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Customer Deliverable Documentation
Revision 1.00, October 28, 2014.

SmartMedia v2.7 Release Notes

New to the v2.7 Release

- N+1 (STM1 Only)
- Lawful intercept
- Radius AAA
- Audit logs
- Test call
- Support CASR2 for DS3/STM-1
- Support H.248 CAS Packages
- Allow remapping of incoming leg profile by route

History of Solved CTS Cases

The following list refers to open cases in the Change Tracking System 'CTS'

Build Series v2.7.148

Enhancements:

- Ability to upgrade to v2.8 via web portal

Bug fixes:

- Missing TONE_END event prevents call to allocate G729 codec resulting in no-voice condition
- Refuse to Save User with no password

Build Series v2.7.147

Bug fixes:

- System rebooted due to excessive RAM usage

Build Series v2.7.146

Bug fixes:
- Outgoing media is cutoff when incoming media is present
- Problem with RSeq sent by SN10K

Build Series v2.7.145

Enhancements:
- SIP-I : append F in ISUP called party number without adding to SIP "to" header

Build Series v2.7.144

Bug fixes:
- Gateway is using 100% CPU when processing an H.248 digitmap completion event with an inline digitmap description
- Tboamapp restarted following database access failure
- H.248 module restart after making call with echo cancellation property set

Build Series v2.7.143

Bug fixes:
- Engine stops playing files when gracefully stopped
- Echo cancellation is always turned off if H.248 tdmc/ec is present in MGC request

Build Series v2.7.142

Enhancements:
- Support Generic Name in ISUP (ANSI95 and Telcordia)
- Report protocol information upon OnCallLegfailure

Bug fixes:
- Authorization problem with "Test Call"
- Toolpack engine may not rotate its log files
- M2UA links are still up when the SCTP connections are down
- V.21 flags following CED in SIP to SIP call don't trigger switch to T.38 mode

Build Series v2.7.141

Enhancements:

- Enable FAX tone detection on SIP to SIP calls
- Support Generic Name in ANSI ISUP

Bug fixes:
- Authorization problem with "Test Call"

Build Series v2.7.140

Enhancements:
- Allow Toolpack major upgrade through WebPortal

Bug fixes:
- Engine can't open database on switch-over
- Play file fails, on Linux, if first file path choice among multiple is not found
- Endpoint info is not correctly printed on the UCT when multiple c lines
- Tone detection may miss a tone after play file finished on TMedia units without DSPs
- Software block does not show active calls
- Add an option to set Forward Call Indicator (FCI) to a specific value
- Engine reports unexpected CmcNotifDigitPlayingDone event when relaying SIP Info
- Allowed multiple ISUP Userpart in the configuration
- Application Param "Call answer timeout" in toolpack_engine not working on outgoing SIP call legs
- CDR timezone (time) not showing normally

Build Series v2.7.127

Enhancements:
- Option to configure the ephemeral termination range
- Add an option to set Backward Call Indicator (BCI) to a specific value
- Add option to enforce BCI in ANM

Bug fixes:
- Tone play not restarted after codec switch, before 200 OK is received
- Toolpack ignores the SDP provided in SIP UPDATE
- Symmetric NAT not working when an odd port is used for the RTP stream
- Play file fails, on Linux, if first file path choice among multiple is not found

Build Series v2.7.126

Bug fixes:
- Detected FAX tones may be ignored in build 2.7.125

Build Series v2.7.125

Enhancements:
- Add incoming NAP into label routing
- Emergency call handling or MLPP
- Transfer bearer capability between TDM and SIP
- ISUB (ISDN subaddress) relay to/from SIP
- Have incoming NAP, trunk and timeslot in incoming leg available in routing script
- tbdebug run in standalone (outside of tboam) with database dump and tb640debug

Bug fixes:
- DS3 LIU generate heavy load of interrupts
- Sending SIP BYE/CANCEL without SDP
- Fixed bad DSP double-tone detection when tone has small gaps
- "Testing Script" jammed the webportal while testing routeset routing
- Cause code was incorrectly picked from reason header
- Invalid community in SNMP request causes to restart adapter
- Cannot choose application_name of Streamserver
- ISUP process crash on SPIROU variant

