



Dynamix DW IAD - 162

VoIP Gateway (H.323)



User Manual

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Preface

About this User's Manual

This user's guide includes specifications, installation guide, web management and command line configuration interface for the Dynamix IAD -162 VoIP Gateway .

Part I: Dynamix IAD-162 Gateway Overview

This part introduces the software/hardware specifications and default settings of the IAD Gateway.

1.1 Overview

Dynamix IAD-161 is a one-port telephone extension and three ports SOHO Router to IP network gateway. It provides Data transfer by 10/100Mbps, telephone services and T.38 fax over IP network with easily operation and configuration. It is most suitable for SOHO and small-to-medium enterprise in Internet communication environment.

Dynamix IAD 162 provides two telephone numbers that one is IP telephone number and the other is PSTN telephone number in one device for end users. You can make phone call via Internet or PSTN in one telephone set now. No more long distance and international telephony fee! Especially, User still can make phone call when external power is failure.

Dynamix IAD also can connect three computers with embedded IP sharing and DHCP server function.

1.2 Software Specifications

Dynamix IAD-162 Gateway Features

- ◆ Provide Voice over IP and Fax over IP features.
- ◆ Built-in NAT/IP sharing function
- ◆ FSK and DTMF Caller ID
- ◆ Transmit Voice and T.38 fax simultaneously
- ◆ Support T.38 ECM function (Error Correction in high speed fax Mode)
- ◆ Provides call progress tone
- ◆ E.164 Common Dial Plan
- ◆ DTMF Dialing
- ◆ Inband/ Out of band DTMF
- ◆ TFTP/FTP software upgrade
- ◆ Remote configuration/reset
- ◆ LED indication for system status
- ◆ Support Static IP, DHCP and PPPoE
- ◆ Set the ring back tone from the IP or local
- ◆ FAX redundancy support
- ◆ RAS and Signal port exchangeable
- ◆ GK id support and GK auto discovery
- ◆ Provide both IP telephone number and PSTN telephone number in one device for end users (IAD 162 only)
- ◆ PSTN backup: user still can make phone call when external power is failure (Dynamix IAD -162 only)

Audio feature

- ◆ Codec: G.711 a/μlaw, G.723.1 (6.3kbps), G.729A, G.729B, G.729AB
- ◆ VAD (Voice Activity Detection)
- ◆ CNG (Comfort Noise Generate)
- ◆ G.168/165-compliant adaptive echo cancellation
- ◆ Dynamic Jitter Buffer
- ◆ Bad Frame Interpolation
- ◆ Voice/DTMF Gain Settings
- ◆ Generate Caller ID (DTMF or FSK)
- ◆ Provide In-band or Out-band DTMF generation/detection
- ◆ Provide Progress tone

System Monitoring

- ◆ System status (WAN, LAN, TEL, Status, Power)

Remote Firmware Upgrade

You can use FTP/TFTP to perform firmware upgrade for the Dynamix IAD-162 Gateway from a remote location.

Security

- ◆ Password protection for system management
- ◆ Built-in NAT function.

Certification

- ◆ CE, FCC

1.3 Hardware Specifications

Chassis

- ◆ 190mm(W) x 110mm(D) x 51.5mm(H)
- ◆ Weight (unit): 0.3 kg

Interface

- ◆ Four 10/100 Base-T Ethernet RJ-45 ports (Auto LAN MDI/MDIX).
- ◆ Input AC 100V-240V, Output DC 12V.
- ◆ One/Two RJ11 Telephone Port (IAD).

Special Housing

- ◆ The plastic housing can be adjustable by manual (Vertical type or Horizontal type)

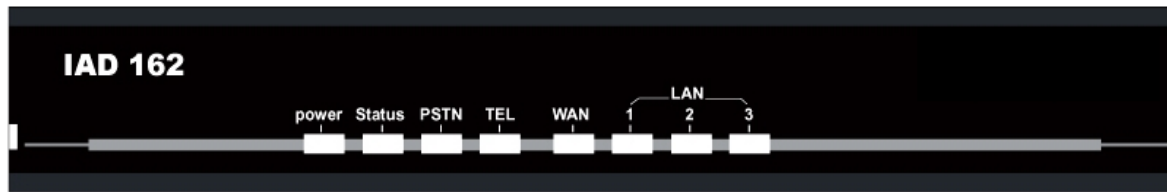
Environment

- ◆ Operational Humidity: 10 to 90 % (Non-condensing)
- ◆ Operational Temperature: 0 to +40 °C
- ◆ Storage Humidity: 10 to 90 % (Non-condensing)
- ◆ Storage Temperature: -10 to +50 °C

Front Panel

The LEDs on the front panel indicate the operational status of the Gateway.





