ZyXEL

Firmware Release Note

Prestige P2002L

Standard version

Release 3.60(MH.2)C0

Date:	**June 6, 2005**
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ZyXEL Prestige P2002L Standard Version release 3.60(MH.2)C0 Release Note

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Supported Platforms:

ZyXEL Prestige P2002L

Versions:

Bootbase Version: V1.08 | 04/15/2005 17:12:54
ZyNOS Version: V3.60(MH.2) | 06/06/2005 13:43:14

Notes:

1. Management port is 192.168.5.1.
2. Bootbase version MH1.02 support AMD Flash only.
3. Bootbase version MH1.03 support Intel Flash only.
4. Support C3 hardware, please use f/w after 3.60(MH.0)b1 and Bootbase 1.03. (Intel flash)
5. Bootbase version after MH1.04 support both Flash type (Intel and AMD).
6. Add "Multi-Upload firmware" feature in bootext for OBM version.
7. In log information on eWC, if you set the SIP number more than 28 characters, we use "…" to represent the remainder.
8. New DSP, Version : Rel 9.1.500.8
9. Support Multiboot client V2
10. Remove T38 Feature.
11. Add new country code [Others] in web→phone→common page, the country code of [Others] is 254, means use default setting in romfile, the country code [Default] is 255, means use country setting in MRD country code field.

Features:

Modifications in V 3.60(MH.2)b4 | June 3, 2005

[Enhancement]

1. Add new country code [Others]

Modifications in V 3.60(MH.2)b3 | June 2, 2005

1 [Bug fixed][SPR No: 050518052]
Symptom: DUT occur exception frequently after click Log icon in GUI. In this case, the log items are over 120.

2 [Bug fixed][SPR No: 050531914]
Symptom: When test call on hold, use "RR+2" switch calls sixth times, the DUT can't work.

3 [Bug fixed][SPR No: 050531916]
Symptom: In CLI, the ringbacktone and musiconhold are only set 1~ 8, now set 1~10.

Modifications in V 3.60(MH.2)b2 | May 23, 2005

4 [Bug fixed]

4.1 (050517954) Suggest change some default value
1. Incoming PSTN call should apply to Phone 1 and Phone 2. Currently is None.
2. Flash MAX/MIN interval should be changed to europe spec (160/40). Currently is 1000/100.

4.2 (050517955) Change incoming PSTN call doesn't work on GUI unless reboot to DUT. Sometimes even reboot DUT, it still only apply to Phone 1 only.

4.3 (050517956) SIP/RTP TOS value can only accept one character on GUI. It should allow range 0~255.

4.4 (050517957) Below dial plan doesn"t work.
1. (<5:>xxxx@) . Dial 56789 -> result 67899. Correct: 6789
2. (x.) Dial any digit, dialplan block any digit.

4.5 (050510499) With G.729 -> G.711 codec selection, G3 fax always failed. With G.711 -> G.729 codec selection, G3 fax is OK.

4.6 (050518118) Press “CTRL+C ” when send ICMP packet by using CI command, DUT will occur exception.

4.7 (050518113) DUT occur exception when DUT receive many Invite. Test item: Create many call test

4.8 (050518114) After unplug WAN connection during VoIP conversation or rining, DUT can't send to Invite or receive VoIP call.
1. Make VoIP call.
2. Cut off DUT's WAN connection during DUT is rining or VoIP call.
3. DUT sometimes can't make or receive VoIP call again.

Two symptom found:
1. DUT can't accept Invite.
2. DUT's one of the Phone doesn't rining when receive Invite.

Result after modification: ATA still ringing after unplug wan connection. But system recover back fine after on-hook.

4.9 (050518116) 1. No "incoming life line" call description on Help page. 2. Remove "email log now" in Log's help page.

4.10 ()

5 [Enhancement]

No new items.

Modifications in V 3.60(MH.2)b1 | May 12, 2005

6 [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 Speed Dial phonebook
1.6 Support On-hold & Un-hold
1.7 Support Second-Call
1.8 Support Call-Waiting
1.9 Support Call-Transfer (Blind Transfer / Consultative Transfer)
1.10 Support MWI
1.11 Support congestion tone
1.12 Support Call Conference
1.13 Support SPTGEN
1.14 Support Auto Provision (TFTP / HTTP)
1.16 Support IVR / Embedded IVR
1.17 Supprot Early Media

1.19 Support Music On Hold
1.20 Support Auto Firmware Update (It works only when sip is registered and have new firmware version need to update)
1.21 Supprot Dial Plan
1.22 [Enhancement Change] Support Diff-Serv Tagging , support set SIP/RTP TOS value from 0 to 255. The default value is 160(0xa0).
1.23 Call forward (unconditional/busy/no answer)
1.24 changeable FXO parameter (CI command only)
1.25 DTMF/FSK CLID transmission (CI command only)
1.26 SIP/PSTN fall back
1.27 Call Feature Key List