Build Series v2.7.124

Enhancements:
- Allow bidirectional Early Media through a configuration option

Bug fixes:
- Fixed bad DSP double-tone detection when tone has small gaps

Build Series v2.7.123

Enhancements:
- Add option to send "In Service" messages to MGC on timeslot base

Bug fixes:
- SS7 stack drops ANM, connected number IE with too many digits
- "Early media mode" is not applied when ISDN call proceeding is not received
- Routing script nap_group_weight_load_balancer.rb with :load_outgoing_only option takes incoming call in calculation
- Audio is not connected when 100-trying is missing and 183 is handled as "progress" instead of "alert"

- RLC is not generated on SIP-I BYE 200 OK
- Delay with call releasing from engine after being connected to mixer

Build Series v2.7.122

Enhancements:
- Accept ISDN calls with encoding (aLaw/uLaw) different from variant's default

Bug fixes:
- The H.248 does not support receiving a LOCAL or REMOTE descriptor containing only an empty line

Build Series v2.7.121

Enhancements:
- Support configurable packetization times for RFC 4040
- Allow to control mixers resize-down timer

Bug fixes:
- OBCI is not inserted in ACM when using Q767 variant
- Receiving a 183 after a 180, if ringback tone is enabled, does not open the path
- sip_layer problem when a parsing error is detected in a mandatory header
- Error during re-sync with toolpack_engine: TbCmcThreadJobPost: Ignoring MsgId=0x0A020009
- Mixers get stuck in engine upon termination after disabling SetRecordPlayback while recording

Build Series v2.7.120

Enhancements:
- Add RTCP/TDM stats in UCT and Radius/text CDR
- DMS CLASS announcement server functionality

Bug fixes:
- ISDN Progress Indicator with Location value 4 is refused when using 5ESS
- Receiving a 183 after a 180, if ringback tone is enabled, does not open the path

Build Series v2.7.103

Bug fixes:
- Prevent Detected sequence errors (from network) with Fax T.38

- Ruby scripts are limited to 64KB
- NAT traversal not re-established after putting call on hold
- Call trace reports unexpected 200ok between some traces
- Support receiving a re-INVITE before receiving the ACK of a previous re-INVITE
- Cannot change TDM line Services Loopback state when backup is active
- Error in Status Menu N+1
- ScriptAttribute:CustomCdrValue is not working with "Test Call" from the web

Build Series v2.7.102

Bug fixes:
- Script variables 'sip_headers' and 'request_uri' not available during a SIP REFER
- Firmware does not support physical HW port count lower than 3 for DS3 units
- Incomplete IUA Link stats if more than one IUA link is present
- Call on hold (remote "a=sendonly") reports "no packets received from network"
- SDP parsing error: attribute pcfg causes invalid SDP to be generated by SDP parser
- [REMOVED]OCN counter must not be added to redirection counter when OCN == RN
- SDP with H248 was missing Session-Level parameters (o,s,t)

Build Series v2.7.101

Bug fixes:
- DC power supplies don't report failure status - Webportal should say 'unknown'
- Announcement server: Call is terminated even if the script ends with a long silence
- ISDN PROGRESS is dropped by the stack when receiving an invalid location value
- OEM references during installation
- Avoid sending a "reset" on a timeslot after reaching the answer timeout (T9)
- Test Call: CallTrace shows the play/record file failure trace BEFORE the record request trace

Build Series v2.7.100

Bug fixes:
- Weight and group routing script % miscalculation when weight is "0"
- STM1 E1#1 having RSN
- Missing ScriptAttribute:CustomCdrValue in text CDR when using both Radius and Text CDRs

Build Series v2.7.90

Bug Fixes:

- Announcement server: Call is terminated even if the script ends with a long silence
- ISDN PROGRESS is dropped by the stack when receiving an invalid location value
- Avoid sending a "reset" on a timeslot after reaching the answer timeout (T9)

Build Series v2.7.89

Enhancements:
- Support ISDN CNAM (caller name) using facility IE

Build Series v2.7.88

Enhancements:
- Support one-line SIP Diversion message
- Forward all SS7 CPG in the call flow

Bug Fixes:
- SIP module occasionally failed to load on first attempt
- SmartMedia may restart due to failure (every few weeks)
- A circuit group remains as resetting state after moving line services to other CIC groups
- N+1 Patch panel may not boot properly
- Long delay in playing tone request after playing file

