



Release Note For HandyTone 386/488/496 Firmware Version 1.0.2.16

Aug. 25, 2005

Release Notes

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



Release Note : HT386/488/496 Firmware 1.0.2.16

- 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