	USA	Europe (Default)
Call-Hold / Call-Retrieve	<RR>: Put call on hold and apply dial-tone. <RR>: Switch between the first call and second call.	<RR>: Put call on hold and apply dial-tone. <RR>: Switch back to first call. This is active only when there is no second call. <RR> +2: Switch between the first call and Second call. <RR> +0: Clear held & Stay on the active call. <RR> +1: Clear active & Switch to held call.
Call-Waiting	<RR>: Accept Waiting, hold active call.	<RR> +0: Reject Waiting Call <RR> +1: Accept Waiting, clear active call. <RR> +2: Accept Waiting, hold active call.
3-way Calling [At DSP]	<RR>: After the second call is successful But this feature will impact the switch between the first call and second call On-Hook: Call Transfer to link the first call and second call.	<RR> +3: Establish 3-way calling [Active after the second call] <RR> +2: Split the 3-way calling. <RR> +2: Switch between the first call and second call. On-Hook: Call Transfer to link the first call and second call.
3-way Calling [At Media server]	Same as above	Same as above. But the details depend on the server.
Blind Transfer	<RR> + *98#:	<RR> + *98#
Consult-On-Hold Transfer	<RR> + *98#:	<RR> + *98#
Internal Call	####	####
	#*1, #*2, #*3 for the specific extension. [Not support now]	#*1, #*2, #*3 for the specific extension. [Not support now]
Speed dial	#01 ~ #10	#01 ~ #10
IVR Main Menu	****	****
Auto Firmware Update	*99#: Do the firmware update. #99#: Cancel the firmware update	*99#: Do the firmware update. #99#: Cancel the firmware update.

Modifications in V 3.60(MH.1)c0 | November 1, 2004

7 [Bug fixed]

7.1 (113) Sometimes MGMT port can't be Ping or Managed after boot up ATA without connect LAN port. After reboot, it works.
7.2 ATA occurs console hang up or reboot during stress testing.
7.3 180 ringing delay sending out.
7.4 System crashes if doing Auto-Provision.
7.5 When Off Hook at first ting, users hear Caller-ID sound. Caller-ID can't display on phone 1.

8 [Enhancement]

No new items.

[Update release note]

Add SPTGEN function since 360(MH.1)b4

Modifications in V 3.60(MH.1)b5 | October 13, 2004

9 [Bug fixed]

9.1 (45) ATA use incorrect DNS server to send out DNS query.
9.2 (48) There is no help page for VoIP setting common page.
9.3 (49) There is no Fake IP/ Outbound Proxy /NAT Keep alive on SIP advance setting help page.
9.4 (52) 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.
9.5 (53) ATA always relay to LifeLine when SIP account 1 is unregistered even the SIP account 2 is registered.
9.6 (54) By using Windows Messenger or XTEN to call ATA, ATA will not send out "BYE" when Phone is On-Hook.
9.7 TFTP timeout time always delay 10sec to be executed

10 [Enhancement]

2.1 Add Autoprovision key word display, use "voice autopro dbdisplay" to get the information

Modifications in V 3.60(MH.1)b4 | October 4, 2004

1 [Bug fixed]

1.1 (45) If PSTN call carrier CID with time information, the time info will be filter by ATA.. (The time information is not support yet.)
1.2 (46) By using G.729 codec and push music on hold on Phone, some of music sounds was filtered.(**This bug is change to "know issue"**)

2 [Enhancement]

2.1 Add SPTGEN function .

Modifications in V 3.60(MH.1)b3 | September 9, 2004

1 [Bug fixed]

1.1 When use HTP to test hardware, Phone 2 LED can not lamp.

2 [Enhancement]

No new items.

Modifications in V 3.60(MH.1)b2 | September 1, 2004

1 [Bug fixed]

1.1 (33) Incoming PSTN call CID display doesn't work.

2 [Enhancement]

No new items.

Modifications in V 3.60(MH.1)b1 | August 31, 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 (46) By using default Volume setting (Volume:0), noise is noticeable.
Suggestion:
Change default Volume setting to "-1"
1.4 (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.

2 [Enhancement]

No new items.

Modifications in V 3.60(MH.0)c0 | August 27, 2004

Change ZyNOS Version from 3.60(MH.0)b5 to 3.60(MH.0)c0

Modifications in V 3.60(MH.0)b5 | August 23, 2004

1 [Bug fixed]

1.1 (31)In PPPoE mode, Mgmt port shows "subnet mask" as "0.0.0.0" in status page on eWC
1.2 (32)Back to Factory Default would redirect WEB page to 192.168.1.1
1.3 (37)Non exist "show Statistics" function show in maintenance Help page
1.4 "Show Statistics - Click Show Statistics to see router performance statistics such as number of packets sent and number of packets received for each port."
1.5 (41)Exception occurs by using below scenario
1. Make PSTN Call.
2. Make VoIP Call.
3. Repeat step 1 and 2 for several times.
1.6 (42) Phone "Volume" shows incorrect dB mapping.

	G.711	G.729
-1	-2.05	-2.67
0	11.48	10.95
1	X	X

1.7 (45)Exception occurs during continuous PSTN call test.
1. Using RADCOM equipment to continous make PSTN call.
2. After 20~30 PSTN call, ATA will occur exception.

2 [Enhancement]

2.1 Add “Multi-Upload firmware” feature in bootext for OBM version
2.2 Bootbase version MH1.04 support both Flash type (Intel and AMD).

