ZyXEL

Firmware Release Note

Prestige P2002

Standard version

Release 3.60(MD.4)c0

Date:	**August 22, 2005**
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Supported Platforms:

ZyXEL Prestige P2002

Versions:

Bootbase Version: V1.08 | 04/15/2005 17:12:54
ZyNOS Version: V3.60(MD.4) | 08/22/2005 14:00:16

Notes:

1. Management port is 192.168.5.1.
2. After 3.60(MD.0)a7, all the firmware can’t use/download on P2002 C2 hardware.
3. Bootbase version MD1.01/1.02 supports AMD Flash only.
4. Bootbase version MD1.03/1.04/1.05 supports Intel Flash only.
5. Support C3-2 hardware and after, please use f/w after 3.60(MD.0)b4 and Bootbase 1.05. (Intel flash)
6. Bootbase version MD1.06 supports both Flash type (Intel and AMD).
7. Add “Multi-Upload firmware” feature in bootext for OBM version.
8. In log information on eWC, if you set the SIP number more than 28 characters, we use “…” to represent the remainder.
9. Bootbase version MD1.08 supports AMD and Intel Flash.
10. New DSP, Version : Rel 9.1.500.8, Build g26
11. Support Multiboot client V2

Features:

Modifications in V 3.60(MD.4)c0 | August 22, 2005

1 **[Change to FCS Version]**

Modifications in V 3.60(MD.4)b3 | August 12, 2005

2 **[Bug fixed]**

2.1 (SPRID:050804190)DUT send out noise seriously.

1. Change MRD country code to others than "USA" (0xFF)
2. Reset DUT to default.
3. Register to SIP server and make call. DUT send out noise seriously.

Or

1. Change MRD country code to others than "USA" (0xFF)
2. Reset DUT to default.
3. dial "****" to enter IVR. DUT send out noise seriously.

2.2 STUN always fails behind NAT (e.g. ZyWALL 5)

Modifications in V 3.60(MD.4)b2 | August 03, 2005

1 **[Bug fixed]**

1.1 (SPRID:050713647)T.38 interoperability fail with Cisco 2600 T.38 gateway.

1. Send T.38 Fax from DUT to Cisco T.38 gateway, the fax operation always failed.
2. The T.30 speed negotation fail between DUT and Cisco T.38 gateway in T.30 DCS and FTT stage.

1.2 (SPRID:050713648)Country code mechanism issue.

1. Change country code from "Default" to "USA", the codec, flash interval, CallerID will not be changed.
2. Change country code from "Default" to "USA" and set SIP setting to change codec to "G.711u-law". However, change back country code back "Default" or others country code, the codec can't be change back to "G.711a-law" unless using CI command.

1.3 (SPRID:050715765)Download v3.60(MD.4)b1 firmware in two of P2002 hardware occur below error messages.

"Copyright (c) 1994 - 2005 ZyXEL Communications Corp.
initialize ch =0, ethernet address: 00:13:49:8E:23:66
initialize ch =1, ethernet address: 00:13:49:8E:23:66
VC5402 Init...OK

DSPCheckCodecAvail: Can't Find Target Codec
IVR User is NOT Usable
DSPCheckCodecAvail: Can't Find Target Codec
IVR User is NOT Usable
DSPCheckCodecAvail: Can't Find Target Codec
IVR User is NOT Usable

1.4 (SPRID:050715767)After switch first call and second call several times by using "USA" dial mode, below symptoms occurs.

1. Make conference failed and can't hear voice on mixer.
2. After hear no voice, on-hook phone. Off-hook phone again but hear no dial tone and Phone LED is not turn on.

Procedure:

1. DUT-A (SIP account 1 and SIP account 2), DUT-B (SIP account 3)
2. DUT-B SIP account 3 call DUT-A SIP account 1.
3. DUT-B SIP account 3 call DUT-A SIP account 2.
4. Press "FLASH" on DUT-B several times to switch first/second call/conference.
5. After switch few times, the above symptoms occur.

1.5 (SPRID:050715771)MOH function will work only if codec selection is "G.729>G.711" or "G.729" only. However, early media function works for all codec preference.

1.6 (SPRID:050715774)By selecting Czech/ Hungary/ Poland/ Slovakia country code in Phone/Common setting. DUT can't generate RINIGNG tone during VoIP call was incoming.
1.7 (SPRID:050715811)Stun parameter (SIP1 or SIP2) can't be updated through Auto-Provision. Through SPTGEN(rom-t) is OK.
1.8 (SPRID:050715814)Auto firmware upgrade still play out "new firmware notification voice" even if the current firmware version is same with Target firmware version in auto-provision file.

2 [Enhancement]

2.1 Add information/Note to show that MOH function will work only if codec selection is "G.729>G.711" or "G.729" only in WEB GUI and Help page
2.2 Add log "IVR User is Not Usable" when finding a unusable IVR user during device booting up. (This case related to problem SPRID:050715765).

Modifications in V 3.60(MD.4)b1 | July 08, 2005

1 [Enhancement]

1.1 Add Fax T.38
1.2 Change DSP version to Rel 9.1.500.8, Build g26 for T.38 support
1.3 Add Fax T.38 / Pass Trough selections to AutoProvision and SPT Gen
1.4 Modification in Country Code Mechanism
1.5 Add IVR Selections (Early Media and Music On Hold) into Web GUI VoIP – Advance Settings
1.6 Add IVR Remaining Time to Web GUI Maintenance – Status
1.7 Add IVR Default for Early Media and Music On Hold

Modifications in V 3.60(MD.3)c0 | May 20, 2005

1 Update ROM file (Flash Max/Min Interval to 160/40)
2 Update ROM file (Phone 1 and Phone 2 only mapped to SIP 1)
3 Change to FCS version

Modifications in V 3.60(MD.3)b7 | May 05, 2005

1 [Bug fixed]

1.1 (050328361)There is no help page for call forward function and Call forward Help page is same with Phone book help page. Besides, there is no "MWI" and "Call forward" Help page in SIP advance setting.
1.2 (050428386)In eWC, System -> System Name is empty. (F/W (MD.3)b5, System Name is P2002)
1.3 (050428388)Please don't show surplus information in source and Destination of Logs. (Replace 0.0.0.0 with blanks)
1.4 (050428390)Press reset buttom about 1 sec that the system don't reboot
1.5 (050428461)The default setting "Call-Waiting Reject Time" = 20 (sec) (Now is 0)
1.6 (050428463) [Consultative Transfer when C don't response and A cancel] is Fail
  1. A goes offhook A receives dialtone
  2. A dials B B rings, A has ringback
  3. B goes offhook A and B converse
  4. A FLASHES A gets secondary DT, B gets MOH
  5. A dials C, *98# + C phone number, C rings, A has ringback, B has MOH
  6. A FLASHES C in idle state, A and B converse
  7. B goes onhook A in idle state
1.7 (050503050)The currently "CDR" logging seems useless. Please remove the selection from eWC log.
1.8 (050503054)Default ROM file contain incorrect setting. Default country code is "USA", but dialing mode is "Europe" and codec is "G.711A". It should be changed to match country code. (In default Rom file settings, Country Code set to Default and Dial Method to Europe Type)
1.9 (050503057)Call waiting function cause DUT occur abnormal sound.
  1. A call B. A and B in conversation.
  2. C call B. C and B in conversation.

3. B FLASH several times to switch call from C and A.
4. C and A sometimes occur abnormal sounds.

1.10 (050503061)RTCP should be removed while whole RTCP format in incorrect and not same with standard. (Timestamp)

2 [Enhancement]

2.1 Add Call Service Mode in WEB GUI; Phone → Common
2.2 Remove Password display in voice config signal display <index> and in voice config signal dump <index>
2.3 Update IVR to Standard version.
2.4 Add dsp initstatus to show DSP init result.
2.5 Reject telnet login before device init finish.
2.6 Apply password hidden in SPTGen.
2.7 Set hidden in some commands when engineering debug flag is turned off.
2.8 Change DSP to newer version : Rel 9.1.500.8
2.9 Change Bootbase to MD108.bm
2.10 Support Multiboot Client v2.
2.11 Support 43 country codes and 1 default.
2.12 Change sipserver.net in VoIP settings and add AutoProvision server address to 127.0.0.1
2.13 AutoProvision will not run when device reboot if activation flag is inactive. (AutoProvision will base on activation flag to run on each time device restart).
2.14 Device will automatically refresh and renew IP after WAN is up.
2.15 Remove Enable Dial-on-demand and Disable Dial-on-demand from SMT Menu 26 Schedule Setup.

Modifications in V 3.60(MD.3)b6 | April 25, 2005

1 [Enhancement]
1.1 PPP information send to syslog server.

2 [050413495][Bug fixed]
2.1 STUN Function Fail.

3 [050413496] [Bug fixed]
The attribute of rom-t file must be hidden.

4 [050413498][Bug fixed]
If use PPPoE client, when reboot device, the Management IP Address will be unable to use.

5 [Enhancement]
5.1 Support the outbound proxy at SPTGEN and Auto-Provision.

6 [Enhancement]
6.1 Support to enable and disable the SIP session timer.

7 [Enhancement]
7.1 Enlarge the length of the PPPoE/SIP username to 72. (SMT don’t support).

