GRANDSTREAM NETWORKS
Firmware Release Notes
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 offsest) 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

==

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 TypeII-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 an 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

===

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"

==

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