Modifications in V 3.60(MH.0)b4 | August 12, 2004

1 [Bug fixed]

1.1 the field 'ptime' in SDP doesn't match actual RTP packetsize.
Condition: Set VIF size other than 20ms since the default ptime is set to 20ms.
Root Cause: ptime always uses default value of 20ms.
1.2 Can’t get correct rtp port when STUN active.
1.3 In CI command mode, when you execute “ppp ipcp fsm open(or close)”,memory will be exception .
1.4 Remove the help icon from Reg_Fail.html and Reg_Wait.html.
1.5 When the SIP 1 is not active, SIP 2 can not be register or unregister.
1.6 (100)"End of Event" doesn't carrier in last RFC2833 DTMF event's payload.
1.7 (106)ATA send DNS server query every 5 sec after used FQDN name in SIP server address on eWC.
1.8 (134) Online Configuration for PHONE setting on eWC. Volume setting/ VAD/ Echo setting doesn't work.
1.9 (146) Online Configuration for RTP port setting doesn't work
1.10 (147) If invalid(not exist) host was used in primary DNS, DNS query will not work.
1.11 (148) With "PPPoE+Static IP", MGMT port show "0.0.0.0" in maintenance page.

1.12 (149)ATA sometimes doesn't generate Busy tone when receiving 4xx response from proxy server.
1.13 (152) Sometimes ATA hang before "VC-5402 Init" print out. Frequency: two times
1.14 (154)Change SIP local port, eWC maintenance info will not be changed
1.15 (156)By using MAC OS IE5.2 to login ATA's eWC doesn't work.
1.16 (158)Volume control doesn't work. G.711/G.729 codec volume always keep in 14.7/14.3 dB
1.17 P2002 FCS sample (AE) will reboot when init SLIC in low temperature.
1.18 (50) By using G.729 codec to transfer FAX sometimes cause ATA occurs exception.
1.19 (86) ATA occur exception when running CERT SIP test.
1.20 (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.
1.21 (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".
1.22 (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.
1.23 (123)After restart ATA and the time setting become default value.
1. Set Time protocol to NONE.
2. Set Current Time.
3. Restart ATA
1.24 (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)
1.25 (128)With default Volume setting, Echo and background noise are noticeable. Expect: Lower default volume setting.
1.26 (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.
1.27 (131)Phone Book show each empty as "¡K". Expect: change to "NULL" character
1.28 (132)SIP Password 19 character is too short for some beta tester's ISP.
1.29 (135)VoIP call GOS(Grade of Loss) issue by measuring with Radcom
1.30 (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.
1.31 (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.
1.32 (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.
1.33 (142)By default ROM file, Ethernet Subnet/Default Gateway field on eWC is not grey out.
1.34 (143)[Help Page] No SIP/RTP priority setting and Dialing interval description in help page.
1.35 (144)Incorrect RTP port was used "50004". eWC setting From RTP port "50000"
1.36 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.
1.37 After turn on VAD, the VoIP communication appear stop for a short period of time then continue. Default Volume setting.
1.38 G.711u/G.729 VoIP conversation delay variance => 10ms+/-6, Expect => 10ms+/-1. Have been fixed in previous version.
1.39 [Help Page on eWC]
1.39.1 Incorrect System DNS server on System/General Help page.

1.39.2 No Ethernet and Log help page.
1.39.3 Statistics help page appear on help page but no Statistics function appear.
1.40 By default rom file, eWC show SIP account 2 as registration failed. Expect: Not register
1.41 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.
1.42 Set "space" in SIP number on Speed dial Phone book, but the entry could not be used to dial out. Not a problem.
1.43 WAN to LAN VoIP call drop after 3 minutes in STUN scenario.
1.43.1 Topology:P2002(Caller) <--- ZyWALL --> P2002 (Callee) with STUN enabled.
1.43.2 Initiate VoIP call from Caller to Callee.
1.43.3 Wait 3 minutes and the VoIP call was dropped.
1.44 No STUN keep-alive interval function implemented. The STUN keep-alive interval value will be changed automatically.
1.45 [Phone Book] SIP number's MAX character is 31. It's too short for most SIP URI case.
1.45.1 Suggest: 64 characters.
1.45.2 Have been update eWC to 128 characters.
1.46 [Help Page on eWC] Chang the all words "ZyAIR" to "ATA"
1.47 [Help Page on eWC] No PHONE BOOK help page.
1.48 SIP Password 19 character is too short for some beta tester's ISP. Have been extended to 96 characters.
1.49 Online Configuration for PHONE setting->Out-Going Call mapping to SIP only on eWC

2 [Enhancement]

2.1 The rtpUptime value is zero and caused exception occur when open "Show Statistics" Page on eWC and Make VoIP Call.
2.2 Support PRACK (RFC 3262).
2.3 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
2.4 Modify for debug tool, show all dlg & tx status.
2.5 Support RFC2833 Like SIP INFO.
2.6 Auto-provision/Auto-firmware upgrade by TFTP

Modifications in V 3.60(MH.0)b3 | July 13, 2004

1 [Bug fixed]

1.1 (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.
1.2 (83)Create Many call by using Prolab at the same time, ATA will occur exception.
1.3 (103)[eWC] By default rom file, eWC show SIP account 2 as registration failed. Expect: Not register.
1.4 (104)[eWC] Two G.168 active selections.
1.5 (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.
1.6 (108)ATA occur exception after Overnight VoIP reliability test.
1. P2002 <--> LAN <--> P2002
2. Initiate dual channel VoIP call.
3. Run for overnight test.