8 [Enhancement]
8.1 USA and Europe dial plan configured.

9 [Enhancement]
9.1 TISP used dial plan.

10 [Enhancement]
10.1 Early media & music on hold can configure and record. (only support by CI command).

Modifications in V 3.60(MD.3)b5 | April 11, 2005

1 [50328362][Bug fixed]
1.1 There is no SIP/RTP/FSM log and some unless log selection appears (etc. TCP reset, packet filter, CDR).

2 [50328363][Not Issue]
2.1 Domain name show invalid name "yyyyy" in default setting. Press "APPLY" button, there will have error message come out.

3 [50328364] Bug fixed]
3.1 Call forward + Call hold cause exception.
1. Set B forward to C.
2. A call B, C ring.
3. C offhook
4. C flashs,
5. C occur exception.

4 [50328365][Bug fixed]
4.1 Call waiting function cause excpetion.
1. A call B. AB in conversation.
2. C call A.
3. A flashs.
4. A occur exception.

5 [50328366][Bug fixed]
5.1 Congestion tone cause user can't dial peer to peer call. User can't dial peer to peer call after SIP unregistered.

6 [50328367][Bug fixed]
6.1 Consultation transfer always failed.
1. A call B.
2. B offhook, AB in conversation.
3. A dial *1* + C number.
4. C can't hear rining.

7 [50328368][Bug fixed]
7.1 DUT send out DNS query between 100tring and 180rining.

8 [50328370][Bug fixed]
8.1 Before using Early media function, user need to enter IVR and clear all memory. Otherwise, the early media function will cause exception.

9 [50328371][Bug fixed]
9.1 Get Rom-t from DUT cause DUT exception.
1. FTP to DUT.
2. Get Rom-t from DUT.
3. DUT occur exception.

10 [50328372][Not Issue]
10.1 Min-SE parameter doesn't work.
1. Set Min-SE value in A which larger than B's session timer.
2. B call A.
3. A receive Min-SE but doesn't send Invite with new timer.

11 [50329408][Not Issue]
11.1 Call waiting indication only continue for two sounds and Calling party doesn't receive busy tone. Call waiting indication need to continue if calling party is not hang-up. Besides, after CWI sound stops. Calling party didn't receive any sound.

12 [50329409][Bug fixed]
12.1 Fill out all characters in Call forward table then eWC show error page.

13 [50329410][Bug fixed]
13.1 For "accept" and "block" method in advance call forward table, forward number should be don't need to input. Currently, all field need to be input, otherwise the rule can't be saved.

14 [50329411][Bug fixed]
14.1 "No answer" function in advance call forward table doesn't have timeout setting.

15 [50329412][Not Issue]
15.1 VoIP call POST dialing delay increase gradually. VoIP call delay become more and more late in every VoIP call.

16 [50329413][Not Issue]
16.1 Auto-Firmware upgrade will be failed if auto-provision name is "360MD3b5.bin", and the FTP server's file name is "360MD3B5.bin". Although DUT can locate the file but upgrade will be failed.

17 [50329415][Not Issue]

17.1 Speed dial can't work if Auto-Firmware upgrade function is enabled.

18 [50329417][Bug fixed]

18.1 Call feature cause DUT occur exception randomly.

19 [50330453][Bug fixed]

19.1 After unplug WAN connection during VoIP conversation, DUT can't send to Invite or receive VoIP call.
1. Make VoIP call.
2. Cut off DUT's WAN connection.
3. DUT sometimes can't make or receive VoIP call again.

20 [50330454][Not Issue]

20.1 Immediate dial can't work. Enable immediate dial function and dial VoIP number plus pond(#) sign, the VoIP call is still delay few seconds.

21 [50330455][Bug fixed]

21.1 PPP information didn't send to syslog server.

22 [50330456][Bug fixed]

22.1 Sending log to email function doesn't work by either clicking "send email now" on eWC or by schedule.

23 [50330457][Bug fixed]

23.1 Log setting help page appears "ZyAIR" wording.

24 [50330458][Not Issue]

24.1 Long term overnight VoIP call failed.
1. Make VoIP call.
2. Wait overnight.
3. VoIP call was disconnected after overnight test.

25 [Enhancement]

25.1 Support Rom-D.

26 [Enhancement]

26.1 Support DTMF Call ID.

27 [Enhancement]

27.1 Support 3 way conference.

Modifications in V 3.60(MD.3)b4 | March 18, 2005

1 [Enhancement]

1.1 **Support DTMF Relay: RFC2833 / SIP INFO / SIP-INFO / SIP-INFO-Link_RFC2833**
1.2 **Support FAX Relay**
1.3 **Support VLAN Tag / DiffServ**
1.4 **Support Point to Point Call**
1.5 **Support STUN**
1.6 **Support Fake IP**
1.7 **Support Sptgen**
1.8 **Support Auto Provision (support TFTP and HTTP)**
1.9 **Support Speed Dial phonebook**
1.10 **Support On-hold & Un-hold**
1.11 **Support Second-Call**
1.12 **Support Call-Waiting**
1.13 **Support Call-Transfer (Blind Transfer / Consultative Transfer)**
1.14 **Support MWI**
1.15 **Support congestion tone**
1.16 **Support IVR**
1.17 **Support Early Media when on-hold (the primary codec should be G.729)**
1.18 **Auto firmware Update**

Modifications in V 3.60(MD.3)b2 | December 16, 2004

1 **[Bug fixed]**
 1.1 DNS query failed after auto-provision change the tftp server address to FQDN.
2 **[Enhancement]**
 2.1 Support 5 REN.

Modifications in V 3.60(MD.3)b1 | November 24, 2004

1 **[Bug fixed]**
 1.1 Callee side fail to hangup peer to peer call on ATA.
 1.2 Auto-Provision will disconnect PPPoE or voice connection.
2 **[Enhancement]**
 2.1 Support country code (in voip->common) mapping to tone.

Modifications in V 3.60(MD.2)c0 | November 9, 2004

1 Change ZyNOS Version from 3.60(MD.2)b7 to 3.60(MD.2)c0

Modifications in V 3.60(MD.2)b7 | November 2, 2004

1 **[Bug fixed]**
 1.1 (177)ATA shows menu 98.5, 98.6, 98.7.1, 98.7.2 in incorrect format when getting ATA configuration from SPTGEN.
2 **[Enhancement]**.
 No enhancement item.

Modifications in V 3.60(MD.2)b6 | November 1, 2004

1 **[Bug fixed]**
 1.1 (113) Sometimes MGMT port can't be Ping or Managed after boot up ATA without connect LAN port. After reboot, it works.
 1.2 ATA occurs console hang up or reboot during stress testing.
 1.3 180 ringing delay sending out.
 1.4 System crashes if doing Auto-Provision.
 1.5 When Off Hook at first ting, users hear Caller-ID sound. Caller-ID can't display on phone 1.
2 **[Enhancement]**.
 2.1 Add rom-d function for Bluecom.

Modifications in V 3.60(MD.2)b5 | October 12, 2004

1 **[Bug fixed]**
 1.1 (178)ATA can't accept "OPTION" packet come from Asterisk server.
 1.2 (181)ATA could not get SIP account 2 RTP port info from Auto-Provision.
 1.3 (182) When clicking restart button on ATA, browser show "your browser can't handle the script……."
 1.4 TFTP timeout time always delay 10sec to be executed.
2 **[Enhancement]**.
 2.1 Add Autoprovision key word display, use "voice autopro dbdisplay" to get the information

Modifications in V 3.60(MD.2)b4 | October 8, 2004

1 **[Bug fixed]**
 1.1 (162) Phone book entry "Delete" function doesn't work by using MAC IE5.2 browser.
 1.2 (163) Configuration File Restore function failed by using MAC IE5.2 browser.
 1.3 (178) ATA can't accept "OPTION" packet come from Asterisk server.
 1.4 (179) There is no help page for VoIP setting common page.
 1.5 (180) There is no Fake IP/ Outbound Proxy /NAT Keep alive on SIP advance setting help page.
 1.6 (181) ATA could not get SIP account 2 RTP port info from Auto-Provision.

1.7 Auto-Provision function show inconsistence with eWC SIP parameter value range. Currently Auto-Provision accept old eWC range and can't consistence with eWC acceptable value range.
1.8 By using Windows Messenger or XTEN to call ATA, ATA will not send out "BYE" when Phone is On-Hook.
1.9 ATA use incorrect DNS server to send out DNS query.

2 [Enhancement]
No enhancement item.

Modifications in V 3.60(MD.2)b3 | October 6, 2004

1 [Bug fixed]
1.1 (176) Clear button doesn't work in Phone Book page on eWC.
1.2 (177)Set "RTP from" port as "1024" and "RTP to" port as "1025".
1.3 (178)ATA can't accept "OPTION" packet come from Asterisk server.

2 [Enhancement]
2.1 Remove ZyNOS version in login page.
2.2 Move VoIP->Common to Phone->Common.

Modifications in V 3.60(MD.2)b2 | September 30, 2004

1 [Bug fixed]
1.1 (168) PPPoE/DHCP mode DNS query doesn't work.
1.2 (169) eWC shows improper "Set Pound" wording.
1.3 (171) ATA shows "Error! RTP start port can't large than end port" when user input same value in RTP from and RTP to port.
1.4 (172) ATA allow different port range in RTP port setting and SIP port setting.
1.5 (174) NAT Keep alive value has no default value and allow range specified on eWC.
1.6 (173) Callee side can't terminate VoIP call on ATA.

