From/To headers from static to heap, increased heap size
- New bootloader 1.0.8.11

1.0.3.48:

- Changed registration back-off interval from 1 hour to 20 minutes
- Added option to allow out-going calls (dial tone) even when not registered

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- Removed MWI field from HT488 FXO port
- Added “Always Skip the Firmware Check” option for firmware upgrade

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- Fixed HT496/488 router port forwarding issue
- Improved HT496/488 router performance

- Fixed HT386/488/496 rings even the held party hangs up
- Fixed un-reliable polarity reversal on FXS/FXO ports
- Fixed HT496 bridge mode not passing PPPoE packets
- Fixed broken HTML when clicking UPDATE after timeout
- Increased output current to support certain cordless phones
- Added new voice prompts
- Added support for configuration file authentication

1.0.3.44:

NOTE: Starting from firmware Release1.0.3.44, HT386/488/496 has a configurable web user interface for ring tones. You don't have to change anything in order to use the firmware. All ring tone default values should be compatible to what previous firmware offers. If you hear unfamilar ring tones after you change web configuration, you may change it back to default values. If you forget default values, a factory reset is the only way to gain back default values.

- Fixed attend transfer results in Busy tone and one way audio
- Fixed removed default account names and empty quotes in all cases
- Fix multi- contact list causes lose Registed
- Fix no busy tone played after receiving BYE when unit has a another party on hold
- Corrected SIPP2 Concurrent Call no audio issue when forced by callee to use 2 instance of LBR
- Make onhook threthhold to be configurable
- Fix cannot pickup 488 FXS phone at the last ring for PSTN incoming calls
- Fix no keep alive packets when it detects Symmetric NAT

- Fixed HT488 FXO call gets cut off issue
- Fixed HT488 PSTN Disconnection issue
- Fixed HT488 PSTN Silence Timeout reliability issue
- Fixed *00 echo issue
- Fixed cannot pick up HT488 FXS phone at the last ring for PSTN incoming calls
- Fixed no keep alive packets sent when it detects Symmetric NAT
- Fixed HT488 FXO off-hook only when the VoIP callee phone off-hook for "Forward to VoIP" features in PSTN to VoIP call flow.
- Fixed HT488 FXO port # and * DTMF digit detection
- Fixed device reboots immediately even if the device is on a call upon GAPS reboot command
- Fixed WAN side HTTP Access issue under PPPoE
- Fixed 3-way conferencing fails when the re-invite to bring the 1st party out of hold status gets challenged
- Fixed cannot start 3 way conference if it is redirected by 3xx response
- Fixed FXS2 port (HT386/HT496) cannot initiate a 3 way conference if its call features is set to no on FXS1 account
- Fixed 488 call switch issue if FXO profile is not configured for PSTN to VOIP call
- Fixed FXS1 jump call from VOIP to life line if FXS2 off-hook and dial *00
- Fixed cannot re-connect to PPPoE server if ADSL modem supports DHCP server
- Fixed sending ARPs under some network condition
- Fixed VLAN problem with Cisco switch

- Fixed HT488 Route to PSTN issue in Italy
- Fixed early dial breaks *code IVR and PSTN access
- Fixed SIP 423 register too brief
- Fixed early dial feature is interrupted by keypad timeout of 4 sec
- Fixed we still do Attended Transfer and conference when call-features are disabled.
- Fixed early dial issue on port 2
- Fixed Lock Keypad Update to prevent IVR changing advanced settings
- Fixed we do not process the IP packets if the first fragment did not arrive first (Lucent issue)
- Fixed shared off-hook MW tone for both ports even if voice mail is only for one account
- Fixed if configure 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
- Shortened the wait-time between downloading files from 1 second to 100ms