1.7 (109)[eWC] Phone book -> SIP number, Name, Non-Proxy doesn't start from first character when user inputted on eWC.
1.8 (111)RTP/SIP ToS range incorrectly limited to ""0~5"" on CI command. Expect: 0~7.
1.9 (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.
1.10 (119)[eWC][Phone Book] Edit a "use proxy" speed dial entry, the entry will show as "non-proxy" type.
1.11 (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.
1.12 (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.
1.13 (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.
1.14 (128)ATA doesn't generate sip log and rtp log
1.15 (128)With default Volume setting, Echo and background noise are noticeable. Expect: Lower default volume setting."

2 [Enhancement]

2.1 (50)FAX Relay. By using G.729 codec to transfer FAX always fail. Default codec is G.729 -> G.711.
2.2 (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.
2.3 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.)
2.4 Add Country option in VOIP->Common group.
2.5 Support privacy call (including RFC3325 and call draft). Add CI command let RFC3325 and call draft adjustable.
2.6 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.
2.7 Don't care SIP Packet when all headers are un-known.
2.8 Support on-line configuration for phone. (CI command)

3 [VoIP SIP Interoperability]

3.1 Can't interoperate with telia softphone through telia sip server which support TCP & UDP.
3.2 IPTEL SIP server.

Modifications in V 3.60(MH.0)b2 | June 10, 2004

3 [Bug fixed]

3.1 Make a call from office outside to Lifeline FXO port but can’t hear Ringing signal. Off hook and make call again, can hear Ringing signal.
3.2 ATA occur exception when receive 302 – Move Temporary.
3.3 ATA occur exception when receiving SIP REFER packet.
3.4 [eWC] ATA occur exception when open eWC->Maintenance page will cause system exception.
3.5 ATA should not send out SIP invite with itself SIP Phone number.
3.6 [eWC] VoIP -> SIP server address / Register server address should accept FQDN format address.
3.7 Phone book: phone number should be fully matched before switch to PSTN.
3.8 Wrong DTFM signaling bits is carried in INFO packets. Only 1~6 work correct.
3.9 [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.

3.10 [eWC] Online configuration didn't work for SIP account configuration page setting.
3.11 [eWC] Change SIP local Port to port 5070, VoIP status on Maintenance menu still show UDP/5060 port
3.12 [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.
3.13 Change default setting to “G.711 > G.729” from “G.729 > G.711”, default codec will use G.711.
3.14 [eWC] Hide VoIP->Advanced->T2 Timer for the T2 Timer can’t configure problem.

4 [Enhancement]

4.1 [eWC] Add Active option to VOIP->SIP page.
4.2 Support Caller ID from Lifeline. (ETST standard, First Ring, FSK)
4.3 Pass CERT test.

5 [VoIP Interoperability]

5.1 Windows Messenger.

Modifications in V 3.60(MH.0)b1 | June 3, 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] Phone 2 voice volume control can't be changed on eWC. (43)
1.14 [eWC] Voice volume control function doesn't work. (44)
1.15 [eWC] Phone 2 G.168 function can’t be actived on eWC. (45)
1.16 [eWC] VoIP Setting -> Apply to Phone 1,2 on eWC doesn’t work. (29)
1.17 [eWC] Didn’t check RTP end port must large than start port. (36)
1.18 [eWC] Didn’t check Voice VLAN ID in the range from 0 to 4024. (65)

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 still reboot to 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.

CI commands

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

To issue the CI commands, you can either use telnet or console connection, and then go to SMT menu 24.8.

Command Syntax and General User Interface

CI has the following command syntax:

command <*iface* | *device* > **subcommand** [*param*]
command subcommand [*param*]
command ? | help
command subcommand ? | help

General user interface:

1.	?	Shows the following commands and all major (sub)commands
2.	exit	Returns to SMT

System Related Command Home

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

			[0:none/1:log/2:alert/3:both]	
			urlforward [0:none/1:log]	record web forward logs
		clear		clear log
		display	[access|attack|error|ipsec|ike|javablocked|mten|urlblocked|urlforward]	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:none]	mail schedule policy
			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 Home

Command				Description
exit				exit smt menu

Device Related Command Home

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

Ethernet Related Command Home

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

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

POE Related Command [Home]

PPTP Related Command [Home]

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
[Home]