Build Series v2.7.87

Bug Fixes:
- Incoming ISDN call refused when containing unknown FACILITY information element

Build Series v2.7.86

Bug Fixes:
- Incoherent call answer timeout behavior in SmartMedia
- OCN counter must not be added to redirection counter when OCN == RN

Build Series v2.7.85

Bug Fixes:
- Announcement script is loaded from wrong configuration
- Trunk alarm is wrong in SmartMedia when cable not properly connected to backup SN10k
- Announcement server: Call is terminated even if the script ends with a long silence
- Routing script limited to 40 call attributes, some calls may require more attributes
- REL to return Public Network Remote User instead of Network beyond interworking point

Build Series v2.7.84

Enhancements:
- Add option to remove Group message from ISUP SPIROU

Bug Fixes:

- ISUP shutdowns after configuration change
- M3UA: PSP bounces for an association

Build Series v2.7.83

Enhancements:
- Support Q.1950 call progress tone generator package (bcg) for H.248 (additional scenarios supported)

Bug Fixes:
- H.248 restarts when receiving an echo cancellation request immediately after a continuity check request

Build Series v2.7.79

Bug Fixes:
- H.248 timeout connecting termination while playing infinite tone

Build Series v2.7.78

Bug Fixes:
- Filter unrecognized SS7 IEs for ANSI variant

Build Series v2.7.77

Bug Fixes:
- DTMF detection not automatically re-enabled after SmartMedia switchover
- Buzzing sound when busy tone is used after ring tone on units without DSPs

Build Series v2.7.76

Enhancements:
- Support Q.1950 call progress tone generator package (bcg) for H.248

Build Series v2.7.75

Bug Fixes:
- SCTP association takes longer than 5-6sec to be declared down
- Send GRS only when required
- INIT message should always include IP address of source
- Problems during SmartMedia upgrade
- RTP stats for SS7 NAP
- No CPG (Alerting) in a SS7 call flow.

Build Series v2.7.73

Bug Fixes:
- TCAP(SCCP) configuration without called no in TC-end

Build Series v2.7.72

Bug Fixes:
- SN10K selects session connection instead of media connection (SDP)

Build Series v2.7.71

Bug Fixes:
- Add routing script option to control if diversion is allowed
- Mixer Noise Suppression Limit causing problems with audio streamed into the mixer
- Incorrect Q850 reason cause mapping

Build Series v2.7.70

Bug Fixes:
- SmartMedia need to retain called/calling + creation/answering time to avoid CDR with empty values upon resync corner cases
- SN10K system could not switch from the Secondary to the Primary

Build Series v2.7.65

Bug Fixes:
- RESTART_ACK from remote equipment refused due to invalid information element

Build Series v2.7.64

Enhancements:
- Be able to insert a custom information in CDR text and radius

Bug Fixes:
- Call drop upon receiving SIP 183 with error "Request MsgId SIP_CMD_UA_PRACK_REQ Failed"
- SmartMedia restarts when using tone detection mask of ".."

Build Series v2.7.63

Enhancements:
- Add an option to copy SIP header from incoming leg to outgoing leg
- Give access to the original drop cause in CDR and UCT

Bug Fixes:
- silenceSupp attribute in SDP returned by SN10k to softswitch is improperly formatted
- HDLC controllers drops all incoming data if sequence is "valid frame+ABORT"
- Calls always get dropped at around 10 mins when SIP session timer is used

Build Series v2.7.62

Enhancements:
- Add routing script option to control if diversion is allowed
- Add UCT/CDR trace about the route that was chosen for a call
- Allow configuring SCTP min and max RTO
- Add SIP Call-ID attribute to text and Radius CDR

- Allow overriding default TDM companding (aLaw/uLaw) for E1/T1/J1 line services

Bug Fixes:
- v2.7.39 Circuits are remaining as "resetting" after changing M3UA SCTP configuration
- Min RTO and max RTO working when no SCTP ACK message is received
- Error in SmartMedia log file on units with DSPs, but less than 512 channels licensed
- Should not remove "remotely blocked" state upon incoming ISUP test call
- Should remove COT request from ISUP IAM when forwarded to SIP-I