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- Enhanced HT496 and HT488 router performance
- Enhanced system registration reliability
- Enhanced IVR to set configuration server
- Added support for configurable HTTP server port
- Added support for configurable Onhook Threshold
- Added support for 491 SIP Request Pending
- Added support for DHCP option Host Name (option 12), Domain Name (option 15), and Vender Class ID (option 60).
- Added multiple DTMF methods support, changes from mutual exclusive to combo of RFC 2833 and SIP INFO
- Added support for dual LEC (G.168) for HT496 and HT386.
- Added support for Bellcore style 3-way conference
- Added support for voice volume control
- Added configurable web UI to enable/disable WAN side ICMP response
- Added support for seperate CSeq for port 1 and port 2 on SIP registration so that each account use independent CSeq on registration
- Added RNK special feature to 488 and 386
- Added CBCOM special feature for RTP encryption/decryption
- Added DTMF-Sweden Caller ID scheme
- Added support of North American Dial-plan on HT496/HT386
- Added support for No Key Entry Timeout UI for each port
- Added support for STUN enable/disable for each port
- Added support for Automatic Daylight Savings
- Added more Route to PSTN entries
- Added support for Open Loop Disconnect for HT488 FXO calls

- Added support for Polarity Reversal for HT488 FXO calls

================Notes on Auto Daylight Saving Time Adjustement==============

Device Configuration

The "Automatic Daylight Saving Time Rule" shall have the following syntax:
start-time;end-time;saving
Both start-time and end-time have the same syntax: month,day,weekday,hour,minute
month: 1,2,3,..,12 (for Jan, Feb, .., Dec)
day: [+|-]1,2,3,..,31
weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month.
hour: hour (0-23),
minute: minute (0-59)

If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" us not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc).

The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition.

The default value for "Automatic Daylight Saving Time Rule" shall be set to "04,01,7,02,00;10,-1,7,02,00;60" which is the rule for US.

Examples

US/Canada where daylight saving time is applicable:
04,01,7,02,00;10,-1,7,02,00;60
This means the daylight saving time starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM. The saving is 60 minutes (1hour).

NOTE:

Firmware release1.0.3.18 is has some major changes compare with 1.0.2.x releases, it may take more than 5 minutes to upgrade from 1.0.2.x to 1.0.3.18.

Make sure all the files in HT386-488-496_Release_1.0.3.18.zip are unzipped into the TFTP server.

Once upgraded to 1.0.3.18, it can NOT be downgraded to any release with 1.0.2.x.

HTTP provisioning and firmware upgrade are supported starting from 1.0.3.18.

Release Notes

1.0.3.18

- New boot loader 1.0.8.9 for better system management
- Added application level TFTP upgrade support
- Added application level HTTP upgrade support
- Added support for firmware encryption
- Added support of HTML image compression
- Added Lucent FS5000 special feature
- Added RNK special feature
- Added DTMF-Brazil/DTMF-Denmark/DTMF-Sweden Caller ID scheme
- Added firmware upgrade logging
- Added firmware validation routine (for HT-386/496)
- Added TFTP/HTTP provision upon DHCP/PPPoE completion if no IP address was available initially
- Added adaptive mechanism to resolve iLBC dual mode 20/30ms
- Added preventive measures from downgrading to any 1.0.2.x builds for all sipp2 models

- Added PPPoE service name configuration field on 386/488/496
- Removed HT488 MWI and T.38 Fax components from FXO port web UI (N/A in this case)
- Change Web UI wording for 488 to indicate call forward on ring-no-answer
- Removed "WAN" word from Status pages for 386
- Fixed fxs2 port (386/496) and fxo (488) port UI : silence suppression cannot be set

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- Fixed TFTP retry
- Fixed VLAN bug
- Fixed random number generation
- Fixed SDP handling issue on /1 after the port number in the SDP m-line (G729r8 problem)
- Changed to accept both Message-Waiting and Messages-Waiting in MWI NOTIFY (415 problem reported)
- Fixed DHCP server (488/496) fails to work if a DHCP client's MAC address is same as the cloned MAC address.