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

<table>
<tr><td></td><td></td><td></td><td></td><td></td><td>No: Delete the oldest half-open session when exceeds the tcp-max-incomplete threshold</td></tr>
<tr><td></td><td></td><td></td><td>block-minute <0~255></td><td></td><td>Only valid when sets 'Block' to yes. The unit is minute</td></tr>
<tr><td></td><td></td><td></td><td>minute-high <0~255></td><td></td><td>The threshold to start to delete the old half-opened sessions to minute-low</td></tr>
<tr><td></td><td></td><td></td><td>minute-low <0~255></td><td></td><td>The threshold to stop deleting the old half-opened session</td></tr>
<tr><td></td><td></td><td></td><td>max-incomplete-high <0~255></td><td></td><td>The threshold to start to delete the old half-opened sessions to max-incomplete-low</td></tr>
<tr><td></td><td></td><td></td><td>max-incomplete-low <0~255></td><td></td><td>The threshold to stop deleting the half-opened session</td></tr>
<tr><td></td><td></td><td></td><td>tcp-max-incomplete <0~255></td><td></td><td>The threshold to start executing the block field</td></tr>
<tr><td></td><td></td><td>set <set#></td><td>name <desired name></td><td></td><td>Edit the name for a set</td></tr>
<tr><td></td><td></td><td></td><td>default-permit <forward|block></td><td></td><td>Edit whether a packet is dropped or allowed when it does not match the default set</td></tr>
<tr><td></td><td></td><td></td><td>icmp-timeout <seconds></td><td></td><td>Edit the timeout for an idle ICMP session before it is terminated</td></tr>
<tr><td></td><td></td><td></td><td>udp-idle-timeout <seconds></td><td></td><td>Edit the timeout for an idle UDP session before it is terminated</td></tr>
<tr><td></td><td></td><td></td><td>connection-timeout <seconds></td><td></td><td>Edit the wait time for the SYN TCP sessions before it is terminated</td></tr>
<tr><td></td><td></td><td></td><td>fin-wait-timeout <seconds></td><td></td><td>Edit the wait time for FIN in concluding a TCP session before it is terminated</td></tr>
<tr><td></td><td></td><td></td><td>tcp-idle-timeout <seconds></td><td></td><td>Edit the timeout for an idle TCP session before it is terminated</td></tr>
<tr><td></td><td></td><td></td><td>pnc <yes|no></td><td></td><td>PNC is allowed when 'yes' is set even there is a rule to block PNC</td></tr>
<tr><td></td><td></td><td></td><td>log <yes|no></td><td></td><td>Switch on/off sending the log for matching the default permit</td></tr>
<tr><td></td><td></td><td></td><td>rule <rule#></td><td>permit <forward|block></td><td>Edit whether a packet is dropped or</td></tr>
</table>

					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

				<port#>	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	firewa ll	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	firewa ll	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 Home

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
	stun			
		status		Display the STUN Client ON/OFF
		display		Display the Stun Clinet infornation
		debug	0:disable 1:keep line 2:nat type 4:nat lifetime 8: status 255: all	
		send	1: shared request | 2: binding request	Select send 1.TCP(no support) or 2.UDP packets
		keepline	1:enable 0:disable	ON/OFF STUN keepline
		t4		RD debug use
		t5		RD debug use
	pr			

PPPoA Related Command Home

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
configu re				

	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 Home

Command				Description
sys	Hdap			
		debug	on or off	
		reset		

Dsp Related Command Home

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]	
toneadjust				
	cadence		cadence <0:Continuous|2> [On Time (ms)] [Off Time (ms)]	
	coeff		coeff <coeff 1> [coeff 2]	
	level		level <Amp>	
	dumpCfg			
	test		test [ID]	
initstatus				Display the DSP initstatus
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
	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
	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
fxop				SI3050 ProSLIC Commands
cc			<command>	
	aer			
	all			Display all of cc command
	assoc		<tcid> <dsp> <chan>	
	bisil		<tcid> <on|off> <thresh> <det_time>	Clear channel mode
	cid_gen		<tcid><call_waiting(on|off)><sdmf|mdmf |euro><calling_number(1-24)><calling_ name(1-15)>	Close channel
	cid_detect		<tcid><CIDMod(na|jp|eu)><call_waiting(on|off) >	
	cid_deact		<tcid>	
	clear		<tcid>	Set coding rate

	close		<tcid>	
	companding		<tcid> <alaw|mulaw|linear|default>	
	connect		<tcid_in> <0=pcm, 1=packet> <tcid_out> <0=pcm, 1=packet>	
	digitm			Download dsp image
	digitr			Display profiles
	disconnect		<tcid_in> <0=pcm, 1=packet> <tcid_out> <0=pcm, 1=packet>	Get dsp version
	disassoc		<tcid>	
	dnld		<dsp> <image_id>	
	dp		[voice|#] <prof#>	Display profiles
	dspver		<dsp>	
	ec		<tcid> <on|off> [nlp_on|nlp_off][nlp_agr <-32768 to 32767>]	Encryption control
			[send_cng_force|send_cng_def][cn_adapt|cn_fix] [noise_lev <noise(dB)>]	encryption_key_def
			[cn_config <0 to 32767>] [4w_on|4w_off] [rst_coeff|keep_coeff]	stat_req_error
			[ec_upd|ec_frz][au_on|au_off][srch_on|srch_frz][mips_on|mips_off]	Fax mode
			[nlp_norm_on|nlp_norm_off][tail_len <tail_len_value(ms)>]	stat_req_fax
			[cfg_bits <hex (default 0x0040)>]	Mute the channel
	ec_coeff		<tcid> <filter:0=background,1=foreground,2=search filter><start idx [0,N-1]> <no_coeffs>	
	ecdbg		<tcid> <clear:0=no,1=yes>	
	encr_ctrl		<tcid> <setup|mac_off> <params>	
	encr_key		<tcid> <rc4|mmh> <tx|rx> <key_str>	
	errstat		<tcid> <clear:0=no,1=yes>	Read dsp memory
	faxm		<tcid>	
	faxstat		<tcid> <clear:0=no,1=yes>	
	faxcallstat		<tcid> <clear:0=no,1=yes>	
	gain_ctrl		<tcid> <tx_dig|na> <rx_dig|na> <tx_analog|na> <rx_analog|na> <rx_s)	
	idle		<tcid>	