Build Series v2.7.61

Bug Fixes:
- ISUP crashes due to SS7 Error double free
- Wrong Tooltip on ToneString in Test Call
- ISUP: the packet reassembly is not working (SGM, INF)
- SN10K responded 401 without digest header caused fax failure
- No snmp trap when link up
- NAP status Unavailable circuits do not show as unavailable

Build Series v2.7.60

Bug Fixes:
- SNMP trap sent, no log in SmartMedia system manager
- M3UA: Too much delay between INIT, ASPUP and ASPAC

Build Series v2.7.53

Bug Fixes:
- SmartMedia fails to re-sync playing file
- Unable to allocate T.38 resources

Build Series v2.7.52

Bug Fixes:
- Codec selection may revert after a RE-INVITE if original codec is still present in SDP
- Log file rotation and performance problem

Build Series v2.7.51

Enhancements:
- Support a new STM-1 to E1 Channel Mapping: Alcatel-Lucent

Bug Fixes:
- Re-invite callflow timing makes VoIP to detect reception of invalid codec and triggers channel protection
- ISUP interface status now button : Refresh Failed, Server is not responding

Build Series v2.7.50

Bug Fixes:
- Remove 0 from the IP address in the Division header
- SIP-I to SIP-I call modifies the NOA
- Digit collection stops working after playing files, on units without DSPs
- ISUP: CIC=0 is not allowed in configuration
- System flags DB on primary server as 'invalid'
- LegId can be reused before expected period
- CDR with Incoming call drop UNSPECIFIED

Build Series v2.7.49

Bug Fixes:
- SIGTRAN: Support traffic handling mode: override mode
- No statistics information values sent to softswitch
- CallTrace shows SN10K sent 183 while SIP capture show SN10K sent 180
- CAS R1 circuits unavailable after saving CAS stack configuration

Build Series v2.7.48

Bug Fixes:
- Early V34 Negotiation failing - V8 to V34HDX fallback loop after JM
- SmartMedia error upon SIP call with no matching NAP

Build Series v2.7.47

Enhancements:
- SIGTRAN: Support traffic handling mode: override mode
- ISUP SBM non-continuous CICs assignation
- Support Japan NTT GRS and GRA ISUP Messages for a Group of 32 Circuits

Build Series v2.7.46

Enhancements:
- Improved maximum SIP call rate in some cases

Bug Fixes:
- SetCustomFieldToTransmit : failed, cannot set more than one custom field per message for now
- RADIUS: Race condition in authorization module
- Graceful streamserver shutdown
- Create_multiple CIC allocation does not follow the selected order

Build Series v2.7.45

Enhancements:
- New global call statistics available in the web portal:
- Total calls (in/out)
- Current calls (in/out)
- Highest calls (in/out)

- Call rates (in/out)
- Highest call rates (in/out)

Build Series v2.7.44

Bug Fixes:
- V.34 False ToneA detection during INFO0a reception
- V.34 Rate renegotiation not working without training
- V34 PPS-*** repeated at 21600

Build Series v2.7.43

Enhancements:
- Options to eliminate the first "click" noise received from outgoing TDM leg upon alert with early media.
- Support Japan NTT GRS and GRA ISUP Messages for a Group of 32 Circuits
- Add Generic Number in Radius Accounting
- Support H.248 Local NAT traversing for RTP streams
- New option to configure limit CDR files segment size (default 100MB)

Bug Fixes:
- After stop of all traffic MEGACO shows as "Current number of eph term" 996
- Sip memory pool extent leading to restart of the SN10k
- Activation of hairpin call with ADD/SUBTRACT request fails on MGW side
- Failed to load routes from more than 256 different source NAPs
- ISUP crashes due to SS7 Error double free
- Outgoing SS7 calls are refused if calling numbers are longer than 20 digits
- Log rotation does not rotate unless having something to write in file

Build Series v2.7.42

New features:
- New dial tone generator option with CASR1 scripts
- New CASR1 scripts for Loopstart (FXO and FXS)

Bug Fixes:
- H.248 fax call fails when call discrimination package with call type is used

Build Series v2.7.41

Bug Fixes:
- Reception of CAS bits in T1 AMI encoding not working
- H.248 fixed termination's Start suffix is limited to 255