2 [Enhancement]
2.1 Show ZyNOS version in login page.

Modifications in V 3.60(MD.2)b1 | September 27, 2004

1 [Bug fixed]
1.1 (40) Exception occurs after setting SIP number to "127" character.
1.2 (45) There is no MGMT, ethernet Port and VoIP status Help page in Maintance HELP page
1.3 (47) Exception occurs when DUT1's compression type change to G.729 > G.711, DUT1 dial to DUT2, DUT2 off hook and on hook immediately.
1.4 (32)Back to Factory Default would redirect WEB page to 192.168.1.1
1.5 (37)Non exist "show Statistics" function show in maintenance Help page
"Show Statistics - Click Show Statistics to see router performance statistics such as number of packets sent and number of packets received for each port."

2 [Enhancement]
2.1 Add “Multi-Upload firmware” feature in bootext for OBM version.
2.2 Add outbound proxy support.
2.3 Add ” NAT keep alive” function.
2.4 Add phone country code selection.
2.5 Free the RTP port restriction (1025~65535).
2.6 Add Fake WAN address.
2.7 Allow for user to use Peer-to-Peer Call.

Modifications in V 3.60(MD.1)c0 | August 9, 2004

1 [Bug fixed]

No newer bug be fixed.

2 [Enhancement]

No enhancement item.

Modifications in V 3.60(MD.1)b4 | August 6, 2004

1 [Bug fixed]

1.1[Help Page]

1. No STUN description on help page.
2. No DTMF function description on help page.
3. No VAD function description on help page.
4. No Dialing Interval description on help page.

1.2 When the SIP 1 is not active, SIP 2 can not be register or unregister.

2 [Enhancement]

No enhancement item.

Modifications in V 3.60(MD.1)b3 | August 6, 2004

1 [Bug fixed]

1.1 Can’t get correct rtp port when STUN active.

1.2 Remove the help icon from Reg_Fail.html and Reg_Wait.html.

2 [Enhancement]

2.1 The rtpUptime value is zero and caused exception occur when open "Show Statistics" Page on eWC and Make VoIP Call.

Modifications in V 3.60(MD.1)b2 | August 5, 2004

1 [Bug fixed]

1.1 (100)"End of Event" doesn't carrier in last RFC2833 DTMF event's payload.

1.2 (106)ATA send DNS server query every 5 sec after used FQDN name in SIP server address on eWC.

1.3 (134) Online Configuration for PHONE setting on eWC. Volume setting/ VAD/ Echo setting doesn't work.

1.4 (146) Online Configuration for RTP port setting doesn't work

1.5 (147) If invalid(not exist) host was used in primary DNS, DNS query will not work.

1.6 (148) With "PPPoE+Static IP", MGMT port show "0.0.0.0" in maintenance page.

1.7 (149)ATA sometimes doesn't generate Busy tone when receiving 4xx response from proxy server.

1.8 (152) Sometimes ATA hang before "VC-5402 Init" print out. Frequency: two times

1.9 (154)Change SIP local port, eWC maintenance info will not be changed

1.10 (156)By using MAC OS IE5.2 to login ATA's eWC doesn't work.

1.11 (158)Volume control doesn't work. G.711/G.729 codec volume always keep in 14.7/14.3 dB

1.12 P2002 FCS sample (AE) will reboot when init SLIC in low temperature.

2 [Enhancement]

2.1 1. Support that online change STUN server IP

2. When change local IP, we need to unregister rtp port to stun.

3. Replace uint8 to uint16 for port parameter

2.2 Modify for debug tool, show all dlg & tx status.

2.3 Support RFC2833 Like SIP INFO.

Modifications in V 3.60(MD.1)b1 | July 28, 2004

1 [Bug fixed]

1.1 (50) By using G.729 codec to transfer FAX sometimes cause ATA occurs exception.
1.2 (86) ATA occur exception when running CERT SIP test.
1.3 (97) On line modify VoIP setting during VoIP call cause ATA setting change fail for more than 3 minute after hang up PHONE.
1. Access ATA through MGMT port.
2. Initiate VoIP call.
3. Modify VoIP setting (Caller ID, Phone 1/2 check). eWC will show setting failed due to VoIP call is running.
4. Hang up PHONE. VoIP setting will always failed for a period of time.
~~1.4 (106) ATA send DNS server query every 5 sec after used FQDN name in SIP server address on eWC.~~
1.5 (116)Configure ATA to use PPPoE+Static IP through MGMT port on eWC. Reboot ATA, then Static IP's subnet Mask or Default gateway will become "0.0.0.0".
1.6 (118) Phone Book] SIP number's MAX characters is 31. It's too short for most SIP URI case. Suggest: 64 characters. Now is 127 character.
1.7 (123) After restart ATA and the time setting become default value.
1. Set Time protocol to NONE.
2. Set Current Time.
3. Restart ATA
1.8 (124) When the sip is registered ,then disconnect the WAN cable,the PWR/VoIP light is still show sip connect. (show orange color not green)
1.9 (128) With default Volume setting, Echo and background noise are noticeable. Expect: Lower default volume setting.
1.10 (129) Log show incorrect "SIP registration" log entry.
1. Press "unregistered" on eWC Maintenance page.
2. Check Log. Log show two entry, one is "SIP unregisteration success". One is "SIP registration success. But the registration is not show success on Maintenance page.
1.11 (131) Phone Book show each empty as "¡K". Expect: change to "NULL" character
1.12 (132) SIP Password 19 character is too short for some beta tester's ISP.
~~1.13 (134)Online Configuration for PHONE setting on eWC. Volume setting/ VAD/ Echo setting doesn't work.~~
1.14 (135) VoIP call GOS(Grade of Loss) issue by measuring with Radcom
1.15 (136) When register fail and eWC show " The registration has failed. Please return to the previous page. Click Help for more information. Click Help page But show no help page.
1.16 (137) Key in wrong SIP username and Password then make register failed. Key in correct one and press register on eWC. The registeration will always failed.
1.17 (141) Disable SIP 1 account and register SIP 1 account on eWC. Then enable SIP 1 account again. SIP 1 will not be able to register unless reboot ATA.
1.18 (142) By default ROM file, Ethernet Subnet/Default Gateway field on eWC is not grey out.
1.19 (143) [Help Page] No SIP/RTP priority setting and Dialing interval description in help page.
1.20 (144) Incorrect RTP port was used "50004". eWC setting From RTP port "50000"
1.21 IAD show "VC5402 init error" in console when IAD was restarted during VoIP communication.
1. IAD <--> LAN <--> IAD
2. Initiate VoIP call
3. Restart IAD through eWC.
4. IAD sometimes generate noise sound when "VC5402 init error" shown.
5. IAD needs to be restarted again in order to recover.
1.22 After turn on VAD, the VoIP communication appear stop for a short period of time then continue. Default Volume setting.

2 [Enhancement]

2.1 Update Bootbase to v1.06 for the dual flash support. Now v1.06 can support Intel/AMD Flash type.
2.2 Auto-provision/Auto-firmware upgrade by TFTP

Modifications in V 3.60(MD.0)b7 | July 15, 2004

1 [Bug fixed]

2.3 G.711u/G.729 VoIP conversation delay variance => 10ms+/-6, Expect => 10ms+/-1. Have been fixed in previous version.

2.4 [Help Page on eWC]

2.4.1 Incorrect System DNS server on System/General Help page.

2.4.2 No Ethernet and Log help page.

2.4.3 Statistics help page appear on help page but no Statistics function appear.

2.5 By default rom file, eWC show SIP account 2 as registration failed. Expect: Not register

2.6 ATA should play "busy tone" when ATA receive SIP 488 not acceptable packet.1. Set two ATA with different code.2. Initiate VoIP call.3. ATA will receive 488 not acceptable but doesn't generate busy tone. Have been fixed in previous version.

2.7 Set "space" in SIP number on Speed dial Phone book, but the entry could not be used to dial out. Not a problem.

2.8 WAN to LAN VoIP call drop after 3 minutes in STUN scenario.

2.8.1 Topology:P2002(Caller) <--- ZyWALL --> P2002 (Callee) with STUN enabled.

2.8.2 Initiate VoIP call from Caller to Callee.

2.8.3 Wait 3 minutes and the VoIP call was dropped.

2.9 No STUN keep-alive interval function implemented. The STUN keep-alive interval value will be changed automatically.

2.10 [Phone Book] SIP number's MAX character is 31. It's too short for most SIP URI case.

2.10.1 Suggest: 64 characters.

2.10.2 Have been update eWC to 128 characters.

2.11 [Help Page on eWC] Chang the all words "ZyAIR" to "ATA"

2.12 [Help Page on eWC] No PHONE BOOK help page.