- Fixed DHCP server always send DHCP NAK when client attempt to renew expired IP lease
- Fixed DHCP client fails to restart DHCP discovery process after receiving DHCP NAK on renewal
- Reduced the DHCP polling interval to 30 seconds from 1 minute (this is the effective minimum DHCP renewal time)
- Fixed a DTMF scaling problem (low DTMF tones)
- Fixed NTP additional CI option issue
- Fixed SIP response fallback (zero IP address)
- Fixed 488 sending INV out upon FXO incoming PSTN call if FXS port has offhook auto dial configured
- Fixed 2nd port using fixed RTP port doesn't seem to work correctly (if set 54008. it uses 5401)
- Fixed HT488 assigned echo canceller to FXS port if FXS call PSTN through FXO
- Adjusted HT488 voice volume on FXS phone
- Fixed early dial issue on 386/488/496
- Fixed # is sent out as dialed number even if Use # as Dial Key is configured
- Fixed subnet issue (customer report.)
- Fixed we respond 200 OK and then a 481 for SIP INFO
- Fixed 386/488/496 cannot process BYE when challenged by 401 unauthorized server response
- Fixed 488/496 router WAN side MAC cloning bug
- Fixed 0 MAC address under PPPoE causes no RTP streams
- Fixed 386/496 FXS 1 calls FS2 crashes under certain scenarios
- Fixed P-Asserted-Header crashes
- Fixed SIP nonce overflow
- Fixed 488 far end disconnect issue

1.0.2.16

- New boot image
- Fixed no Denmark CID with recent releases for all HT product
- Fixed sending packets to dst port 0 if DNS SRV is used on both ports
- Added ARP for SIP server/proxy if it is in same LAN prior to registration, fixing the problem Nortel DMS-10 team reported
- 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 NTP does not work when NTP server is in local subnet
- Fixed we unnecessarily encode certain non-reserved characters in the Replaces parameter of Refer-To as Nortel DMS-10 team reported
- Fixed we cannot handle Call-ID over 80 bytes long correctly (Alcatel's report)

- Fixed dialog matching problem (for early dialog)
- Fixed comma-in-"Contact" bug
- Fixed we always send REGISTER to SIP server first although outbound proxy is configured when both are in IP address form
- Fixed we use different call leg information for SUBSCRIBE (MWI) throughout 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 DNS SRV send to dst port 0; (HT386/496/488)
- Fixed PCMA not offered on second call of 3WC
- Fixed block caller ID does not work on second call of 3WC
- Fixed iLBC codec is always used even if it is not configured in Preferred Codec list.
- Added special features support, current only STD and Lucent FS5000 CE
- Fixed unreliable CWCID with some phones generating near-standard signals
- INVITE is sent to NTP server if sip server in IP format, outbound proxy server in FQDN, no sip registration (Go2Call issue)
- Separated 2 accounts Web UI and added 2nd account parameters of registration required, un-register on reboot, use random port, off hook auto-dial, etc.
- Do not send SIP INVITE if no sip registration, sip server in FQDN, and use outbound proxy sever (Go2Call)
- Disallow iLBC on answering a call when the other port has a pending call (un-established) that has iLBC offered
- Added logic to prevent from starting a conference if other port has an alive call

HT386 Specific:

- Removed 386 "WAN side access" and "FXS phone line failover".
- Moved Web UI "PSTN access code" to BASIC from ADVANCED

HT488 Specific:

- Added PIN code authentication through RFC2833/SIP INFO for VoIP->PSTN call flow
- Added PIN code authentication for PSTN->VoIP call flow
- Fixed one more ring count heard than configured value and removed max ring limit of 10
- Added VoIP->PSTN direction silence detection and handler
- Fixed low volume issue for *00 and VoIP->PSTN calls