	jcid_generate		<tcid> <calling_number>	
	lev		<tcid> <clear:0=no,1=yes>	
	lpcm		<tcid> <0=off, 1=on>	
	lrx		<tcid> <0=off, 1=on>	
	ltx		<tcid> <0=off, 1=on>	
	marks_gain		<#marks: 0-1023> <gain: 0 .. -63 dBm>	
	mfr2		<tcid> <fwd|bck|off>	
	modemm		<tcid> <dir: 0=org, 1=ans>	
	modemr_stat		<tcid> <clear:0=no,1=yes>	
	modemstat		<tcid>	
	mute		<tcid> <on|off>	
	open			
	parse		<tcid>	
	pattern_gen		<tcid> <on|off> <pattern (0xhhhh)]>	
	pcm_pat		<tcid> <0=off,1=on> <pattern: cust|t1|e1> <det_time> <cust_pat>	
	pcm_samp		<tcid> <0=off, 1=on>	
	pktmsg		<0=off,1=on>	
	read_mem		<dsp><0=data,1=prog><32-bit-address> <length in words, 1-49>	
	reconfig		<tcid><rx_timeslot><tx_timeslot><rep_tcid> <dup_tx_tslot>	
	req_gains		<tcid>	
	resample		<tcid> <on|off>	
	rtp_ctrl		<tcid> <code(1:Config Host routing)> <rtp_pload_type (0-127,128:Route none,129:Route all except voice,dtmf,sid>	tone_off
	rxtx		<tcid> <clear:0=no,1=yes>	
	send_rtp_msg		<tcid><eventspace(1or2)><0xbitmap><duration_offset><init_repeat_#><init_repeat_ interval><keep_alive_interval>	Set vad
	show_tcid		<tcid>	

	svpd		<tcid> <on|off (frame_loss_conceal)> <nom_delay[msec]> <max_delay[msec]> <min_delay[msec]> <0|1|2 (adapt playout ctrl: 0=disable,1=adj dur. silence, 2=adj now)>	
	sync		<tcid> <time, [0.5ms]>	
	test_hpi_mem		<dsp> <word offset> <word size>	
	tone			
	toff		<tcid>	
	ton		<tcid> <# tones> <freq1> <amp1> [freq2] [amp2] [freq3] [amp3] [freq4] [amp4> Note: Number of freq, amp to be entered depends on # tones	
	vad		<tcid> <0=off, 1=on>	
	ver			
	vmwi_gen		<tcid> <sdmf|mdmf|euro> <on|off> <send on hook:0=no,1=yes>	
	vmwi_deact		<tcid>	
	voice		<tcid>	
	vpstat		<tcid>	
	xconn		<tcid1> <tcid2> <0=off,1=on> <delay (0-2000ms)> <random loss (0-100%)>	

Voice Related Command Home

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 10 ms)
		rtcpinterval			<index> <ms>	Change the RTCP transmission interval(default 0)
		packetsize			<index>g711<0:10ms|1:20ms|2:30ms>g729<0:10ms|1:20ms|2:30ms>	Change the transmit packetized period(default 20ms)
		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					
		index			<index>	Select SIP index.
		active			<index> <0:off|1:on>	Active/in-active this setting.
		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.
		sessiontimerActive			<index> <0:off|1:on>	Active/in-active sessiontimer
		sessiontimeout			<index> <30-3600 second>	Setup session timeout value(default 300 sec)
		minse			<index> <20-1800 second>	Setup minimum session timeout value.
		retransmitT2			<index> <4|8|16|32sec>	Not support yet
		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> <0-96 chars>	Signaling SIP user-id
		password			<index> <0-96 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|rfc2833like>	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> (40000~65535)	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.
		mwiactive			<index> <0:off|1:on>	Disable or enable the voice message
		mwitimeout			<index> <minute>	Setup mwi expiration time
		rfc3325			<index><1: privacy call using RFC3325, 0: privacy call using draft-01>	Disable or enable the rfc3325
		prack			<index> <0:off|1:on>	Active/in-active prack message
		fakesipactive			<index> <0:off|1:on>	Active/in-active FAKE WAN IP service.
		fakesipservaddr			<index> <ip address>	Setup FAKE WAN IP service IP Address
		fakesipservport			<index> <port>	Setup FAKE WAN IP service port
		outboundactive			<index> <0:off|1:on>	Active/in-active Outbound proxy service.
		outboundaddr			<index> <ip address>	Setup Outbound proxy service. IP Address
		outboundport			<index> <port>	Setup Outbound proxy service. port
		outboundkaactive			<index> <0:off|1:on>	Active/in-active Outbound proxy service keep alive
		outboundkaintvl			<index> <second>	Setup Outbound proxy service keep alive Interval
		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.
		ringbackactive			<index> <0:off|1:on>	Disable or enable early media
		ringbacktone			<index> <tone>	Select early media tone
		musiconholdactive			<index> <0:off|1:on>	Disable or enable music on hold
		musiconholdtone			<index> <tone>	Select music on hold tone
		callfwd			<index> <1-2>	Select call forward table
		mixermode			<index><0: Local / 1: Remote>	Select 3-way conference mixermode
		transafterconf			<index><0:off|1:on>	ON/OFF transfer conference
		rfc3263			<index> <0:off|1:on>	Disable or enable the rfc3263
		featuresenable			<index> <0~1>	ON/OFF feature bits
		Save			<index>	Save the configured value
		display			<index>	Display the configured value.
		dumpCfg				Display working buffer value.
		Free				Free working buffer.
	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
		dialshortinterval			<index> <0~256 sec>	Change the short dialing interval . If the digital interval is smaller than dialshortinterval , the device will call the