◆ Power (Green):

- (1) Light on: IAD is connected with power adapter correctly and power on.
- (2) Light off: IAD is not connected with power adapter correctly or not power on.

◆ Status (Green):

- (1) Light on: IAD is under Gatekeeper mode and successfully register to Gatekeeper.
- (2) Light off: Under Peer-to-Peer mode.
- (3) Light Blanking: IAD is under Gatekeeper mode and not successfully register to Gatekeeper.

◆ TEL (Orange):

- (1) Light Blinking: IAD IP side has incoming call.
- (2) Light On: IAD IP side is in communication.
- (3) Light Off: IP Line of IAD is in standby mode.

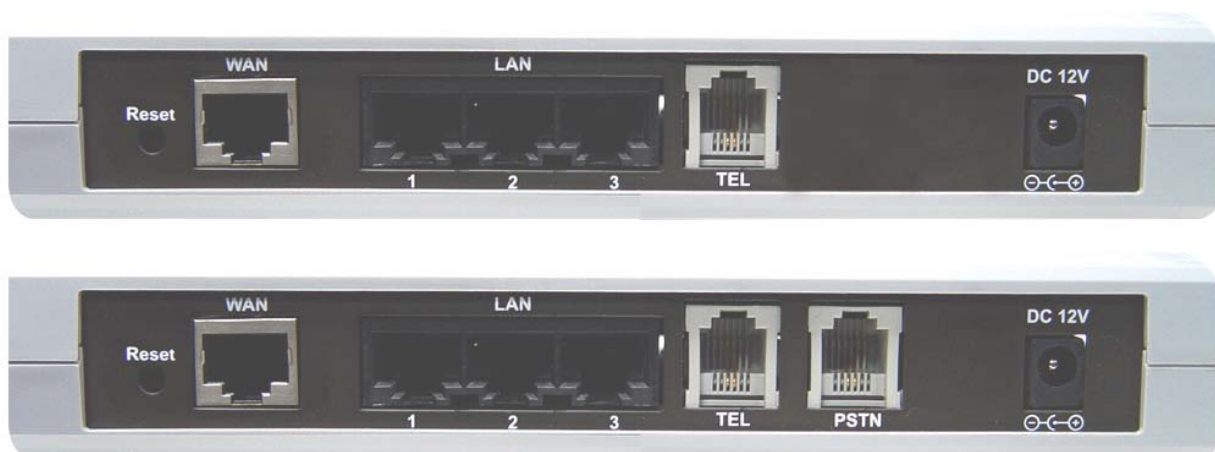
◆ WAN/LAN (Green):

- (1) Light on: Ethernet port successfully connected with network.
- (2) Light Blanking: Ethernet port is transmitting or receiving data.

◆ PSTN (Orange): (IAD 162 only)

- (1) Light Blinking: IAD IP side has incoming call.
- (2) Light On: IAD PSTN side is in communication.
- (3) Light Off: PSTN Line of IAD is in standby mode.

Back Panel

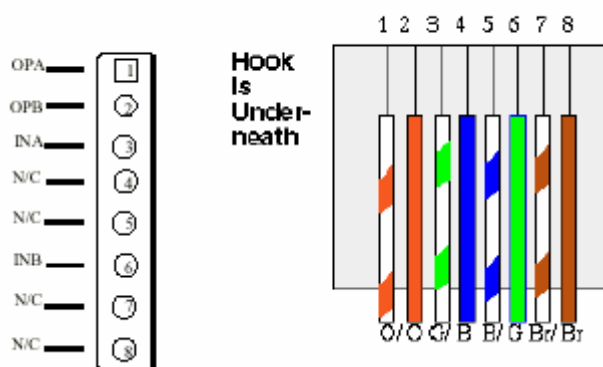


◆ **Reset Button:**

All of the configuration, including network interface configuration, can be back to default value by pressing Reset Button.

◆ **Ethernet Port:**

Ethernet port is for connecting VoIP Gateway to network, transmit rate supports 10/100 Base-T.



Ethernet connector (LAN/WAN)

◆ **TEL Port:**

RJ-11 connector, IAD interface to connect analog phone sets or trunk port of PABX.

◆ **PSTN Port (IAD 162 Only):**

RJ-11 connector, IAD interface to CO PSTN line or extension port of PABX.

◆ **DC 12V Port:**

DC 12V Power supply.

Part II: Start-UP

This part explains how to configure essential and basic items before user can run Dynamix IAD-162 Gateway

2.1 Software Installation Guide

This guide covers all essential configurations under different application, user can follow steps below to configure basic items to run Dynamix IAD-162 Gateway.

