
GRANDSTREAM NETWORKS

Firmware Release Notes

Firmware Version 1.0.9.4

Product Name: HT701/HT702/HT704

Date: January 26, 2018

CHANGES SINCE FIRMWARE RELEASE 1.0.8.2**BUG FIXES**

- Update default certificate gxe50xx.pem for provisioning due to old one expired.

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Firmware Version 1.0.8.2

Product Name: HT701/HT702/HT704

Date: September 28, 2015

SUMMARY OF UPDATES

The main purpose of this release is additional features implementing and addressing stability issues observed in previous releases.

CHANGES SINCE FIRMWARE RELEASE 1.0.7.3**BUG FIXES**

- Fixed during concurrent call performance test, other devices cannot hear the audio from HT7XX
- Fixed device sometimes enters fax mode during a call
- Fixed incoming INVITE gets rejected under TCP/TLS specific scenario
- Fixed under TFTP upgrade mode, sometimes device cannot upgrade or access the WebUI
- Fixed device sometime failed to boot-up after firmware upgrade
- Fixed WEB_UI status page DID always displays NO with or without a DID assigned
- Fixed device failed to boot up if setting the wrong format address on option 66
- Fixed during a CW call flash back to the 1st leg and hanging up the second leg can cause the HT802 to initiate a call transfer
- Fixed NAPTR/SRV change does not take effect without reboot
- Fixed option "Outgoing Call Duration Limit" does not take effect without reboot
- Fixed "Force Timer" will not take effect if "Can Reinvite" is set
- Fixed [TR-069]: Missing Parameters
- Fixed "SIP Registration Failure Retry Wait Time" is fixed in 20 minutes when receiving 403 Forbidden
- Fixed [BJ Telcom] Attended Transfer failure
- Fixed Pulse Dialing does not work
- Fixed under "Enabled and forced" SRTP Mode, device cannot set up call with IPPhone
- Fixed Time Zone "GMT+12:00 (Auckland, Wellington)" shows wrong time
- Fixed New Zealand time zone shows wrong time
- Fixed XML Parser does not take multi-line P values in config file
- Fixed SIP TLS Private key and certificate cannot be provisioned using XML
- Fixed DNS-SRV connection failed if first option is unavailable using TCP
- Fixed crash issue with core dump
- Fixed device will never register if the 1st and 2nd NTP request at boot up have different time
- Fixed device does not register if DNS server responds without a IP address A record
- Fixed all ports stopped working under auto dial performance tes
- Fixed device does not take an IP-Address from PPPoE server through VLAN
- Fixed FSK tone heard locally on outgoing call under Broadsoft special mode

- Fixed during SIP registration device does not use the shortest expires time between the proxy and it's locally configured timer
- Fixed device port did not work after the phone Hold/Unhold many times
- Fixed device sends a wrong stop Tone if CID exceeds 9 digits
- Set TLSv1 protocol to be used with wget
- Fixed MWI issue
- Fixed Event lost issue during performance testing

ENHANCEMENTS

- Added support of 3-Way Conference utilizing MTAS Ad-Hoc Conference
- Add configurable option to enable/disable Connected Line ID Presentation
- Added option for Fax Tone Detection Mode
- Customized 3WC and transfer features
- Added Call limit option on HT702 and HT704
- Added support to hide passwords from command line
- Updated time settings for Moscow, Russia
- Added support of sending SIP options messages to verify connectivity to the SIP server
- Added option to enable/disable Crypto life time when using SRTP
- Added support for DHCP OPTION 160
- Added option to disable PnP SUBSCRIBE
- Changed "Use Random SIP Port" to correct P values. Set "Enable High Ring Power" to "Yes" in China ITSP mode.
- Added support to redirect firmware/config file upgrade request upon receiving HTTP 302 response
- Modified provisioning download to work when protocol is changed from HTTP to HTTPS