Build Series v2.7.40

Bug Fixes:
- After stop of all traffic MEGACO shows as "Current number of eph term" 996
- H.248 interface did not resume after restart of the Softswitch

- RADIUS: Gateway race condition when leg is freed before authorization is completed
- SIP-I ISUP release cause on Decline doesn't match with Q850 header

Build Series v2.7.39

Enhancements:
- Support ISUP NTT variant

Bug Fixes:
- Activation of hairpin call with ADD/SUBTRACT request fails on MGW side

Build Series v2.7.38

Bug Fixes:
- Stuck CASR1 timeslots if remote hangup while SN10k is sending a Wink

Build Series v2.7.37

Bug Fixes:
- Support for CAS R2 variable called number lengths on backward side
- hairpin call with H.248 MOVE request will be cancelled after RTP timeout

Build Series v2.7.35

Bug Fixes:
- Ignore "Forward SIP hold type " doesn't work

Build Series v2.7.34

Enhancements:
- Support R1 CAS (Q.310-Q.332) (Wink Start variant only for now)

Build Series v2.7.32

Bug Fixes:
- Isup standby failed to activate after reboot

Build Series v2.7.31

Enhancements:
- RADIUS accounting and authorization association
- CAS R2 metering on incoming calls

Build Series v2.7.30

Enhancements:
- Support for CAS R2 variable called number lengths on backward side
- VoIP bonding support

Bug Fixes:

- Occasional error incorrectly reported when applying system configuration
- Can't configure SIP refusal code for a refused T.38 REINVITE to use 406 instead of 488
- Grow max_call_duration maximum value to support over 30 days

Build Series v2.7.29

Enhancements:
- Support Bell2225Hz signal to trigger switch to fax/modem passthrough
- Give access to Redirecting indicator in CTBCMCExtraNumber class

Bug Fixes:
- H.248 interface did not resume after restart of the Softswitch
- H248 communication stops after "MGC Impending Failure"
- SS7 Raw data is passed as binhex to routing script, rather than as raw binary
- Wrong SIP current active call count leading to congestion
- Framing issues with STM1->VC4->VC12
- Wrong CSeq value in 200 OK response for OPTIONS request

Build Series v2.7.28

Bug Fixes:
- H248 stack doesn't recognize SDP attribute SilenceSupp:off - - - - to switch in passthrough mode
- Vlan configuration requires "use virtual IP address" setting in H.248

Previous releases

Enhancements:
- Increase the max amount of NAPs
- Application transport parameter removal from IAM
- APM message support on ITU97
- Support Radius authentication and authorization (AAA)
- Additional RADIUS accounting attributes
- New Radius CDR parameters - timestamps/diversion
- Add routing script option to control if diversion is allowed
- Support new scriptable IE in the SIP-I SS7 payload
- Add a variable to include custom attributes for routeset in the CDR
- Add Authorization in Web Portal
- Support SN10K behind a NAT for SIP
- DSP/VoIP resource usage Information/Alarms
- Attach an automatic logging for all Save, Create, Delete operations on Web Portal
- Added support for part of the SPIROU specification
- Lawful Interception, based on ETSI ES 201 671 V2.1.1
- Adjusted global capacity #defines to support 4xSTM1/OC3 mezzanines
- Support for Stats History
- Fixes and enhancements for beta support for audio mixers
- Support for audio mixers API (requires SN10k unit with DSPs).

This can be used for conferencing:
> background music playback
> recording of both legs of a call together
> Support for audio forking (half-dupex, mulitple join per leg/call flow).
No DSP required on SN10k unit.
- Support for audio forkingAPI (requires SN10k unit, no DSPs required).
This can be used for:
> Lawful intercept
> Broadcasting a speaker to many listeners
- Make test call from the Web Portal when using the Gateway application
- View graphical history of system statistics from the Web Portal
- Support for NSE protocol for fax/modem passthrough auto-switching

WWW

Please refer to the following online resources:

- Software Configuration Guide SmartMedia:
  http://www.patton.com/manuals/SmartMedia-scg.pdf
- SmartMedia Configuration Library:
  http://www.patton.com/support/kb.asp?cat=114
- SmartMedia FAQ:
  http://www.patton.com/support/kb.asp?cat=75

How to Upgrade

Please see http://www.patton.com/support/kb_art.asp?art=326.