- Fixed sending *23 when making 3 way conference on FXS port if FXO port is not connected;
- Added configurable PSTN disconnect tone parameters (AC Termination, Disconnect Tone, Tone Cadence, and silence timeout)
- Removed "FXS phone line failover".
- Better handling alerting dual tone signal in driver level
- Enhancements in PSTN hang up detection, smart alerting on silence, and VAD time out on long silence after alerting
- Special ring tone as PIN code reminder

HT488 PIN Code Usage

HT488 PIN codes are used to control access to PSTN networks for VoIP- to-PSTN call flow and VoIP networks for PSTN-to-VoIP call flow. A PIN consists of up to 8 numeric digits can be configured through BASIC SETTINGS of the web configuration page. By default, there is no password protection, i.e. there is no authentication required on callers regarding the use of PSTN or VOIP networks through HT488.

For VoIP-to-PSTN calls, users need to configure “PIN for PSTN Calls” for the safe use of PSTN network. When a PIN is configured for VoIP-to-PSTN call flow, the VoIP device that calls into the HT488 FXO account needs to use RFC2833 or SIP Info for DTMF digit transmission. Users dialing the HT488 FXO account number will hear one ring tone followed by a special dial tone indicating that HT488 is ready to accept PIN code input from users. Users may enter PIN via phone keypad. The special dial tone will stop after HT488 receives the first DTMF digit. Users may continue to enter the rest of the PIN code. If PIN is correct, users will be authorized to use the PSTN network and a regular dial tone will be prompted. At this point, users may enter the PSTN destination number to make the call. When done, simply hang up.
For PSTN-to-VoIP calls, users need to configure “PIN for VOIP Calls” for the safe use of VoIP network. For incoming PSTN calls, the analog phone attached to the HT488 FXS port will ring 4 times (configurable) before prompting PSTN caller a special dial tone indicating that HT488 is ready to accept PIN code.
Please note that upon hearing the special dial tone for PIN code input, if users didn’t enter any digit, HT488 will time out and hang up the call in 10 seconds. During any stage of DTMF digits input, a 4 seconds timeout is applied to serve as an end of PIN or destination number input. Users may also use the “#” key to indicate the end of an input. If a wrong PIN is entered, the special tone will be replayed for users to try again. Users can try up to 3 times in a row. If all 3 attempts fail, users have to hang up and start again.

Please note that password protection applies for regular calls only. Un-conditional Call Forward to PSTN/VOIP and Route Calls to PSTN are not affected by PIN authentication.

1.0.2.9

- Fixed call waiting toggling crash and forced to use 711 issue on port1 while port2 is in alive call(386/496)
- Added logic to not failover to PSTN if initial boot up and FXS failover is set to Yes (488/496)
- Enabled SYSLOG on 496, both Web UI and codes
- Fixed of polarity reversal for 2 port models (386/496)
- Merged fix of SIP 183 early media ignorance.(386/496/488)
- Merged fix of SIP Info in-bound digit problem of * and # from production branch (HT386/496/488)
- HT496/386 port2 off hook hears busy tone and still be able to dial out if there is a T.38 session on port1
- HT488 fixed PSTN to VOIP issues when there is a T.38 session on port 1
- Added no ARP for 0 IP if static IP issue with HT386/496/488
- HT386/488 added Web UI radio button for enable/disable FXS failover to PSTN.
- Fixed CW crashes with new attempts to fix no voice
- HT488 added ring no answer for PSTN to VOIP call flow if there is a T.38 session already in the system

1.0.2.6

- Fixed FXS1 RFC2833 no DTMF event (broken in last build)
- Attempted to fix 488 FXS only ring once for incoming PSTN call to FXO with very first upgrade from non-web UI of number of rings
- Fixed Even though the web page still shows 0, but the actual ring times is 4. So the incoming PSTN call can be bridged to VOIP now.
- Re-merged Fix ill-formatted contact header in 302 response
- Fixed the merging issue where INVITE after 302 response bypasses outbound proxy if 302 sender use IP format in Contact header