Firmware Version 1.0.7.3

Product Name: HT701/HT702/HT704

Date: November 4, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.6.1

BUG FIXES

- Fixed [TR-069] dateTime Type typo
- Fixed sometimes there is slow ring, or no ringing at all when using certain analog phones
- Modified some options' value for Chinese web UI display
- Fixed web UI display issue under Chinese. Added "Allow DHCP Option 120" to Chinese web UI
- Fixed when registered to outside SIP Server remotely, SIP Server cannot receive register request
- Fixed device rejects SIP reply from Outbound Proxy under some conditions
- Fixed after receiving some DTMF via SIP INFO, quickly press digit button, audio volume will become very low
- Fixed SIP Server and Outbound Proxy cannot be configured with host address like x.x.x.0
- Fixed modem test failure. Added web UI option "Disable Network Echo Suppressor" under each account/profile (P 4441/4442/4443/4444)
- Fixed the problem when offhook auto dial is configured with nonzero delay timer, device does not play stutter dial tone if there is voice mail
- Fixed device fails to download provisioning file by HTTPS with mutual TLS enabled
- Fixed no ring back tone when "Disable Call-Waiting Caller ID" is set to "YES"
- Fixed device fails to handle multiple m lines in SDP correctly causing incoming call fail. Added web UI option "Disable Multiple m line in SDP".
- Fixed after clearing the dial plan, user can still successfully Save and Apply
- Fixed device does not show P20713 option in Chinese Web UI

ENHANCEMENTS

- Added support to announce the registration number via feature code
- Added SYSLOG message when Registration LED goes out
- Added support for NAT transfer feature. Added web UI option "SIP REGISTER Contact Header Uses"
- Added support to configure the ringing voltage and frequency
- Added support to configure DTMF generation timing. Added web UI option "Generate Continuous RFC2833 Events" on web UI
- Added configurable RFC 3261 timer D length support
- Added 911 emergency call support
- Added %MODEL% support for provisioning

Firmware Version 1.0.6.1

Product Name: HT701/HT702/HT704

Date: May 30, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.5.10

BUG FIXES

- Fixed unable to establish a call to extension #9
- Fixed device doesn't handle multiple m lines in SDP correctly causing incoming call fail
- Fixed device broadcasts ARP at a very frequent interval. Modified ARP to unicast. ARP interval is configurable. Number of consecutive gateway ARP response failures is configurable
- Fixed HTTP GET provisioning request displays HTTP username/password in plain text. Added web UI option "Always Authenticate Before Challenge"
- Fixed TCP inter-arrival Jitter and Cumulative number of packets lost does not match Wireshark RTP stream analysis for the RTP stream from the media relay to the device
- Fixed iLBC Frame Size could cause calls without audio
- Fixed device doesn't send UPDATE after "Enable force Timer" is set to "Yes"
- Fixed device does not honor expires timer for registration of 20 seconds
- Fixed after setting SIP profile as inactive device can still make calls although web UI displays "Saved and Applied"

ENHANCEMENTS

- Added support for configurable TTL Value for Keep-Alive Messages
- Added separate Tos/CoS settings for SIP and RTP
- Added option "Do Not Escape '#' as %23 in SIP URI" in web UI

Firmware Version 1.0.5.10

Product Name: HT701/HT702/HT704

Date: April 2, 2014

CHANGES SINCE FIRMWARE RELEASE 1.0.5.8

- Removed 802.1p priority value (NATed traffic) setting from web GUI
- Fixed device keeps re-sending Invite after phone is put onhook
- Fixed SLIC setting set back to default after reboot

Firmware Version 1.0.5.8
Product Name: HT701/HT702/HT704
Date: February 21, 2014

BUG FIXES

- Fixed if P143 is set to No and Router/Modem offers DHCP Option 2(time offset) the ATA will still accept the time offset.
- Fixed ShoreTel Interop: no audio issue after HT704 Workgroup Agent answers call
- Fixed MetaSwitch Interop: device cannot retrieve call after hold
- Fixed Metaswitch Interop: added timeout for Blind transfer in case transfer fails
- Fixed duplicate packets when 802.1p priority value is set and VLAN tag is 0
- Fixed memory leak problem when processing SIP OPTIONS
- Fixed device traps into reboot loop under certain condition
- Fixed getting empty upgrade server path via IVR causes device crash
- Fixed HT704: FXS1-4 making hot line call causes device get busy tone
- Fixed DTMF via RFC2833 loses audio
- Fixed DTMF quality from inbound RTP Events are out of spec
- Fixed device sends Loop Current Disconnect for 4xx SIP responses
- Fixed device sends 0.0.0.0 in Register Via header when unplug and plug in Ethernet cable
- Fixed OnHook CAS and CID level is not in range of SIN 227 spec of -5.8dBm to -37.8dBm
- Fixed Off hook CID silence time after the DTMF D Ack is 285ms-280m

ENHANCEMENTS

- Added support for Subscribe authentication
- Added support for Distinctive ringtones using Alert-info string
- Changed DHCP options 42 and 2 to be enabled by default
- Added SHA-256 support
- Added checking for "100rel" Require header if "Validate Incoming SIP Message" is set to Yes