2.1.1 Default Settings of Dynamix IAD-162 Gateway

WAN IP Parameters

(1) WAN

- ◆ IP Address = 10.1.1.3
- ◆ Subnet mask = 255.0.0.0
- ◆ Default gateway = 10.1.1.254

(2) LAN

- ◆ IP Address = 192.168.123.123
- ◆ Subnet mask = 255.255.255.0

Telnet and Web Login Password

- ◆ Login User Name= root (or administrator)
Password = "Null" (default)

2.1.2 Additional Installation Requirements

In addition to the contents of your package, there are other hardware and software requirements you need before you can install and use your Dynamix IAD-162 Gateway. These requirements include:

1. A computer with an Ethernet NIC (Network Interface Card) installed.
2. Use Internet Explorer 5.5 or later / Netscape Navigator 6 or later versions.
3. Analog telephone set.
4. Software tools: H323 Gatekeeper (optional)
5. Installation Wizard (optional): This is a configuration tool for users can easy access products and configuring IP address. Please contact with your retailer for more information.

Please follow steps below to access IAD configuration interface:

Step 1. Connect WAN Port of Dynamix IAD-162 Gateway to public network

Connect the WAN port (silver) on the Dynamix IAD-162 Gateway to the Ethernet port of your network (e.g. Cable Modem, ADSL Modem) using the standard CAT-5 straight Ethernet cable.

Step 2. Connect your PC to the LAN port of IAD

Connect your PC to the LAN port of IAD with standard CAT-5 straight Ethernet cable.

Step 3. Set your PC as DHCP mode

Please go to the network setting of your PC and set it as DHCP mode, let your PC can automatically search for DHCP server and get one valid IP address. IAD has embedded DHCP server (default is enabled) so that your PC will get one IP address from IAD DHCP server.

Step 4. Open your browser and input IP address 192.168.123.123

Once your PC has got an IP from IAD, you may connect IAD via WEB browser to do more configurations. The default LAN IP address of IAD is "192.168.123.123"; please input this IP address on web browser to connect with web management interface. Please refer to Part III Web management for more information.

Step 5. Advanced Setting via Telnet (Optional)

If user wants to do more advanced and complete settings that cannot be found via web management interface, please Telnet to the IAD to do more detail configurations.

Step 6. Connect other PC with LAN Ports (Optional)

If you have more than one PC, you can connect them with LAN Ports (black) on the Dynamix IAD-162 Gateway. Please set these PCs as DHCP mode so that they can automatically get IP from IAD DHCP server. DHCP server of IAD can assign 253 IP most.

Caution:

To prevent damage to the Dynamix IAD-162 Gateway, please make sure you have connected with the correct power adapter.

2.1.3 Essential Configuration via Web Management interface

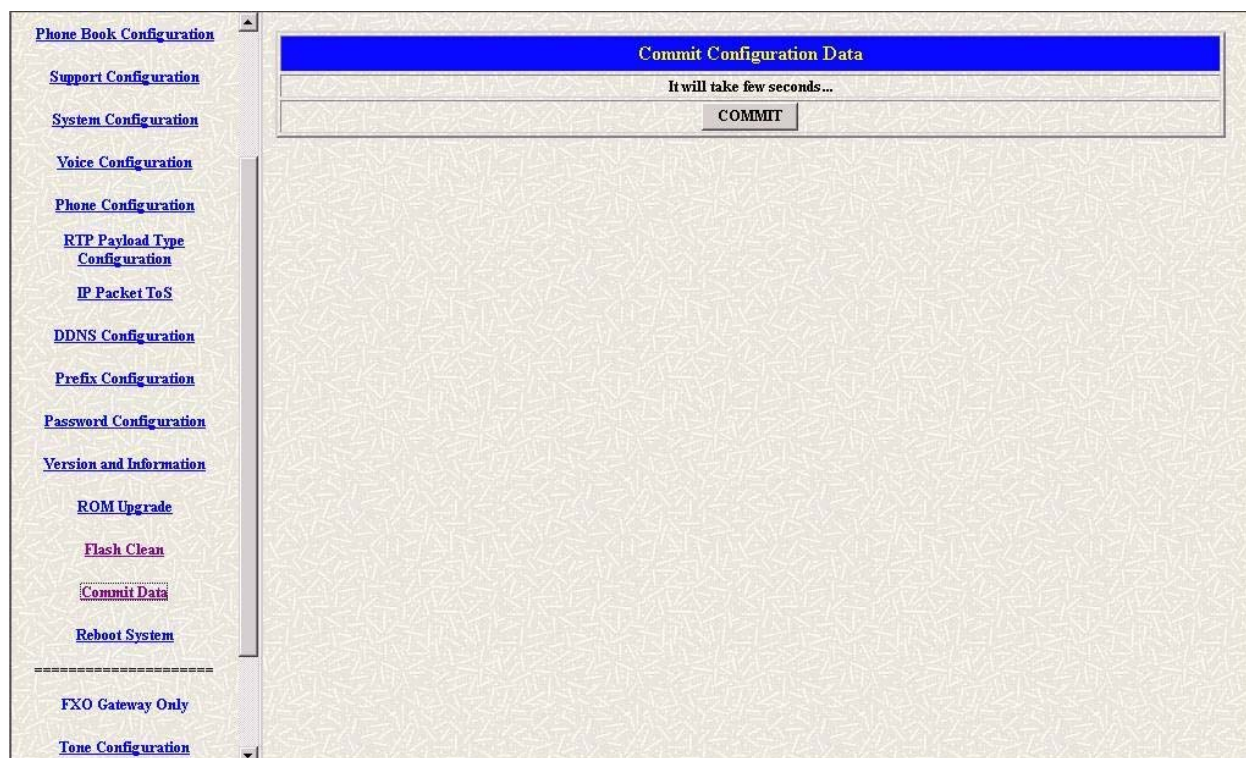
This section describes how to setup IAD via Web management interface. Please follow procedures below to configure essential items before you use Dynamix IAD-162 Gateway.