						phone number, No matter whether you have pressing finishing or not
		diallonginterval			<index> <0~256 sec>	Change the long dialing interval , if the first digital time over diallonginterval The device will send busy tone
		flashmaxinterval			<index> <0~65535 msec>	Setup flash key max interval, If press flash key interval < fxoflashmax ,device will neglect the flash key and hang up.
		flashmininterval			<index> <0~65535 msec>	Setup flash key min interval. If press flash key interval < flashmininterval ,device will neglect the flash key.
		inputvolume			<index> <-14~14>	Change the input volume gain
		outputvolume			<index> <-14~14>	Change the output volume gain
		receivetonetype			<index> <ir|sir>	Change the receive tone type.
		sipselect			<index><phone port|0:All><0:No|1:Yes> >	Select SIP index and Disable or enable the SIP
		callwaitingtime			<index> <0~128 sec>	Setup call waiting time
		cidtype			<index><0:During Ring |1: Prior Ring>	Call ID display moment
		cidpayload			<index><0:FSK|1:DTMF>	Setup DTMF payload type
		cidfskstartinterval			<index> <0~65535 msec>	Setup FSK start interval. This commands actually generate extra 200ms delay. ex: set to 0 ms -> 200ms set to 200ms -> 400 ms
		ciddtmfstartinterval			<index> <0~65535 msec>	Setup DTMF start interval. This commands actually generate extra 200ms delay. ex: set to 0 ms -> 200ms set to 200ms -> 400 ms
		cidringtimeout			<index> <0~65535 msec>	Setup call id ring time out. This commands actually generate extra 200ms delay. ex: set to 0 ms -> 200ms set to 200ms -> 400 ms
		featuresenable			<index> <0~7>	ON/OFF feature
		save			<index>	Save the configured value
		display			<index>	Display the configured value.
		dumpCfg				Display working buffer value.
		free				Free working buffer.
	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>	Select common index
		save			<index>	Save the configured value.
		ivrsyspermit			<index><0:not permit ivrsys change, 1:permit ivrsys change>	Play ivr sys data
		specialFlag			<index><special flag|h:for help>	bit 0 --> 0: sent rtp after send out 200OK / 1:sent rtp after receive ACK! bit 1 --> 0: respons 200 OK when recieve NOTIFY / 2:follow RFC3265!
		ivrlanguage			<index><0~2>	Sutup ivr language
		pstnfallback			<index><0: Disable PSTN Fallback / 1: Enable PSTN Fallback>	Active/in-active PSTN Fallback Func.
		sipfallback			<index><0: Disable SIP Fallback / 1: Enable SIP Fallback>	Active/in-active SIP Fallback Func.
		dialmethod			<index><0: European [<RR>+Number] / 1: USA [<RR>]	Select dialmethod
		removepound			<index> <0:not removed|1:removed pound>	On/OFF the removed pound
		countrycode			<index><CountryCode|h:for help>	Setup CountryCode
		webenable			<index><0~1>	Disable or enable the web
		display			<index>	Display the configured value.
		dumpCfg			<index>	Display working buffer value.
	autopro					
		index			<index>	Select autopro index.
		active			<index> <0:off|1:on>	Active/in-active autopro.
		servaddr			<index> <ip address>	Setup autopro server IP address
		timeout			<index> <second>	Setup timeout
		retry			<index> <second>	Setup retry time
		protocol			<index> <0:TFTP|1:HTTP|2:HTTPS>	Setup autopro protocol
		method			<index> <0:Common|1:Bluewin>	Select the autopro method
		save			<index>	Save the configured value.
		display			<index>	Display the configured value.
		dumpCfg			<index>	Display working buffer value.
	fxo					
		index			<index>	Select fxo index
		fxolongdial			<index> <Long dial interval(ms)>	Setting DSP first digital to second digial interval time.
		dtmfpausedur			<index> <Short dial interval(ms)>	Setting DSP send each digital interval time.
		dtmfdigitdur			<index> <DTMF Duration(ms)>	DSP sending DTMF duration time through FXO.