Firmware Version 1.0.5.2
Product Name: HT701/HT702/HT704
Date: October 11, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.4.14

BUG FIXES

- Fixed BroadSoft interop: device cannot switch between two lines after receiving server NOTIFY
- Fixed DNS Issue during Internet outage
- Fixed device continuously sends SUBSCRIBE SIP messages even though voicemail is disabled
- Fixed OnHook CAS and CID level is not in range of SIN 227 spec of -5.8dBm to -37.8dBm requirement, suggests -20dBm
- Fixed device cannot normally boot up if upgrade firmware together with configuration file via HTTPS
- Fixed device cannot boot up if it try to download the same version via HTTPS
- Fixed Internet light stays on after Ethernet cable has been unplugged
- Fixed device creates extra call object leading to one channel stuck in CALL_ENDING, causing no more incoming phone call unless device off/on hook
- Removed HTTP Access option from web UI
- Fixed IVR voice prompt does not read the correct IP address in certain scenario
- Fixed wrong order on Route header with ACK and BYE requests

- Fixed device rejects NOTIFY for UPnP Auto-provisioning. If NOTIFY is for multicast UPnP provisioning, device does not check to-tag
- Fixed in basic call conversation, after hold and resume, device keeps sending ReINVITE
- Fixed device makes anonymous calls under TLS mode, can cause CPU usage high and leads to abnormal behavior
- Fixed off hook CID silence time after the DTMF D ACK is 285ms-280ms
- Fixed UK – outbound call Ring back tone will have 440Hz overlap
- Fixed device does not support BT style MWI stutter dial tone
- Fixed inbound faxing with T.38 protocol having HDLC:fcs-BAD frames the SIP stack will stop working
- Fixed after listening voice mail the FXS LED still blinks
- Fixed device does not unsubscribe and re-subscribe if account information has been changed
- Fixed event lost during performance testing. Fixed a memory leak that occurs with the combination of MWI and ongoing call
- Fixed a crash problem caused by CANCEL on early media server while receiving 200 OK
- Fixed in new profile change the vocoder via IVR does not take effect
- Fixed device ringing didn't timeout
- Fixed device has Dial Tone after Current Disconnect instead of silence
- Fixed CPE doesn't report gateway related info in the TR-069 INFORM
- Fixed port in a response is not set properly if not explicitly specified in the Via header for TLS transport
- Fixed TLS block read issue where the read may not return the complete message
- Changed message receiving reassembling logic for SIP over TCP/TLS
- Fixed the problem when the device does not follow record route header route. This occurs when the route set is not empty and the request URI contains maddr parameter

ENHANCEMENTS

- Added option to remove PPI & Privacy Header
- Added Option "Add Auth Header On Initial REGISTER" under FXS port page (P value 2359 and 2459). If set to "yes", include authorization header in the Register request
- Added option for "Voice Frame per TX"
- Added option to disable the Hook Flash Function
- Added support for Voice Frame per TX in TR-069 Data Model
- Added the support for receiving TCP/TLS "\r\n" keep-alive messages

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Firmware Version 1.0.4.14
 Product Name: HT701/HT702/HT704
 Date: August 1, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.4.8

BUG FIXES

- Fixed device does query for provisioning when powered on and connected to LAN, but without Internet connectivity
- Fixed device crashes when it keeps receiving SRTCP: Receiver Report
- Fixed device works abnormally after we keep testing call-transfer on it for 40 hours.
- Fixed a Current Disconnect problem
- Fixed HT704 DTMF issue on second FXS
- Fixed Internet light stays on after Ethernet cable has been unplugged
- Fixed issue with fault line recovery mechanism
- Fixed Broadsoft Interop: device does not update "anonymous" with "Privacy:id"
- Fixed with SRTP set to enable and forced, device crashes if it makes a call to an extension that is