2.1.3.1 Save Data and Reboot

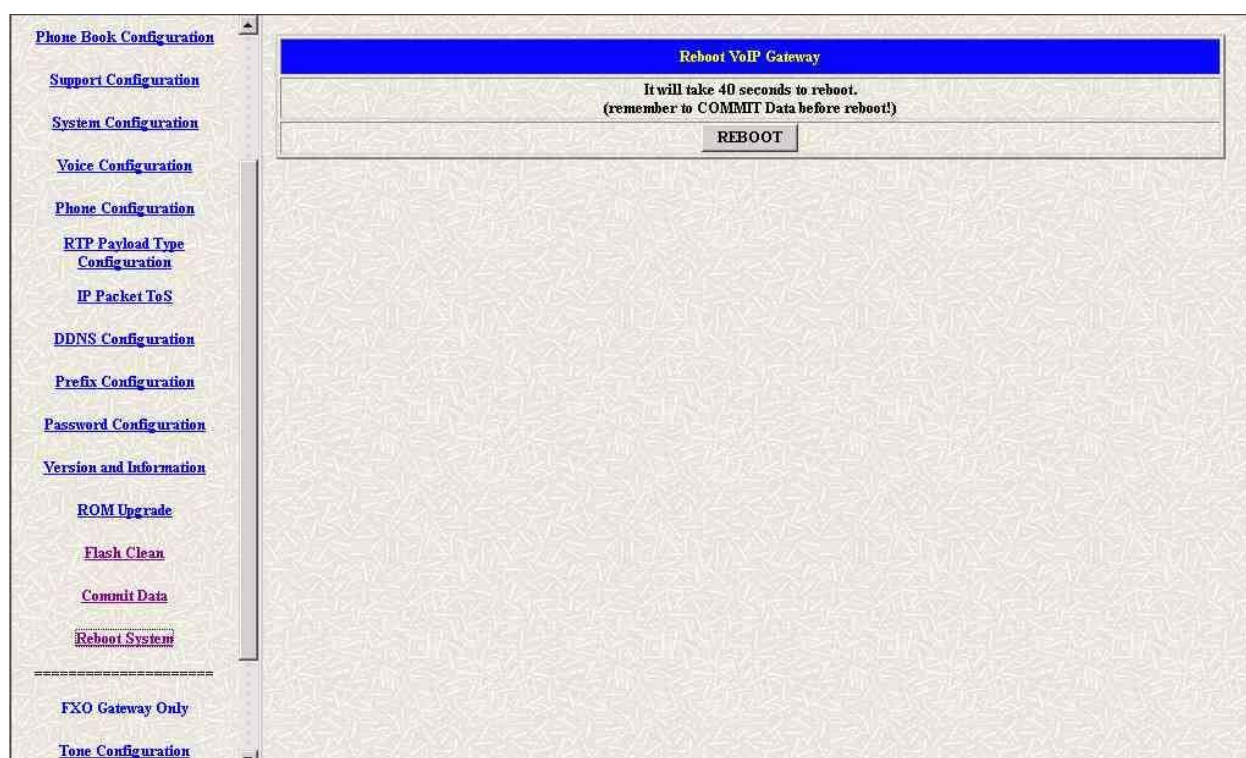
After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Click [Commit Data] on the navigation panel. In the Commit Configuration

Data screen, click the [Commit] button. In the Commit Configuration Data screen will Display [Commit to Flash OK!], when IAD finished committing data.



Step 2. Click [Reboot System] on the navigation panel. In the Dynamix IAD-162 Gateway screen, click the [Reboot] button. It will take around 40 seconds to reboot.



Step 3. Close the current browser windows and launch your web browser again.