<table>
<tr><td></td><td></td><td></td><td></td><td></td><td></td><td>Setup to 0 will be 500ms</td></tr>
<tr><td></td><td></td><td>fxoflashmin</td><td></td><td></td><td><index> <Flash Min Interval(ms)></td><td>Setup fxo flash key min interval. If press flash key interval < fxoflashmin ,device will neglect the flash key.</td></tr>
<tr><td></td><td></td><td>fxoflashmax</td><td></td><td></td><td><index> <Flash Max Interval(ms)></td><td>Setup fxo flash key max interval, If press flash key interval < fxoflashmax ,device will neglect the flash key and hang up.</td></tr>
<tr><td></td><td></td><td>fxolifestable</td><td></td><td></td><td><index> <LifeLine Stable Interval(ms)></td><td>Setup LifeLine stable interval</td></tr>
<tr><td></td><td></td><td>fxophselect</td><td></td><td></td><td><index> <phone port|0:All><0:No|1:Yes></td><td>Select fxo mapping phone</td></tr>
<tr><td></td><td></td><td>save</td><td></td><td></td><td><index></td><td>Save the configured value.</td></tr>
<tr><td></td><td></td><td>display</td><td></td><td></td><td></td><td>Display the configured value.</td></tr>
<tr><td></td><td></td><td>dumpCfg</td><td></td><td></td><td></td><td>Display working buffer value.</td></tr>
<tr><td></td><td>forward</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>index</td><td></td><td></td><td><index></td><td>Select forward index</td></tr>
<tr><td></td><td></td><td>unconditional</td><td></td><td></td><td><index> <phone number></td><td>Setup unconditional call forward number</td></tr>
<tr><td></td><td></td><td>busy</td><td></td><td></td><td><index> <phone number></td><td>Setup busy tone call forward number</td></tr>
<tr><td></td><td></td><td>noanswer</td><td></td><td></td><td><index> <phone number></td><td>Setup noanswer call forward number</td></tr>
<tr><td></td><td></td><td>noanstime</td><td></td><td></td><td><index> <second></td><td>Setup no answer time</td></tr>
<tr><td></td><td></td><td>table</td><td></td><td></td><td><index> <entry_id> <caller> <dest> <type|0:unconditional 1:busy 2:noanswer 3:block 4: accept></td><td>Setup the rule table for call forward.</td></tr>
<tr><td></td><td></td><td>clear</td><td></td><td></td><td><index> <entry | uncond | busy | noans | all > <entry_id for entry></td><td>Clear the call forward rule</td></tr>
<tr><td></td><td></td><td>save</td><td></td><td></td><td><index></td><td>Save the configured value.</td></tr>
<tr><td></td><td></td><td>display</td><td></td><td></td><td></td><td>Display the configured value.</td></tr>
<tr><td></td><td></td><td>free</td><td></td><td></td><td></td><td>Free working buffer.</td></tr>
<tr><td></td><td>version</td><td></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><td></td></tr>
<tr><td></td><td>status</td><td></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></td><td><reset|disp ></td><td>Display convert information or reset convert.</td></tr>
<tr><td></td><td>debug</td><td></td><td></td><td></td><td><phone|ccm|sip|ua|tx|rtp|all></td><td>Turn on debug message.</td></tr>
<tr><td>sip</td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

	tl				<0|1|2|3|...|7>	SIP debug trace level 0:None 1: All 2: Tx 3:Rx 4: TXRx 5: State 6: Error 7: Warning
	siginit					Init SIP SIG task
	sipinit					Init SIP protocol stack
	sipterm					Delete SIP protocol stack
	sigmakecall				type in phone num	Make Call
	sigbusy				Response Busy Call	Response Busy Call
	sigringback				Response Ring	Response Ring
	sigreg				<index>	Register to SIP server
	sigunreg				<index>	Unregister Sip server.
	regstatus				<index>	Show register's information.
	sigok				pick up a call	pick up a call
	sigbye				drop or cancel a call	drop or cancel a call
	sipclose					Close the SIP
	username					
	proxy				[0:off|1:on]	Use SIP proxy
	contact				<LAN Addr:0|Remote Node# WAN Addr:1-2>	
	ackbranch				<on: the Ack bring branch ID, off: the Ack didn't bring branch ID>	Setup bring branch ID.
	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
	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>	Send SUBSCRIBE packet
	sigunmwi				<index>	Send UNSUBSCRIBE packet

	disconnect				<index> or 'all'	Disconnect call.
	txmaxsize					Setup the max tx number
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	
	statistics				<index>	Show the statistics
	linktime				<index>	Show the RTP linktime
autopro						
	active					Active/in-active autopro.
	startnow					Start autopro now
	terminate					Terminate autopro.
	itemdisplay					Display all parameter setting value.
	size					Adjust size of buffer (autoPro.data) which store incoming data from tftp server. .
	start					Start autopro
	status					Show current status of auto provision.
	showdata					Show content of buffer (autoPro.data) which store incoming data from tftp server.
	debug				<0:off|1:on>	on/off debug messages of auto-provision.
	httpdebug				<0:off|1:on>	ON/OFF Autopro httpdebug mode
dialplan						
	clear					Clear dial plan in memory
	dial				dial <phone number>	Simulate dialing digits for dial plan parsing
	load					Load dial plan from flash and overwrite dial plan in memory

	save					Save dial plan to flash
	set				set <all dial plans>	Setup dial plan rule
	show					Show dial plan in detail
	switch					ON/OFF dial plan
	debug					ON/OFF dialplan debug mode
logTest						RD debug command.
tebasic						For TE use only.