2.13 SIP Password 19 character is too short for some beta tester's ISP. Have been extended to 96 characters.

2.14 Online Configuration for PHONE setting->Out-Going Call mapping to SIP only on eWC

Modifications in V 3.60(MD.0)b6 | July 11, 2004

3 [Bug fixed]

3.1 (79)ATA occur exception when suddently off-on-hook

1. Caller <--> ATA(1) <--Internet--> ATA(2) <--> Callee
   Caller side make VoIP call to Callee side.
2. Callee side Rining.
3. Make Callee side Off hook and on hook quickly.
4. Callee side ATA occur exception.

3.2 (83)Create Many call by using Prolab at the same time, ATA will occur exception.

3.3 (103)[eWC] By default rom file, eWC show SIP account 2 as registration failed. Expect: Not register.

3.4 (104)[eWC] Two G.168 active selections.

3.5 (105)re-Invite by 302_move_temporary, callee side can't hang up.

Topology: P2002 (Caller) <--> LAN <--> P2002 (redirect)

1. Caller call Callee and proxy redirect to P2002 (redirect).
2. P2002 (redirect) answers the phone then hangs up.
3. P2002 (Caller) would not hear BUSY tone.

3.6 (108)ATA occur exception after Overnight VoIP reliability test.

1. P2002 <--> LAN <--> P2002
2. Initiate dual channel VoIP call.
3. Run for overnight test.

3.7 (109)[eWC] Phone book -> SIP number, Name, Non-Proxy doesn't start from first character when user inputted on eWC.

3.8 (111)RTP/SIP ToS range incorrectly limited to ""0~5"" on CI command. Expect: 0~7.

3.9 (117)[eWC][Phone Book] Add two speed dial entries with MAX characters in SIP number and name field. The first entry's name field will include and overlap with second entry's name.

3.10 (119)[eWC][Phone Book] Edit a "use proxy" speed dial entry, the entry will show as "non-proxy" type.

3.11 (120)[eWC][Phone Book] Fill in MAX characters in name field and Add the entry. Then edit the entry and check "Name" field. The last character will show one more "space" character.

3.12 (121)[eWC][Phone Book] Fill in MAX characters in SIP number, name and input non-proxy IP. Then edit the entry. Problem: The ""Non-proxy IP"" will show in ""Name"" field.

3.13 (122)[eWC][Phone Book] Fill in MAX characters in SIP number, name and input non-proxy IP in entry "#10". The entry will can't be deleted unless press "Clear Phone Book" button.

3.14 (128)ATA doesn't generate sip log and rtp log

3.15 (128)With default Volume setting, Echo and background noise are noticeable. Expect: Lower default volume setting."

4 [Enhancement]

4.1 (50)FAX Relay. By using G.729 codec to transfer FAX always fail. Default codec is G.729 -> G.711.

4.2 (82)Transfer FAX in one channel will affect another channel to drop VoIP communication when FAX tranfer is completed.

Procedure:
1. Switch both channels to G.711.
2. Make VoIP communication in channel 1.
3. Transfer FAX (5 pages) in another channel.
4. When FAX machine is completed, sometimes the VoIP communication was also dropped.

4.3 SIP number's MAX character is 31. It's too short for most SIP URI case. Extend to 95 characters. (CI command only, eWC will be next version.)

4.4 Add Country option in VOIP->Common group.

4.5 Support privacy call (including RFC3325 and call draft). Add CI command let RFC3325 and call draft adjustable.

4.6 Add Ring Timeout Timer = 180 seconds. Ring timer would fire once ringing. If timeout then terminate itself and send 486(Request Terminated) out to terminate the caller.

4.7 Don't care SIP Packet when all headers are un-known.

4.8 Support on-line configuration for phone. (CI command)

5 [VoIP SIP Interoperability]

5.1 Can't interoperate with telia softphone through telia sip server which support TCP & UDP.

5.2 IPTEL SIP server.

Modifications in V 3.60(MD.0)b5 | July 1, 2004

1 [Bug fixed]

1.1 ATA occur exception when receive 302 – Move Temporary.

1.2 ATA occur exception when receiving SIP REFER packet.

1.3 [eWC] ATA occur exception when open eWC->Maintenance page will cause system exception.

1.4 ATA should not send out SIP invite with itself SIP Phone number.

1.5 [eWC] VoIP -> SIP server address / Register server address should accept FQDN format address.

1.6 Wrong DTFM signaling bits is carried in INFO packets. Only 1~6 work correct.

1.7 [eWC] Default URL type should be "SIP" not "TEL". After enter "SIP advance setting page", the URL setting was changed to URL:TEL and can't be changed back unless changed in CI command. It caused VoIP call always failed.

1.8 [eWC] Online configuration didn't work for SIP account configuration page setting.

1.9 [eWC] Change SIP local Port to port 5070, VoIP status on Maintenance menu still show UDP/5060 port

1.10 [eWC] Use eWC to configure device to PPPoE mode and reboot. SMT show "ip alias enif1 FAIL iface enif1:0 not available" and Management port failed to access.

1.11 [Romfile] Change default setting to "G.711 > G.729" from "G.729 > G.711", default codec will

use G.711.
1.12 [eWC] Hide VoIP->Advanced->T2 Timer for the T2 Timer can't configure problem.
1.13 When STUN active, RTP port is still local port not WAN port.
1.14 SIP1 call SIP2 in the same device, callee will auto send out BYE SIP message.
1.15 Different Codec can communicate. Test result: ATA1 set G.711 only. ATA2 set G.729. ATA1 can make call each other.
1.16 Update Bootbase to version 1.05 to the default WP OFF on Intel flash.

2 [Enhancement]
2.1 [eWC] Add Active option to VOIP->SIP page.
2.2 Pass CERT test.
2.3 Play out DTMF tone after receive "SIP INFO" packet.
2.4 Support SIP ONHOLD/UNHOLD:
2.5 Set destination IP to 0.0.0.0 for RTP packet when receiving SIP ONHOLD request, and set back to non-zero IP when receiving SIP UNHOLD request.
2.6 Free RTP packets if the destination IP is zero.
2.7 Change volume control by SLIC to DSP. The range will change to -14 to 14 from 1 to 5. In eWC, 1 means -14 and 5 means 14.
2.8 Support different TOS setting in RTP and SIP packet. Now only CI command supported. eWC will setup both to the same setting.
2.9 Support Non/Force proxy call.
2.10 Support RemovePound: Press '#' when inputting dial digit number will make device to make call immediately.
2.11 [eWC] Add STUN setup field – Active, Server Address and Server Port in VOIP->Advanced page.
2.12 [eWC] Add DTMF Type setup field in VOIP->Advanced page.
2.13 Add Phone Book->Speed Dial page for speed dial and non/force proxy call.

3 [VoIP Interoperability]
3.1 Windows Messenger.
3.2 MWI
3.3 ASTERISK
3.4 GRANDSTREAM

Modifications in V 3.60(MD.0)b4 | June 4, 2004

1 [Bug fixed]
1.1 System logs function doesn't work. (24)
1.2 Change Phone to 10/20 pulse dialing mode cause device exception. (48)
1.3 Reboot button on eWC sometimes doesn't work.(PPPoE mode) (75)
1.4 ATA carry incorrect TOS value. (39)
1.5 Sometime ATA doesn't register automatically after boot up. Register manually will be successfully. Happen 2 times / 5 Try. (68)
1.6 Dialing interval doesn't work. By default: dialing interval is 3 sec. After 3 seconds timeout, FXS port should send out. (9)
1.7 ATA occur exception when using dual channel with long VoIP conversation (two days). (84)
1.8 [eWC] Primary and Secondary DNS can't be saved on Web. (26)
1.9 [eWC] PPPoE mode unstable and sometimes PPPoE connection drop abnormally. Use eWC to configure device to PPPoE mode will make VOIP call fail after idle timeout. Force eWC configure device to PPPoE mode with Nail-up connection. (91)
1.10 Change ATA default session expire time to 180 sec from 1800. (33)
1.11 Default RTP from port should be even value. Default value is 49153 on eWC, but ATA use port 49154 as the first channel. (34)
1.12 [eWC] Fix eWC->MAINTENANCE page wrong string display.
1.13 [eWC] Change SIP local Port to port 5070, VoIP status on Maintenance menu still show UDP/5060 port. (81)

1.14 [eWC] Phone 2 voice volume control can't be changed on eWC. (43)
1.15 [eWC] Voice volume control function doesn't work. (44)
1.16 [eWC] Phone 2 G.168 function can’t be actived on eWC. (45)
1.17 [eWC] VoIP Setting -> Apply to Phone 1,2 on eWC doesn’t work. (29)
1.18 [eWC] Didn’t check RTP end port must large than start port. (36)
1.19 [eWC] Didn’t check Voice VLAN ID in the range from 0 to 4024. (65)
1.20 [eWC] VoIP -> Advanced -> URL Type setting changed to SIP will save TEL.
1.21 [eWC] Phone -> Echo Cancellation enable will make it disable.

2 [Feature]

2.1 SDP in ACK.
2.2 STUN client.
2.3 Respond to OPTION request.
2.4 DTMP in INFO message.
2.5 RFC 2833 – RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals.
2.6 Caller ID support SIP phone call.
2.7 Device shouldn’t call itself.
2.8 [eWC] On line configuration.
2.8.1 User can’t save the change in SIP with conversation.
2.8.2 If the SIP is free, the configuration will be saved and active.
2.9 [eWC] Add Web VOIP help page.
2.10 [eWC] Add Outgoing Call use SIP mapping option.