in call

- Fixed no audio when return to call after fax. Removed "Send Re-INVITE After Fax" from web UI, since HT7XX does not support this feature
- Fixed Broadsoft Interop: device Connected Line ID is not updated after Call Transfer or Call Forward. Fixed problem where Connect Line ID was not updated when we receive the 2nd 18x in a back-to-back 183-180, 183-183, or 180-183 case
- Fixed with Firmware Server Path and Config Server Path set to blank, retrieving the Firmware Server Path and Config Server Path via IVR will cause device crash
- Fixed device stops sending RTP after a long duration 3 hour call in iLBC
- Fixed the problem when "Validate Incoming SIP Message" is set to Yes
- Fixed device cannot identify SIP NOTIFY's contact-type with subtype "url"
- Fixed HT702 FXS2 port wrong default value for "Preferred DTMF method"
- Fixed Prompt Tone switch slow and wrong frequency
- Fixed device does not send out DTMF after receiving 183
- Fixed device will fall into reboot cycle if setting Pvalues via XML config file
- Fixed device still tries to send Register messages to Failover SIP Server even though it has successfully registered to primary sip server.
- Fixed issue that 3CX UPnP Multicast SUBSCRIBE is sent to the outbound proxy if outbound proxy is configured
- Fixed crash problem when "Subscribe for MWI" setting is changed
- Made changes so that when "Subscribe for MWI" is changed from No to Yes, unsubscribe will not be sent
- Removed setting of pvalues when user presses *16, *17, *30, *31, *50, *51. Changes done via * code feature will only be applied after reboot.
- Fixed Line 2 (Lines 2-4 on HT704) Calling Waiting Static Noise
- Fixed HT704 loses Dial Tone after power cycling unit once after factory reset
- Fixed HT704 stops trying to reach fm.grandstream.com after being powered for some time and not connected to the Internet
- Fixed Memory Leak issue when "SUBSCRIBE for MWI" is set to Yes
- Fixed Broadsoft Interop: device returns 500 Internal Error to MWI NOTIFY
- Fixed with NAT Traversal set to UPnP, device will always try to DNS resolve SIP server, even though SIP server is in IP address format
- Fixed device fails to boot up after two consecutive power failures during provisioning
- Fixed Typell-CWCID MARK and CW to CAS timing are out of scope. Reduced Off-hook mark with additional 6.66ms
- Fixed Call Waiting tone heard when there is no Call Waiting call
- Fixed device crashes during IP Call under some conditions
- Fixed Broadsoft Interop: HT5xx Connected Line ID is not updated after Call Transfer or Call Forward
- HT701 fax issue with Call Centric and PSTN
- Fixed when dial plan is \uff5b[*x#]+\uff5d, call feature doesn't work
- Fixed device does not parse the SDP included in the 181 message
- Fixed device does not reply to INVITE with Diversion header
- Fixed incoming calls from GW failed because the device did not respond to the INVITE with a 180
- Fixed call transfer to a nonexistent number, the call ends
- Fixed device cannot hear remote party while remote responses the ACK with no SDP

ENHANCEMENTS

- Added support for DHCPv4 Option 120
- Enhanced Dial Plan implementation. Modified "." to indicate there is zero or more of the preceding element
- Added support for Brazilian CID
- Added SUBSCRIBE messages to multi-cast address from AA product
- Added support for SIP NOTIFY "resync" event
- Added support for compact session timer header

- Added "ETSI-DTMF prior to ringing with LR" to the Caller ID scheme's drop-down-list
- Added web UI configuration for DTMF Caller ID Start/Stop Tone (P4661-4668)
- Added support for one step SIP NOTIFY reboot event
- Added support for dual frequency dial tone
- Expanded " Lock Keypad Update" option to be able to reset only Basic Settings

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Firmware Version 1.0.4.8
 Product Name: HT701/HT702/HT704
 Date: January 16, 2013

CHANGES SINCE FIRMWARE RELEASE 1.0.4.3

BUG FIXES

- Reduced Howler tone volume
- Fixed Broadsoft Interop: HT5xx returns 500 Internal Error to MWI NOTIFY
- Fixed HTTP GET request uses HTTP 1.1 causing XML provisioning fail with certain HTTP servers
- Fixed Device does not reboot after configuration server path changes in configuration file
- Fixed stutter dial tone is still played when Disabling Visual MWI is set
- Fixed device does not play busy tone when call is ended by the other side
- Fixed device gets stuck in a reboot loop with P231value set in configuration file
- Fixed interrupting upstream router connection for an extended period of time causes SIP stack to crash
- Fixed RFC2833 does not work as expected
- Fixed device does not play dial tone after successful semi-transfer
- Fixed device Subscribe for MWI negotiation error
- Fixed with Dial Plan Prefix set to a large number, device will crash after dialing about 40 digits

ENHANCEMENTS

- Added support to disable HTTP access
- Added support for "Authenticate incoming INVITE"
- Added support for " Register before Expiration"

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Firmware Version 1.0.4.3
 Product Name: HT701/HT702/HT704
 Date: November 7, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.3.1

