Release Note for Firmware Version 1.1.0.10

November 19, 2007

NOTE:

Once upgraded to 1.1.0.10 and above, the firmware can NOT be downgraded to any previous releases other than 1.1.0.x.

Please 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_1.1.0.10.zip are unzipped into the TFTP or HTTP server.

Release Notes

1.1.0.10 11/19/2007

- Fixed RTP padding issue
- Fixed G723/G729 random crash issue
- Fixed DHCP Option 60 string cut off issue
- Fixed PPPoE IP Compression issue
- Fixed HT will pick up any incoming call in offhook idle state
- Added support to play receiver offhook tone
- Added support for remote restart via SIP NOTIFY (Event: check-sync)
- Added support for Privacy and P-Asserted-Identity headers for anonymous calls
- Separated HT287/487 Advanced Settings into 2 pages

1.1.0.5 10/23/2007

- Fixed we send registration request with empty realm in Authorization header after receiving 423 error code
- Fixed fax pass through does not fall back to G711 from LBC
- Fixed we ignore refresher= if it comes after space
- Fixed router products cannot handle large TCP frame between client and server(reducing MSS of TCP-SYN)

- Improved security handling when caller sends 180/183
- Added support for BroadSoft special feature
- Added support for G729b (annexb=yes) under BroadSoft mode

1.1.0.3 10/5/2007

<u>Bug Fixes:</u>

- Fixed bad voice quality for duplicate RTP frames from telecom blocking
- Fixed we do not offer RFC2833 when multiple DTMF schemes are selected
- Fixed we incorrectly parsed the nonce param
- Fixed we do not update SIP server IP address returned by DNS

BT-120 specific

- Fixed BT-120 crash when MESSAGE key is pressed and voicemail user ID set to "sip:voicemail@domain.com"
- Fixed we accept broadcast SIP messages

HandyTone specific (HT-287/487):

- 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

<u>New Features:</u>

- Added option to allow disable CWCID, fixing some HT-287 CWCID not working in 1.1.0.2
- Added configurable registration backoff timer, P138 in seconds (1-3600, default 20)
- Changed when registration fails, the SIP port is reset to another random port if system is configured to use random port
- Changed description for "Enable Call Features" in web UI

1.1.0.2 03/18/2007

- 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
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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

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