3 [VoIP Interoperability] ATA occur exception and reboot when called by XTEN softphone.

Modifications in V 3.60(MD.0)b3 | May 28, 2004

1. **This version is for P2002 C3-2 sample production only.**
2. **Update Bootbase to 1.04 to meet C3-2 hardware with Intel flash.**

Modifications in V 3.60(MD.0)b2 | May 25, 2004

1. **[Bug FIXED]**
   Symptom: ATA carry incorrect TOS value.
   Procedure:
   1. Input TOS value "7" on eWC.
   2. Measure "0x07" value on RTP traffic in IPv4 TOS field.
   3. Expect: the value should be "0xE0" (BIN:11100000).
   Condition:
2. **[Bug FIXED]**
   Symptom: ATA occur exception when suddently off-on-hook
   1. Caller <--> ATA(1) <--Internet--> ATA(2) <--> Callee
      Caller side make VoIP call to Callee side.
   2. Callee side Rining.
   3. Make Callee side Off hook and on hook quickly.
   4. Callee side ATA occur exception.
   Condition:
3. **Disable SMT 3 – LAN Setup**
4. **TE test CI command support unregistered test.**
5. **Change Phone LED blinking to hardware trigger.**
6. **Bootbase 1.03 support Intel Flash but AMD Flash.**

Modifications in V 3.60(MD.0)b1 | May 4, 2004

1. **[Bug FIXED]**
   Symptom: ATA occur exception when switch ethernet setting from Static IP to Dynamic IP on SMT.
   Condition:

2. **[Bug FIXED]**
Symptom: ATA set to Ethernet PPPoE mode cause continue Exception.
Condition:

3. **[Bug FIXED]**
Symptom: Phone LED keep blinking even after Phone was on-hook. Sometimes Phone LED is not blinking when ringing.
Condition:

4. **[Bug FIXED]**
Symptom: Use Soft restart, ATA will not change PWR/VoIP LED status. VoIP LED always show registered after soft restart.
Condition:

5. **[Bug FIXED]**
Symptom: eWC show incorrect default setting.
Condition:

Modifications in V 3.60(MD.0)a7 | April 22, 2004

1. [ENHANCEMENT]
   Phone LED blinking when Ringing. The timer is 0.5s.
2. [ENHANCEMENT]
   FXS-SIP mapping. (Support CI Command only)
3. [ENHANCEMENT]
   VLAN Tag support for SIP/Voice packet.
4. [ENHANCEMENT]
   TOS setting SIP/Voice packet.
5. [ENHANCEMENT]
   SIP Register/Unregister. (support CI Command/eWC)
   [Bug FIXED]
   Symptom: Short-term noise, voice disappearance, and system exception during voice communication.
   Condition:
   1. Establish
   2 voice communication. Short-term noise occurs within 10 minutes. Voice disappearance occurs within an hour. System will crash for over-night voice communication

Modifications in V 3.60(MD.0)a6 | April 14, 2004

1. **[MODIFICATION]**
Update Model ID, after and from 3.60(MD.0)a6 can not use MD101.BM (Bootbase).

2. **[MODIFICATION]**
Cause the H/W design change, C3 and after H/W can not use F/W 3.60(MD.0)a1-a5 version (The phone can’t be used).

3. [Bug FIXED]
   Symptom: SLIC will cause system hang after phone ON/OFF hook continuous test. Single phone need long test time.
   Condition:

CI commands

Annex A CI Command List

Command Class List Table		
System Related Command	Exit Command	Device Related Command
Ethernet Related Command	POE Related Command	PPTP Related Command
Configuration Related Command	IP Related Command	PPP
Hdap	DSP	Voice

System Related Command

Command				Description
sys				
	adjtime			retrive date and time from Internet
	callhist			
		display		display call history
		remove	<index>	remove entry from call history
	countrycode		[countrycode]	set country code
	date		[year month date]	set/display date
	domainname			display domain name
	edit		<filename>	edit a text file
	extraphnum			maintain extra phone numbers for outcalls
		add	<set 1-3> <1st phone num> [2nd phone num]	add extra phone numbers
		display		display extra phone numbers
		node	<num>	set all extend phone number to remote node <num>
		remove	<set 1-3>	remove extra phone numbers

		reset		reset flag and mask
	feature			display feature bit
	hostname		[hostname]	display system hostname
	logs			
		category		
			access [0:none/1:log/2:alert/3:both]	record the access control logs
			attack [0:none/1:log/2:alert/3:both]	record and alert the firewall attack logs
			display	display the category setting
			error [0:none/1:log/2:alert/3:both]	record and alert the system error logs
			ipsec [0:none/1:log/2:alert/3:both]	record the access control logs
			ike [0:none/1:log/2:alert/3:both]	record the access control logs
			javablocked [0:none/1:log]	record the java etc. blocked logs
			mten [0:none/1:log]	record the system maintenance logs
			upnp [0:none/1:log]	record upnp logs
			urlblocked [0:none/1:log/2:alert/3:both]	record and alert the web blocked logs
			urlforward [0:none/1:log]	record web forward logs
		clear		clear log
		display	[access|attack|error|ipsec|ike|javablocked|mten|urlblocked|urlfor ward]	display all logs or specify category logs
		errlog		
			clear	display log error
			disp	clear log error
			online	turn on/off error log online display
		load		load the log setting buffer
		mail		
			alertAddr [mail address]	send alerts to this mail address
			display	display mail setting
			logAddr [mail address]	send logs to this mail address
			schedule display	display mail schedule
			schedule hour [0-23]	hour time to send the logs
			schedule minute [0-59]	minute time to send the logs
			schedule policy [0:full/1:hourly/2:daily/3:weekly/4:non	mail schedule policy

			[0:full/1:hourly/2:daily/3:weekly/4:none]	
			schedule week [0:sun/1:mon/2:tue/3:wed/4:thu/5:fri/6:sat]	weekly time to send the logs
			server [domainName/IP]	mail server to send the logs
			subject [mail subject]	mail subject
		save		save the log setting buffer
		syslog		
			active [0:no/1:yes]	active to enable unix syslog
			display	display syslog setting
			facility [Local ID(1-7)]	log the messages to different files
			server [domainName/IP]	syslog server to send the logs
	log			
		clear		clear log error
		disp		display log error
		online	[on|off]	turn on/off error log online display
		resolve		Resolve mail server and syslog server address
	mbuf			
		link	link	list system mbuf link
		pool	<id> [type]	list system mbuf pool
		status		display system mbuf status
		disp	<address>	display mbuf status
		cnt		
			disp	display system mbuf count
			clear	clear system mbuf count
		debug	[on|off]	
	pwderrtm		[minute]	Set or display the password error blocking timeout value.
	rn			
		load	<entry no.>	load remote node information
		disp	<entry no.>(0:working buffer)	display remote node information
		nat	<none|sua|full_feature>	config remote node nat

		nailup	<no|yes>	config remote node nailup
		mtu	<value>	set remote node mtu
		save	[entry no.]	save remote node information
	smt			not support in this product
	stdio		[minute]	change terminal timeout value
	time		[hour [min [sec]]]	display/set system time
	trcdisp			monitor packets
	trclog			
	trcpacket			
	syslog			
		server	[destIP]	set syslog server IP address
		facility	<FacilityNo>	set syslog facility
		type	[type]	set/display syslog type flag
		mode	[on|off]	set syslog mode
	version			display RAS code and driver version
	view		<filename>	view a text file
	wdog			
		switch	[on|off]	set on/off wdog
		cnt	[value]	display watchdog counts value: 0-34463
	romreset			restore default romfile
	server			
		access	<telnet|ftp|web|icmp|snmp|dns> <value>	set server access type
		load		load server information
		disp		display server information
		port	<telnet|ftp|web|snmp> <port>	set server port
		save		save server information
		secureip	<telnet|ftp|web|icmp|snmp|dns> <ip>	set server secure ip addr
	fwnotify			
		load		load fwnotify entry from spt
		save		save fwnotify entry to spt
		url	<url>	set fwnotify url

		days	<days>	set fwnotify days
		active	<flag>	turn on/off fwnotify flag
		disp		display firmware notify information
		check		check firmware notify event
		debug	<flag>	turn on/off firmware notify debug flag
	cmgr			
		trace		
			disp <ch-name>	show the connection trace of this channel
			clear <ch-name>	clear the connection trace of this channel
		cnt	<ch-name>	show channel connection related counter
	socket			display system socket information
	filter			
		netbios		
	roadrunner			
		debug	<level>	enable/disable roadrunner service 0: diable <default> 1: enable
		display	<iface name>	display roadrunner information iface-name: enif0, wanif0
		restart	<iface name>	restart roadrunner
	ddns			
		debug	<level>	enable/disable ddns service
		display	<iface name>	display ddns information
		restart	<iface name>	restart ddns
		logout	<iface name>	logout ddns
	cpu			
		display		display CPU utilization
	filter			
		netbios		
	upnp			
		active	[0:no/1:yes]	Activate or deactivate the saved upnp settings
		config	[0:deny/1:permit]	Allow users to make configuration changes. through UPnP

		display		display upnp information
		firewall	[0:deny/1:pass]	Allow UPnP to pass through Firewall.
		load		save upnp information
		save		save upnp information