2.1.3.2 Setup Network

(1) Fixed IP

To configure the VoIP Gateway IP address, please click [Network Interface] on the navigation panel. In the Network Interface screen, type a new IP address, subnet mask and the default routing gateway (e.g. IP Address: 192.168.13.62, Subnet mask: 255.255.248.0, Default routing gateway: 192.168.8.254) and click the OK button.

The screenshot displays the 'Network Interface' configuration screen. On the left is a navigation menu with options like 'Network Interface', 'H323 Configuration', 'Line Configuration', etc. The main area contains the following fields:

Network Interface	
IP Mode:	<input checked="" type="radio"/> Static <input type="radio"/> DHCP <input type="radio"/> PPPoE
IP Address:	192 . 168 . 13 . 62
Subnet Mask:	255 . 255 . 248 . 0
Default Routing Gateway:	192 . 168 . 8 . 254
DHCP DNS primary:	
DHCP server:	
DHCP Server switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
NAT function switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
LAN Port IP for NAT:	192 . 168 . 88 . 88
DNS primary:	168 . 95 . 192 . 1
DNS secondary:	168 . 95 . 1 . 1
HTTP Port:	80
SNTP:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

(2) DHCP

Click [Network Interface] on the navigation panel. In the Network Interface screen, enable the DHCP function if you are using the cable modem or DHCP server and click the [OK] button.

The screenshot displays the 'Network Interface' configuration page. On the left is a navigation menu with options like 'VoIP Gateway Configuration Menu', 'Network Interface', 'H323 Configuration', 'Line Configuration', 'Phone Book Configuration', 'Support Configuration', 'System Configuration', 'Voice Configuration', 'Phone Configuration', 'RTP Payload Type Configuration', 'IP Packet ToS', 'DDNS Configuration', 'Prefix Configuration', 'Password Configuration', 'Version and Information', 'ROM Upgrade', and 'Flash Clean'. The main area is titled 'Network Interface' and contains the following fields:

IP Mode:	<input type="radio"/> Static <input checked="" type="radio"/> DHCP <input type="radio"/> PPPoE
IP Address:	192 . 168 . 15 . 205
Subnet Mask:	255 . 255 . 248 . 0
Default Routing Gateway:	192 . 168 . 8 . 254
DHCP DNS primary:	
DHCP server:	
DHCP Server switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
NAT function switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
LAN Port IP for NAT:	192 . 168 . 88 . 88
DNS primary:	168 . 95 . 192 . 1
DNS secondary:	168 . 95 . 1 . 1
HTTP Port:	80
SNTP:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

(3) PPPoE

Configuring the VoIP Gateway IP address for PPPoE Mode. Click [Network Interface] on the navigation panel. In the Network Interface screen, Select IP mode: PPPoE, and put the info of the PPPoE User Name, password, and Reboot After Remote Host Disconnection: Enable (e.g. User: 123456@hinet.net, password: 123456) and click the [OK] button.

The image displays two screenshots of the Dynamix IAD-162 Gateway configuration interface, specifically the 'Network Interface' screen.

Top Screenshot: The 'Network Interface' screen is shown. The 'IP Mode' is set to 'PPPoE' (highlighted with a red box). Other settings include IP Address: 192.168.15.205, Subnet Mask: 255.255.248.0, Default Routing Gateway: 192.168.8.254, DHCP DNS primary, DHCP server, DHCP Server switch: Enable, NAT function switch: Enable, LAN Port IP for NAT: 192.168.88.88, DNS primary: 168.95.192.1, DNS secondary: 168.95.1.1, HTTP Port: 80, SNTP: Enable, SNTP Server Address: 168.95.195.12, GMT: 8, and IP Sharing: Disable.

Bottom Screenshot: The 'Network Interface' screen is shown again. The 'IP Change' is set to 'Disable'. The 'PPPoE User Name' is set to '123456@hinet.net' and the 'PPPoE Password' is set to '*****' (both highlighted with a red box). Other settings include SNTP Server Address: 168.95.195.12, GMT: 8, IP Sharing: Disable, IP Sharing Server Address: 210.59.163.198, PPPoE IP Address, PPPoE Destination, PPPoE DNS primary, Reboot After Remote Host Disconnection: Enable, Send PPPoE Echo Request: Disable, EMS IP(except 37xx): 1.1.1.1, EMS User Name(except 37xx): 111, EMS Password(except 37xx): **, and EMS Time(except 37xx): 0. The 'OK' button is visible at the bottom right.

- **PPPoE User Name:** Set PPPoE authentication User Name.
- **PPPoE Password:** Set PPPoE authentication password.
- **Reboot After Remote Host Disconnection:** Enable/Disable auto reboot after PPPoE disconnection.

If user enables this function, after PPPoE being disconnected, IAD will automatically reboot to re-connect, and after rebooting, if IAD still can't get contact with server, IAD will keep trying to connect. After re-connected, IAD will also restart system. On the other hand, if user disables this function, IAD won't reboot and keep trying to connect.