BUG FIXES

- Fixed device does not check for binary configuration file after GAPS redirection under certain conditions
- Fixed CNG negotiation does not work
- Fixed device does not play second dial tone after pressing #90 feature code
- Fixed Type II-CWCID MARK timing out of scope
- Fixed On-Hook CID signal strength is too high
- Fixed pass-through fax failed
- Fixed device does not play Ring Back Tone when it off hook auto dials another device
- Fixed BroadSoft Interop: device does not play ring back tone under BS special feature
- Fixed if P8 is set to 01, IVR does not announce the correct IP address

- Fixed device does not increment nc on the first re-register
- Fixed device DHCP client keeps rebooting if default router is set to 0.0.0.0
- Fixed crash problem – if account 1 is enabled, and account 2 is disabled, device will crash when trying to make an outbound call
- Fixed when caller is not in transfer or hold or waiting status, Current Disconnect does not work when caller's call setup failed
- Fixed device gets in a reboot loop if XML configuration file contains same P numbers but with different values
- Fixed BroadSoft Interop: Connected Line ID is not updated after Call Transfer or Call Forward
- Fixed device crashes under long term testing
- Fixed device starts to register every 2 seconds after receiving IB PCMU fax
- Fixed device SIP stack stopped responding to inbound calls under certain conditions
- Fixed Circular Hunting Group ringing error
- Fixed Time Zone is delayed by one hour
- Fixed Blind Transfer failed under certain condition
- Fixed device cannot hang up with FLASH button when hook flash is disabled
- Fixed device does not Detect BT Timed Break Recall
- Fixed Direct IP Call fails when use # as re-dial key
- Fixed iLBC codec does not work
- Fixed Symmetric RTP does not work when device is caller
- Reduced device boot-up time
- Fixed IVR prompts cut off last few ms
- Fixed device failed to parse second XML encrypted profile

ENHANCEMENTS

- Added UPnP client support
- Added option “Use DNS to detect network connectivity”
- Separated Reset functions under Web UI. Display ISP data reset for user and all three types of reset for admin
- Expanded Lock Keypad Update option to reset only Basic Setting options
- Added support to apply settings change without reboot
- Added new IVR for WAN Cable Disconnected, Internet Connection Down, and Device not Registered
- Reduced device boot-up time
- Improved Jitter Buffer performance
- Improved fax pass-through performance
- Updated gs_cpe release to 1.0.1.28

Firmware Version 1.0.3.1
 Product Name: HT701/HT702/HT704
 Date: August 21, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.1.6

BUG FIXES

- Fixed crash when call is canceled before SIP stack is ready
- Fixed device locked up after provisioning under certain condition
- Removed firmware key option from web UI
- Fixed device will download corrupted firmware without doing a checksum
- Fixed device does not try to connect to server for firmware files under certain conditions
- Fixed during a call if the device goes on hold then remote Hold/Unhold then device unholds RTP media will not reconnect

- Fixed SIP stack stops responding on an outbound call
- Fixed FSK generation for on-hook transmission is out of spe
- Increased TX/RX gain settings down to -12dB from -6dB
- Fixed Call Waiting tone interference
- Fixed device takes too long to respond to 407 after sending a BYE
- Fixed changing Gain configuration will cause Call ID/DTMF not functioning
- Fixed poor PCMU faxing with jitter
- Fixed device does not ring if INVITE contains: Call-Info: answer-after=0
- Fixed with SRTP enabled, device crashes when callee FLASH during conversation
- Fixed with SRTP enabled, FLASH in conversation (FXS1 calls FXS2) will result in crash
- Fixed Hunting Group with second incoming call issue
- Fixed device not negotiating codec correctly
- Fixed device does not play Call Waiting tone correctly
- Fixed device cannot do upgrade when use TFTP method and the upgrade path contains digital
- Fixed call fails when device get challenged by 407 followed by 401
- Fixed device does not support the expansion of processing
- Fixed after Redirection registered in from and to the server IP is not the same.
- Fixed device will proxy jump on new DNS refresh
- Fixed Half-Attended Transfer/Attended Aransfer crash issue
- Fixed device ends call when caller request timer is enabled
- Set default DTMF method as RFC2833 / SIP-INFO / In-Audio
- Fixed Call Waiting tones causes current conversation mute