Exit Command

Command				Description
exit				exit smt menu

Device Related Command

Command				Description
dev				
	channel			
		drop	<channel_name>	drop channel
	dial		<node#>	dial to remote node

Ethernet Related Command

Command				Description
ether				
	config			display LAN configuration information
	driver			
		cnt		
			disp <name>	display ether driver counters
		ioctl	<ch_name>	Useless in this stage.
		status	<ch_name>	see LAN status
	version			see ethernet device type
	pkttest			
		disp		

			packet <level>	set ether test packet display level
			event <ch> [on|off]	turn on/off ether test event display
		sap	[ch_name]	send sap packet
		arp	<ch_name> <ip-addr>	send arp packet to ip-addr
	debug			
		disp	<ch_name>	display ethernet debug infomation
		level	<ch_name> <level>	set the ethernet debug level level 0: disable debug log level 1:enable debug log (default)
	edit			
		load	<ether no.>	load ether data from spt
		mtu	<value>	set ether data mtu
		accessblock	<0:disable 1:enable>	block internet access
		save		save ether data to spt

POE Related Command

Command				Description
poe				
	status		[ch_name]	see poe status
	dial		<node>	dial a remote node
	drop		<node>	drop a pppoe call
	ether		[rfc|3com]	set /display pppoe ether type

PPTP Related Command

Command				Description
pptp				
	dial		<rn-name>	dial a remote node
	drop		<rn-name>	drop a remote node call
	tunnel		<tunnel id>	display pptp tunnel information

Configuration Related Command

Command	Description

config					The parameters of config are listed below.
edit	firewall	active <yes|no>			Activate or deactivate the saved firewall settings
retrieve	firewall				Retrieve current saved firewall settings
save	firewall				Save the current firewall settings
display	firewall				Displays all the firewall settings
		set <set#>			Display current entries of a set configuration; including timeout values, name, default-permit, and number of rules in the set.
		set <set#>	rule <rule#>		Display current entries of a rule in a set.
		attack			Display all the attack alert settings in PNC
		e-mail			Display all the e-mail settings in PNC
		?			Display all the available sub commands
		e-mail	mail-server <mail server IP>		Edit the mail server IP to send the alert
			return-addr <e-mail address>		Edit the mail address for returning an email alert
			e-mail-to <e-mail address>		Edit the mail address to send the alert
			policy <full | hourly |daily | weekly>		Edit email schedule when log is full or per hour, day, week.
			day <sunday | monday | tuesday | wednesday | thursday | friday | saturday>		Edit the day to send the log when the email policy is set to Weekly
			hour <0~23>		Edit the hour to send the log when the email policy is set to daily or weekly
			minute <0~59>		Edit the minute to send to log when the email policy is set to daily or weekly
			Subject <mail subject>		Edit the email subject
		attack	send-alert <yes|no>		Activate or deactivate the firewall DoS attacks notification emails
			block <yes|no>		Yes: Block the traffic when exceeds the tcp-max-incomplete threshold
					No: Delete the oldest half-open session when exceeds the tcp-max-incomplete threshold
			block-minute <0~255>		Only valid when sets 'Block' to yes. The unit is minute

			minute-high <0~255>		The threshold to start to delete the old half-opened sessions to minute-low
			minute-low <0~255>		The threshold to stop deleting the old half-opened session
			max-incomplete-high <0~255>		The threshold to start to delete the old half-opened sessions to max-incomplete-low
			max-incomplete-low <0~255>		The threshold to stop deleting the half-opened session
			tcp-max-incomplete <0~255>		The threshold to start executing the block field
		set <set#>	name <desired name>		Edit the name for a set
			default-permit <forward|block>		Edit whether a packet is dropped or allowed when it does not match the default set
			icmp-timeout <seconds>		Edit the timeout for an idle ICMP session before it is terminated
			udp-idle-timeout <seconds>		Edit the timeout for an idle UDP session before it is terminated
			connection-timeout <seconds>		Edit the wait time for the SYN TCP sessions before it is terminated
			fin-wait-timeout <seconds>		Edit the wait time for FIN in concluding a TCP session before it is terminated
			tcp-idle-timeout <seconds>		Edit the timeout for an idle TCP session before it is terminated
			pnc <yes|no>		PNC is allowed when 'yes' is set even there is a rule to block PNC
			log <yes|no>		Switch on/off sending the log for matching the default permit
			rule <rule#>	permit <forward|block>	Edit whether a packet is dropped or allowed when it matches this rule
				active <yes|no>	Edit whether a rule is enabled or not
				protocol <0~255>	Edit the protocol number for a rule. 1=ICMP, 6=TCP, 17=UDP...
				log <none|match|not-match|both>	Sending a log for a rule when the packet none|matches|not match|both the rule
				alert <yes|no>	Activate or deactivate the notification when a DoS attack occurs or there is a violation of any alert settings. In case of such instances, the function will send an email to the SMTP destination address and log an alert.
				srcaddr-single <ip address>	Select and edit a source address of a packet which complies to this rule

				srcaddr-subnet <ip address> <subnet mask>	Select and edit a source address and subnet mask if a packet which complies to this rule.
				srcaddr-range <start ip address> <end ip address>	Select and edit a source address range of a packet which complies to this rule.
				destaddr-single <ip address>	Select and edit a destination address of a packet which complies to this rule
				destaddr-subnet <ip address> <subnet mask>	Select and edit a destination address and subnet mask if a packet which complies to this rule.
				destaddr-range <start ip address> <end ip address>	Select and edit a destination address range of a packet which complies to this rule.
				tcp destport-single <port#>	Select and edit the destination port of a packet which comply to this rule. For non-consecutive port numbers, the user may repeat this command line to enter the multiple port numbers.
				tcp destport-range <start port#> <end port#>	Select and edit a destination port range of a packet which comply to this rule.
				udp destport-single <port#>	Select and edit the destination port of a packet which comply to this rule. For non-consecutive port numbers, users may repeat this command line to enter the multiple port numbers.
				udp destport-range <start port#> <end port#>	Select and edit a destination port range of a packet which comply to this rule.
				desport-custom <desired custom port name>	Type in the desired custom port name
delete	firewall	e-mail			Remove all email alert settings
		attack			Reset all alert settings to defaults
		set <set#>			Remove a specified set from the firewall configuration
		set <set#>	rule <rule#>		Remove a specified rule in a set from the firewall configuration
insert	firewall	e-mail			Insert email alert settings
		attack			Insert attack alert settings
		set <set#>			Insert a specified rule set to the firewall configuration
		set <set#>	rule <rule#>		Insert a specified rule in a set to the firewall configuration
cli					Display the choices of command list.
debug	<1|0>				Turn on|off trace for firewall debug information.

IP Related Command

Command				Description
ip				
	address		[addr]	display host ip address
	alias		<iface>	alias iface
	aliasdis		<0|1>	disable alias
	arp			
		status	<iface>	display ip arp status
	dhcp		<iface>	
		client		
			release	release DHCP client IP
			renew	renew DHCP client IP
		status	[option]	show dhcp status
	dns			
		query		
		server	<primary> [secondary] [third]	set dns server
		stats		
			clear	clear dns statistics
			disp	display dns statistics
	httpd			
	icmp			
		status		display icmp statistic counter
		discovery	<iface> [on|off]	set icmp router discovery flag
	ifconfig		[iface] [ipaddr] [broadcast <addr> |mtu <value>|dynamic]	configure network interface
	ping		<hostid>	ping remote host
	route			
		status	[if]	display routing table
		add	<dest_addr|default>[/<bits>] <gateway> [<metric>]	add route
		addiface	<dest_addr|default>[/<bits>] <gateway> [<metric>]	add an entry to the routing table to iface
		addprivate	<dest_addr|default>[/<bits>] <gateway> [<metric>]	add private route

		drop	<host addr> [/<bits>]	drop a route
	smtp			
	status			display ip statistic counters
	stroute			
		display	[rule # | buf]	display rule index or detail message in rule.
		load	<rule #>	load static route rule in buffer
		save		save rule from buffer to spt.
		config		
			name <site name>	set name for static route.
			destination <dest addr>[/<bits>] <gateway> [<metric>]	set static route destination address and gateway.
			mask <IP subnet mask>	set static route subnet mask.
			gateway <IP address>	set static route gateway address.
			metric <metric #>	set static route metric number.
			private <yes|no>	set private mode.
			active <yes|no>	set static route rule enable or disable.
	traceroute		<host> [ttl] [wait] [queries]	send probes to trace route of a remote host
	xparent			
		join	<iface1> [<iface2>]	join iface2 to iface1 group
		break	<iface>	break iface to leave ipxparent group
	ave			anti-virus enforce
	urlfilter			
		reginfo		
			display	display urlfilter registration information
			name	set urlfilter registration name
			eMail <size>	set urlfilter registration email addr
			country <size>	set urlfilter registration country
			clearAll	clear urlfilter register information
		category		
			display	display urlfilter category
			webFeature [block/nonblock] [activex/java/cookei/webproxy]	block or unblock webfeature