- **Other PPPoE items:** for reference only, cannot allow to be configured.

2.1.3.3 Application mode-GK/Peer-to-Peer Mode

After setting IP address, user must assign IAD to work under GK routed mode or Peer-to-Peer mode. If there is no Gatekeeper, please set your IAD as Peer-to-Peer Mode.

2.1.3.3.1 GK routed mode

GK mode means that there will be an intermediate Gatekeeper between Dynamix IAD-162 Gateway and the remote entity. While operating at this mode, Dynamix IAD-162 Gateway will first register to the Gatekeeper located at the ISP side.

Step 1. Configure the Dynamix IAD-162 Gateway H323 Configuration. Click [H323 Configuration] on the navigation panel. In the H323 Information screen, select GK routed Mode function.

Step 2. Set the H323 information from your service provider: Gatekeeper IP Address, Gateway Type, Registered Prefix, Line1 Number, and click the [OK] button.

H323 Configuration	
Mode:	<input checked="" type="radio"/> GK routed <input type="radio"/> Peer-to-Peer
Gatekeeper IP:	192.168.13.77
2nd Gatekeeper IP:	10.1.1.2
Default Gateway IP:	x
Gateway Type:	<input type="radio"/> Terminal <input checked="" type="radio"/> Gateway1 <input type="radio"/> Gateway2
Registered Prefix:	79811
Line1/TEL1 Number:	798111
Line2/LINE1 Number:	
Line3/TEL2 Number:	
Line4/LINE2 Number:	
Line5 Number:	
Line6 Number:	
Registered Alias:	798111234
Display Information:	IFXS
Gatekeeper Discovery:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

2.1.3.3.2 Peer-to-Peer Mode

Peer-to-Peer Mode allows users to call other VoIP devices without the Gatekeeper. When in Peer-To-Peer mode, Dynamix IAD-162 Gateway use Phone Book, which will dial predefined phone number, and press “#” (optional, to accelerate the dial) as end of dial.

To configure Peer-To-Peer Mode in Dynamix IAD-162 Gateway, follow the steps below:

Step 1. Configure the Dynamix IAD-162 Gateway H323 Configuration. Click [H323 Configuration] on the navigation panel. In the H323 Configuration screen, select Peer-to-Peer Mode function, set Line1 number and click the [OK] button.

H323 Configuration	
Mode:	<input type="radio"/> GK routed <input checked="" type="radio"/> Peer-to-Peer
Gatekeeper IP:	10.1.1.1
2nd Gatekeeper IP:	10.1.1.2
Default Gateway IP:	x
Gateway Type:	<input type="radio"/> Terminal <input checked="" type="radio"/> Gateway1 <input type="radio"/> Gateway2
Registered Prefix:	x
Line1/TEL1 Number:	798111
Line2/LINE1 Number:	
Line3/TEL2 Number:	
Line4/LINE2 Number:	
Line5 Number:	
Line6 Number:	
Registered Alias:	1fxs-001a87
Display Information:	1FXS
Gatekeeper Discovery:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

Step 2. Configure Phone Book in the IAD Gateway. Click [Phone Book] on the navigation panel. In the Phone Book screen, enter the Index, Name, IP address and e164 (phone number) of the destination and click the Add Data button.

Phone Book						
Index	Name	E164	IP Address	Port	Drop	Insert

New Record						
Index	Name	E164	IP Address	Port	Drop Prefix	Insert Prefix
					<input checked="" type="radio"/> Disable <input type="radio"/> Enable	
Add Data		Delete Data				

2.1.4 Essential Configuration via Telnet Command Line interface

This section describes how to setup IAD via Telnet command line interface. Please follow procedures below to configure essential items before you use Dynamix IAD-162 Gateway.

2.1.4.1 Save Data and Reboot

After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Confirm the changed configurations, input [commit] and press [enter] key to save it.

Step 2. Input [reboot] then press [enter] key to restart Gateway.

Step 3. After around 40 seconds, Gateway will take effect in new configurations.

Do not turn off your Gateway or remove the Gateway while saving your configuration.

2.1.4.2 Setup Network

Use command [ifaddr] to configure Gateway IP Address and related information.

(1) Static IP

```
usr/config$ ifaddr -mode 0
usr/config$ ifaddr -ip 192.168.1.11 -mask 255.255.255.0 -gate 192.168.1.254
```

In this case is to configure Gateway IP Address as [192.168.1.11], subnet mask as [255.255.255.0], default router gateway as [192.168.1.254].

(2) DHCP

```
usr/config$ ifaddr -mode 1
```

In this case is to enable DHCP mode of IAD, once IAD reboot system, it will automatically capture IP from DHCP server.