ENHANCEMENTS

- Add an option to disable the Hook Flash function
- Improved jitter buffer performance
- Display different web page if unit does not require reboot after parameter changes
- Added support for CFG fetch via SIP NOTIFY resync event.
- Added option to wget to use ipv4 only
- Updated IVR with new options
- Added support for Call limit option
- Reduced system boot-up time
- Added an option to Enable/Disable each FXS Port
- Added support for programmable ringing frequencies

Firmware Version 1.0.1.6

Product Name: HT701/HT702/HT704

Date: July 5, 2012

CHANGES SINCE FIRMWARE RELEASE 1.0.0.18

BUG FIXES

- Fixed device crashes, if it makes Direct IP call without register to SIP server
- Fixed device crashes if set a strange dial plan
- Fixed HT701/HT702:illegal value can be saved in webUI option :off hook auto dial
- Fixed Blind Transfer abnormal if set sip transfer to TLS. Change to always use even number for RTP port
- Fixed basic call abnormal if set the SIP Transport to TLS
- Fixed HT702:Three way conference cannot be set up
- Fixed dialing Busy Forward is invalid
- Fixed when "Use # as dial key" is set No, for direct IP call the # key can also be used as the send key
- Fixed device cannot boot up when upgraded from firmware version 1.0.0.18

- Fixed Off-hook Auto-dial doesn't accept star key
- Fixed BroadSoft interop: fax pass-through fails with re-INVITE
- Fixed while using TLS incoming calls will stop working and go straight to voicemail
- Fixed HT704: update button is missing on webUI page :FXS PORTS
- Fixed BroadSoft interop: HT7xx fails to handle INVITE with Diversion Inhibitor
- Fixed BroadSoft interop: REGISTER Failback does not send to the primary server
- Fixed BroadSoft interop:INVITE Failback does not send to the primary server
- Fixed HT70x default Dial Plan not consistent with other products
- Fixed device auto ends calls with long time conversation
- Fixed no PPI in INVITE
- Fixed BroadSoft interop: add support for P-Preferred-Identity and Privacy header
- Fixed device leaks memory when STUN and "Validate Incoming Messages" are enabled
- Fixed callee would not use the audio coder negotiate in the 200OK
- Fixed Syslog for NTP does not include MAC address
- Fixed Cancel was sent without UAS provision response
- Fixed device does not handle the RFC-3891 correctly
- Fixed device cannot register when SIP transport set to TCP(or TLS) and both Primary and Failover used sip server
- Fixed device only do not failover to second server when DNS mode set to SRV
- Fixed Prack CSeq Always Incremented upon receiving a 18X message retransmission
- Fixed device does not accept Re-INVITE with "t38+other codec" when pass-thru
- Fixed # key abnormal in 3cx platform
- Fixed device does not roll to Failover SIP Server if Primary SIP Server is down
- Fixed some analog phones cannot display CID
- Fixed "SSL.." related web UI description not clear
- Fixed device does not respect dual cadence for call progress tones
- Fixed CNG negotiation does not work.
- Fixed Ringback Tone when dialing port which is already off-hook
- Fixed device auto ends the conversation when set caller request timer
- Fixed Half Attended Transfer failed when set busy transfer
- Fixed one-way-audio issue of SRTP
- Reduced boot-up time
- Fixed sending Registration Request every second
- Fixed HT704-v1.4 Ethernet LED is on without connecting RJ-45
- Fixed direct IP call cannot get waiting tone
- Fixed Transfer failed led to set up the erroneous 3-way conference
- Fixed REFER_TO header - SIP URI unreserved characters "." and "-".
- Fixed BroadSoft Interop: HT70x does not use To tag from 200OK to re-SUBSCRIBE MWI.
- Fixed XML file can be download and write when file format is wrong
- Fixed press flash can enable SRTP
- Fixed BroadSoft interop: add support to display Connected Line ID
- Fixed device doesn't use maddr from the Route Record Header
- Fixed after flash RTP continues being sent out
- Fixed device gets in a loop of sending BYE messages
- Fixed device can't auto dial single number
- Fixed device didn't ringing in some DTMF schemes
- Fixed an issue with dial plan {x+}

ENHANCEMENTS

- Added support for Current disconnect
- Added an option to Enable/Disable each FXS Port
- Added support to send CID DTMF "0000000000" instead of "00" when anonymous
- Added ability to configure delay for the off hook auto dial

- Added support for generic XML config file
- Added display of gs_cpe version in status page
- Added CPE SSL Certificate and CPE SSL Private Key in "Advanced" web page
- Added support for OK and Apply changes in web UI
- Added a configuration parameter to overdrive User-Agent header
- Added support for IP change notification to application