			logAndBlock [log/logAndBlock]	set log only or log and block
			blockCategory [block/nonblock] [all/type(1-14)]	block or unblock type
			timeOfDay [always/hh:mm] [hh:mm]	set block time
			clearAll	clear all category information
		listUpdate		
			display	display listupdate status
			actionFlags [yes/no]	set listupdate or not
			scheduleFlag [pending]	set schedule flag
			dayFlag [pending]	set day flag
			time [pending]	set time
			clearAll	clear all listupdate information
		exemptZone		
			display	display exemptzone information
			actionFlags [type(1-3)][enable/disable]	set action flags
			add [ip1] [ip2]	add exempt range
			delete [ip1] [ip2]	delete exempt range
			clearAll	clear exemptzone information
		customize		
			display	display customize action flags
			logFlags [type(1-3)][enable/disable]	set log flags
			add [string] [trust/untrust/keyword]	add url string
			delete [string] [trust/untrust/keyword]	delete url string
			clearAll	clear all information
		logDisplay		display cyber log
		ftplist		update cyber list data
		listServerIP	<ipaddr>	set list server ip
		listServerName	<name>	set list server name
	tredir			
		failcount	<count>	set tredir failcount
		partner	<ipaddr>	set tredir partner

		target	<ipaddr>	set tredir target
		timeout	<timeout>	set tredir timeout
		checktime	<period>	set tredir checktime
		active	<on|off>	set tredir active
		save		save tredir information
		disp		display tredir information
		debug	<value>	set tredir debug value
	nat			
		server		
			disp	display nat server table
			load <set id>	load nat server information from ROM
			save	save nat server information to ROM
			clear <set id>	clear nat server information
			edit active <yes|no>	set nat server edit active flag
			edit svrport <start port> [end port]	set nat server server port
			edit intport <start port> [end port]	set nat server forward port
			edit remotehost <start ip> [end ip]	set nat server remote host ip
			edit leasetime [time]	set nat server lease time
			edit rulename [name]	set nat server rule name
			edit forwardip [ip]	set nat server server ip
			edit protocol [protocol id]	set nat server protocol
			edit clear	clear one rule in the set
		service		
			irc [on|off]	turn on/off irc flag
		resetport		reset all nat server table entries
		incikeport	[on|off]	turn on/off increase ike port flag
	igmp			
		debug	[level]	set igmp debug level
		forwardall	[on|off]	turn on/off igmp forward to all interfaces flag
		querier	[on|off]	turn on/off igmp stop query flag
		iface		

			<iface> grouptm <timeout>	set igmp group timeout
			<iface> interval <interval>	set igmp query interval
			<iface> join <group>	join a group on iface
			<iface> leave <group>	leave a group on iface
			<iface> query	send query on iface
			<iface> rsptime [time]	set igmp response time
			<iface> start	turn on of igmp on iface
			<iface> stop	turn off of igmp on iface
			<iface> ttl <threshold>	set ttl threshold
			<iface> v1compat [on|off]	turn on/off v1compat on iface
		robustness	<num>	set igmp robustness variable
		status		dump igmp status
	pr			

PPPoA Related Command

Command				Description
ppp				
bod				
ccp				
lcp				
	acfc			Address/Control Field Compression
	pfc			Protocol Field Compression
	mpin			Incoming calls' MP
	callback			Callback
	bacp			Bandwidth Allocation Control
	echo			
		retry	<retry-count>	retry count to send echo-request
		time	<time(s)>	time interval to send echo-request
ipcp				
	close			
	list		<iface>	

	open			
	timeout			
	try			
		configure		
		failure		
		terminate		
	compress			Disable or enable VJ header compress
	slots			Set number of slots
	idcompress			Disable or enable Slot ID compress
	address			Disable or enable IPCP address option
mp				
	default			Link default on as rotat
	split		y ,yes, true, on 1 set enable n ,no, false off 0 clear disable	
	rotate		y ,yes, true, on 1 set enable n ,no, false off 0 clear disable	Link rotate
	sequence			Set MP start sequence
configur e				
	ipcp			
		compress		Disable or enable VJ header compress
		slots		Set number of slots
		idcompress		Disable or enable Slot ID compress
		address		Disable or enable IPCP address option
	ccp			
		ascend		Disable or enable Ascend Stac
		history	only 0 or 1 allowed	Set Stac History Cnt
		check		Stac check mode
		reset		Ascend Stac reset mode
		pfc		Disable or enable PFC
		debug		Disable or enable CCP debug

iface			<name> <ptcl>	
show			<channel>	
fsm				
	trace			
		break		
		clear		
		disp		
		filter		
	tdata			
		filter		
		disp		
		clear		
delay				Set PPP pkt delay(ms)

Hdap Related Command

Command				Description
sys	Hdap			
		debug	on or off	
		reset		

Dsp Related Command

Command				Description
dsp				
phonetest			1 or 0	
ringtest			[ID] [Item]	
slic_dump			[tcid:0/1]	
spylevel			0: General Info 1: Entering Function 2: Normal and Expected Event 3: Minor Unexpected Event 4: Major Unexpected Event 5: Fatal Error 6: Turn off Spy	Set spy level
companding			1: A Law 2: Mu Law 3: Linear	Set current companding

tonetest			[ID] [Number]	
dspi			show [<image_id> <level>]	
dim				
	opt		<opt_value>	Set a DSP Option
	ver			Display DSP version information
	stats		<dsp> [clear] <dsp> <chan> [clear]	Display/Clear Dsp Stats Display/Clear Channel Stats
	frfstats			Display FRF.11 Stats
	dump		<dsp> <offset (hex)> <length (decimal - words)>	
	silence		<tcid>	
	cfg			Dim Config info
	pcm_trace		<tcid> <on> [on_count] [off_count] <tcid> <off>	Turn on PCM sample trace Turn off PCM sample trace
	poll stats			Dim poll period stats
	set_poll		<msecs>	Set the DIM poll period manually
	gsync		<tcid> <msecs>	Send grant sync to DSP
	aal2_prof		<tcid>	Display the active AAL2 profile
	set_voice_poll		<on|off>	Enable/disable voice pkt polling
fxs				
	reg		<tcid> <regnum> [<regval>]	Display/Set direct register
	ireg		<tcid> <regnum> [<regval>]	Display/Set indirect register
	power			Display Power up status
	calib			Display Calibration status
cc			TI DSP command	
initstatus				Display DSP init result

Voice Related Command

Command					Description
voice					
config					
	rtp				
		index		<index>	Select RTP index.

		sortingbuffer		<index> <0:0ms|1:10ms|2:20ms>	Disable or enable receive sorting buffer(default 0 ms)
		rtcpinterval		<index> <ms>	Change the RTCP transmission interval(default 15000)
		packetsize		<index>g711<0:10ms|1:20ms|2:30ms>g729<0:10ms|1:20ms|2:30ms>	Change the transmit packetized period(default 10ms)
		save		<index>	Save the configured value
		display		<index>	Display the configured value.
		dumpCfg			Display working buffer value.
		free			Free working buffer.
	pstn				
		index		<index>	Select the PSTN index
		phonebook		<index> <0~32 digits/blank>	Signaling the phone number
		prefixcode		<index> <1:enable | 0:disable>	Disable or enable the prefix code
		active		<index> <1:active | 0:in-active>	Disable or enable the speed dial
		save		<index>	Save the configured value
		display			Display the configured value.
		dumpCfg		<index>	Display working buffer value.
		free			Free working buffer.
	signal				
		active		<index> <1:active | 0:in-active>	Active/in-active this setting.
		index		<index>	Select SIP index.
		register		<index> <auto|enter|exit>	Change the registrar type
		registertimeout		<index> <second>	Setup registration timeout value. (default 3600 sec)
		registerresendtime		<index> <second>	Setup registration resend timeout value.
		minse		<index> <100-1800 second>	Setup minimum session timeout value.
		retransmitT2		<index> <4|8|16|32sec>	Not support yet
		sessiontimeout		<index> <100-3600 second>	Setup session timeout value(default 300 sec)
		serveraddress		<index> <ip address>	Signaling server address

		serverport		<index> <1024-65535>	Signaling server port (default 5060)
		registeraddress		<index> <ip address>	Signaling register address
		registerport		<index> <1024-65535>	Signaling register port(default 5060)
		userid		<index> <0-20 chars>	Signaling SIP user-id
		password		<index> <0-20 chars>	Signaling SIP password
		urltype		<index> <sip|tel>	SIP URL type
		port		<index> <1024-65535>	Signaling port
		phonenumber		<index> <0-32 chars>	Signaling phone number
		domain		<index> <1-128 chars>	Setup domain of SIP service
		dtmf		<index> <rfc2833|pcm|sipinfo>	Setup DTMF key type.
		pri_compression		<index> <0:G711mu|8:G711A|18:G729>	Change the primary compression type
		sec_compression		<index> <0:G711mu|8:G711A|18:G729>	Change the secondary compression type
		portrange		<index> <start port> <end port> (49152~65534)	RTP/RTCP port range setting
		transport		<index> <udp|tcp>	Setup SIP transport type.
		callerid		<index> <disable|enable>	Disable or enable the caller id feature for VoIP.
		autoredialpstn		<index> <disable|enable>	Disable or enable the auto redial
		phoneselect		<index><phone port|0:All><0:No|1:Yes>	Setup incoming call mapping to phone port.
		vlantag		<index> <disable|enable>	Enable/disable VLAN Tag in VoIP packet.
		tpid_vlantag		<index> <TPID value (4 Byte)>	Setup VLAN Tag - TPID.
		vid_vlantag		<index> <VID value (3 Byte)>	Setup VLAN Tag - VID.
		priority_vlantag		<index> <TCI value (0-7)>	Setup VLAN Tag - TCI.
		diffservrtp		<index> <0-7>	Setup DiffServRtp for QoS.
		diffservsip		<index> <0-7>	Setup DiffServSip for QoS.
		stunactive		<index> <0:off|1:on>	Active/in-active STUN service.
		stunservaddr		<index> <ip address>	Setup STUN server IP Address.
		stunservport		<index> <port>	Setup STUN server port number.
		mwiactive		<index> <0:off|1:on>	Disable or enable the voice message