(3) PPPoE

Step 1. To Set PPPoE mode, please use [pppoe] command:

```
usr/config$ ifaddr -mode 2
```

Step 2. Input the user id & password provided by your ISP:

```
usr/config$ ifaddr -id 84460791@hinet (PPPoE login account)
usr/config$ ifaddr -pwd 123 (PPPoE login Passowd)
```

Step 3. Commit and reboot IAD.

```
usr/config$ commit
usr/config$ reboot
```

Step 4. When IAD successfully establish PPPoE connection, use command [ifaddr -print] to see detail information.

For example:

```
usr/config$ ifaddr -print
IP mode : PPfPoE
PPPoE adapter information
    Status                : Ready
    IP address             : 61.223.39.135
    Destination            : 61.223.128.254
    DNS primary             : 168.95.192.1
    Subnet Mask             : 255.255.255.255
    Authenticate           : PAP
    Protocol               : TCP/IP
    Device                 : PPP/PPPoE
    DHCP server switch      : Enable
    NAT function switch     : Enable
    LAN port IP (for NAT)   : 192.168.123.123
    DNS primary             : 168.95.192.1
    DNS secondary           : 168.95.1.1
    HTTP port              : 80
    SNTP                    : mode=1
                           server 168.95.195.12
                           time zone : GMT+8
                           cycle=1024 mins
    IPSharing               : no IPSharing device.
    IP change               : Disable
    PPPoE user name         : 84460791@hinet.net
    PPPoE password          : *****
    PPPoE reboot            : Yes
    PPPoE echo              : Enable
    EMS IP                  : 192.168.10.71
    EMS user name           : vwusr
    EMS password            : *****
    EMS time                : 0
```


2.1.4.3 Application mode-GK/Peer-to-Peer Mode

After setting IP address, user must assign IAD to work under GK routed mode or Peer-to-Peer mode. If there is no Gatekeeper, please set your IAD as Peer-to-Peer Mode.

2.1.4.3.1 GK routed mode

GK mode means that there will be an intermediate Gatekeeper between Dynamix IAD-162 Gateway and the remote entity. While operating at this mode, Dynamix IAD-162 Gateway will first register to the Gatekeeper located at the ISP side.

Step 1. Set GK routed Mode, using “h323” command

```
usr/config$ h323 -mode 0
```

Mode 0 is for GK mode, while mode 1 is for Peer-to-Peer mode.

Step 2. Set the H323 information from your service provider: Gatekeeper IP Address, Gateway Type, Registered Prefix, Line1 Number, and click the [OK] button.

For example:

```
usr/config$ h323 -gk 192.168.13.77
usr/config$ h323 -gwtype 1
usr/config$ h323 -prefix 798
usr/config$ h323 -line1 7981
```

2.1.4.3.2 Peer-to-Peer Mode

Peer-to-Peer Mode allows users to call other VoIP devices without the GK server. When in Peer-To-Peer mode, IAD Gateway use Phone Book, which will dial predefined phone number, and press “#” (optional, to accelerate the dial) as end of dial.

To configure Peer-To-Peer Mode in Dynamix IAD-162 Gateway, follow the steps below:

Step 1. Set Peer-To-Peer Mode, using “h323” command

```
usr/config$ h323 -mode 1
```

Mode 0 is for GK mode, while mode 1 is for Peer-to-Peer mode.

Step 2. Configure Phone Book, using “pbook” command.

```
usr/config$ pbook -add name TEST1 ip 10.1.1.1 e164 10
```

In this case user add one callee record named as TEST1, IP address as 10.1.1.1, and mapping e.164 number as 10. After phone book data has been set, user can dial 10 to make a call for IP 10.1.1.1.

After the command completed, you can type “pbook -print” to see if the input record is correct.

When adding a record to Phone Book, user does not have to reboot the machine, and the record will be effective immediately.

2.1.5 Essential Configuration via Installation Wizard

Installation Wizard is a friendly software tool that can provide you an easy way to configure your VoIP devices. You only need to Input the MAC address of your product and Click [Search Device]; you can configure your VoIP device without changing your PC's setting.

Additionally, when you forget IP address of the VoIP device, Installation Wizard gives you a solution to solve this problem.

For more information, please refer to the Installation Wizard user manual.

2.1 Special Housing Installation Guide

Dynamix IAD 162 has special adjustable housing for vertical or horizontal type. Please follow procedures as below to change type you like.

2.1.1 Horizontal Type

2.1.1.1



Insert stand board on one side.

2.1.1.2



Insert the other stand board on the other side.

2.1.1.3



Finally IAD can stand as horizontal type.

2.1.2 Vertical Type

2.1.2.1



Insert stand board on one side.

2.1.2.2



Insert the other stand board on the same side.

2.1.2.3



Finally IAD can stand as vertical type.