		save		<index>	Save the configured value
		display		<index>	Display the configured value.
		dumpCfg			Display working buffer value.
		free			Free working buffer.
		t38Protocol		<index><0:UDPTL|1:RTP>	Set T38 Protocol
		t38RateMgnt		<index><0:localTCF|1:transferredTCF>	Set T38 Rate
		t38HsPktRate		<index><10~80>ms	Set T38 High Speed Packet Rate
		t38HsRedun		<index><0~3>	Set High Speed Redundancy
		t38LsRedun		<index><0~8>	Set Low Speed Redundancy
		t38T30Ecm		<index><0:disable|1:enable>	Enable / Disable ECM
		t38MaxRate		<index><rate>	Set T38 Max Rate
	dsp				
		index		<index>	Select DSP index.
		echocancellation		<index> <enable|disable>	Disable or enable the echo cancellation
		jittersize		<index> <0~90>ms	Change the jitter buffer size for DSP
		start		<index> <loop|ground>	Change the pots type.
		vad		<index> <enable|disable>	Disable or enable the VAD
		dialtype		<index> <pstn|tone|pulse>	Change the dialing type
		dialtonetype		<index> <ntt:pdt>	Change the dial tone type
		dialinterval		<index> <3~5>	Change the dialing interval
		flashinterval		<index> <1:1.2sec|2:2.2sec>	Change the flash key interval
		inputvolume		<index> <0 ~ -15>	Change the input volume gain
		outputvolume		<index> <0 ~ -15>	Change the output volume gain
		receivetonetype		<index> <ir|sir>	Change the receive tone type.
		sipselect		<index><phoneport|0:All><0:No|1:Yes>	Select SIP index and Disable or enable the SIP
		vifsize		<index> <1:10ms 2:20ms 3:30ms>	
		save		<index>	Save the configured value
		display		<index>	Display the configured value.

		dumpCfg			Display working buffer value.
		free			Free working buffer.
		dspi		<image_id> <level>	level is 1 for least info up to 4 for most levels at or above 4 produce several minutes worth of listings
		dim			
			opt	<opt_value>	
			ver		
			stats	<dsp> [clear] <dsp> <chan> [clear]	Display/Clear Dsp Stats Display/Clear Channel Stats
			frfstats		Display FRF.11 Stats
			dump	<dsp> <offset (hex)> <length (decimal - words)>	
			silence	<tcid>	
			cfg		Dim Config info
			pcm_trace	<tcid> <on> [on_count] [off_count] <tcid> <off>	Turn on PCM sample trace Turn off PCM sample trace
			poll stats		Dim poll period stats
			set_poll	<msecs>	Set the DIM poll period manually
			gsync	<tcid> <msecs>	Send grant sync to DSP
			aal2_prof	<tcid>	Display the active AAL2 profile
			set_voice_poll	<on|off>	Enable/disable voice pkt polling
		fxs			si3210 Debug Commands
			reg	<tcid> <regnum> [<regval>]	Display/Set direct register
			ireg	<tcid> <regnum> [<regval>]	Display/Set indirect register
			power		Display Power up status
			calib		Display Calibration status
		cc		TI DSP command	RD debug command.
		fax		<index> <0:FAX_Relay|1:T.38>	Select Fax Option
	phbook				

		index		<index>	Select phone book index.
		active		<index> <1:active|0:inactive>	Active/in-active this setting.
		orignum		<index> <0~32 digits>	Setup phone number for this index of phone book.
		forcesipuri		<index> <1-128 chars>	Setup force SIP URI.
		speednum		<index> <0~32 digits>	Setup speed dial number.
		name		<index> <name>	Setup the name for the description.
		type		<index> <0:Proxy|1:NonProxy>	Select Proxy or Non-Proxy type.
		save		<index> Please select the Phone Book index to configure: 1 ~ 10	Save the configured value
		display		<index>	Display the configured value.
		dumpCfg		<index>	Display working buffer value.
		free		<index> <1:active|0:inactive>	Free working buffer.
	common				
		index		<index>	
		save		<index>	Save the configured value.
		removepound		<index> <0:not removed|1:removed pound>	
		countrycode		<index><CountryCode|h:for help>	Set the countrycode
		display		<index>	Display the configured value.
		dumpCfg		<index>	Display working buffer value.
	autopro				
		index			
		active			Disable or enable autopro
		servaddr			Set the server address
		timeout			
		retry			Disable or enable retry
		save			Save the configured value.
		display			Display the configured value.
		dumpCfg			Display working buffer value.

<table>
<tr><td></td><td>version</td><td></td><td></td><td></td><td>Show the VOIP version</td></tr>
<tr><td>fsm</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>status</td><td></td><td></td><td><phone | ccm | convert | rtp | ua></td><td>Display the status of fsm.</td></tr>
<tr><td></td><td>convert</td><td></td><td></td><td><disp | reset<u_id s_id ype>></td><td>Display convert information or reset convert.</td></tr>
<tr><td></td><td>debug</td><td></td><td></td><td><all | phone | ccm | sip | ua | rtp></td><td>Turn on debug message.</td></tr>
<tr><td>sip</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>tl</td><td></td><td></td><td><0|1|2|3|…|7></td><td>SIP debug trace level
0:None 1: All 2: Tx 3:Rx
4: TXRx 5: State 6: Error
7: Warning</td></tr>
<tr><td></td><td>siginit</td><td></td><td></td><td></td><td>Init SIP SIG task</td></tr>
<tr><td></td><td>sipinit</td><td></td><td></td><td></td><td>Init SIP protocol stack</td></tr>
<tr><td></td><td>sipterm</td><td></td><td></td><td></td><td>Delete SIP protocol stack</td></tr>
<tr><td></td><td>sigmakecall</td><td></td><td></td><td>type in phone num</td><td>Make Call</td></tr>
<tr><td></td><td>sigbusy</td><td></td><td></td><td></td><td>Response Busy Call</td></tr>
<tr><td></td><td>sigringback</td><td></td><td></td><td></td><td>Response Ring</td></tr>
<tr><td></td><td>sigreg</td><td></td><td></td><td><index></td><td>Register to SIP server</td></tr>
<tr><td></td><td>sigunreg</td><td></td><td></td><td><index></td><td>Unregister Sip server.</td></tr>
<tr><td></td><td>regstatus</td><td></td><td></td><td><index></td><td>Show register’s information.</td></tr>
<tr><td></td><td>sigok</td><td></td><td></td><td></td><td>pick up a call</td></tr>
<tr><td></td><td>sigbye</td><td></td><td></td><td></td><td>drop or cancel a call</td></tr>
<tr><td></td><td>sipclose</td><td></td><td></td><td></td><td>Close the SIP</td></tr>
<tr><td></td><td>username</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>proxy</td><td></td><td></td><td><0:off|1:on></td><td>Use SIP proxy</td></tr>
<tr><td></td><td>contact</td><td></td><td></td><td><LAN Addr:0|Remote Node# WAN Addr:1-2></td><td></td></tr>
<tr><td></td><td>ackbranch</td><td></td><td></td><td><on: the Ack bring branch ID, off: the Ack didn't bring branch ID></td><td>Setup bring branch ID.</td></tr>
</table>

	rfc3262			<1: turn on RFC3262, 0: turn off RFC3262>	Disable or enable the rfc3262
	rfc3325			<1: privacy call using RFC3325, 0: pirvacy call using draft-01>	Disable or enable the rfc3325
	wanip				STUN debug command.
	port				STUN debug command.
	keepalive			<index>	STUN debug command.
	checkwan			<index>	Display WAN IP/Port for SIP.
	changemedia			<index> <RTP addr> <RTP port>	RD debug command.
	siglistdump			<index>	RD debug command.
	dl				Set up SIP trace level
	sigmwi			<index>	
	sigunmwi			<index>	
	disconnect			<index> or 'all'	
rtp					
	bye	[port]	[sessid]		
	connect			port[0-3] destIp destPort srcPort PT	
	add			port[0-3] destIp destPort srcPort PT	
	table				Display all the current active RTP session
	usage				Display all the used port
	rxtime			<msec>	Setup RX time
	txtime			<msec>	Setup TX time
	Dtmf			digit# 1 = 1 digit# 2 = 2 digit# 3 = 3	Set the DTMF value.
	statistics			<index>	Show the statistics
	linktime			<index>	Show the RTP linktime
logTest					RD debug command.
tebasic					For TE use only.