Part III: Special Applications and Features

This part explains how to configure IAD Gateway under special application mode, such as behind NAT, and how to upgrade firmware.

3.1 Behind IP-Sharing

3.1.1 IP Sharing Configuration

3.1.1.1 One Dynamix IAD-162 Gateway behind IAD

This application is only for the user who is using the IP Sharing device. It is said Gateway is connected behind IP Sharing. The IP Sharing Device must support the DMZ or Virtual server functions such as ADSL network.

- Step 1.** The WAN IP Address obtained from ADSL has two kinds of methods. One is fixed IP Address, while user applies for one or more fixed IP Addresses. Another is dynamic IP Address while user applies for dial-up connection way. Only when the IP address is fixed user can put IAD behind NAT device.
- Step 2.** The LAN IP Address of User's PC can be set as DHCP client in order to gain a valid one.
- Step 3.** One can also assign a fixed IP address, which belongs to the same network segment as the LAN interface of IP Sharing device.
- Step 4.** IAD Gateway must enable the IP Sharing function for the fixed / dynamic WAN IP Address.

Note:

With Dynamic WAN IP Address, a valid GK for Dynamix IAD-162 Gateway to get register on is necessary. In other words, it is not workable in Peer-to-Peer mode while dynamic WAN IP Address.

- Step 5.** IP Sharing device must have a function to do IP/Port mapping. Some is named as DMZ, some is named as virtual server. The VoIP messages from WAN have to completely pass forward to the LAN. It mean that if the Dynamix IAD-162 Gateway is assigned a virtual fixed IP Address such as 192.168.1.5, IP Sharing device must forward the VoIP message to 192.168.1.5.

Please see following for example:

>Advanced setting > NAT setting > DMZ Host setting

DMZ Host setting

Activate DMZ
DMZ Host IP: 192.168.1.5

Step 6. Configure the Dynamix IAD-162 Gateway IP address for IP Sharing Mode. Click [Network Interface] on the navigation panel. In the Network Interface screen, enter the IP address, Subnet mask and the default gateway in the network table. Please follow up your IP Sharing device

Step 7. Enable the IP sharing function and input the static IP address in the IP Sharing server address (e.g. 210.59.163.198) and click the OK button.

The screenshot displays the 'VoIP Gateway Configuration Menu' on the left sidebar with 'Network Interface' selected. The main configuration area shows various settings. A red box highlights the 'IP Sharing' section, which includes:

- IP Sharing:** ☒ Enable ☐ Disable
- IP Sharing Server Address:** 210 . 59 . 163 . 198
- IP Change:** ☐ Enable ☒ Disable

Other visible settings include:

- SNTP Server Address:** 168 . 95 . 195 . 12
- GMT:** 8
- PPPoE User Name:** 123456@hinet.net
- PPPoE Password:** *****
- PPPoE IP Address:** [Empty field]
- PPPoE Destination:** [Empty field]
- PPPoE DNS primary:** [Empty field]
- Reboot After Remote Host Disconnection:** ☒ Enable ☐ Disable
- Send PPPoE Echo Request:** ☐ Enable ☒ Disable
- EMS IP(except 37xx):** 1 . 1 . 1 . 1
- EMS User Name(except 37xx):** 111
- EMS Password(except 37xx):** **
- EMS Time(except 37xx):** 0

An 'OK' button is located at the bottom right of the configuration area.

Step 8. Click [Commit Data] on the navigation panel. In the Commit Configuration Data screen, click the Commit button. In the Commit Configuration Data screen will Display [Commit to Flash OK!], when IAD finished committing data.

Step 9. Click [Reboot System] on the navigation panel. In the VoIP Gateway screen,

click the [Reboot] button. It will take around 40 seconds to reboot.

Step 10. Close the current browser windows and launch your web browser again.
Enter the new IP address in the Location or Address field.

3.2 NAT mode (PPPoE)

- Step 1.** Set PPPoE mode, using [ifaddr -mode 2], input the user id & password provided by your ISP, using [ifaddr -id -pwd], reboot the device after disconnection, using [ifaddr -reboot 1]

```
usr/config$ ifaddr -mode 2
usr/config$ ifaddr -id 123@hinet.net (PPPoE login account)
usr/config$ ifaddr -pwd 123 (PPPoE login Passowd)
usr/config$ ifaddr -reboot 1 (Enable)
```

- Step 2.** Set NAT function (Default NAT function is enable)

```
usr/config$ ifaddr -nat 1
```

- Step 3.** Set DHCP server function (Default DHCP server function is enable)

```
usr/config$ ifaddr -dhcpsv 1
```

For example:

```
usr/config$ ifaddr -print

IP mode : PPPoE
PPPoE adapter information
  Status           : Ready
  IP address        : 61.223.39.135
  Destination       : 61.223.128.254
  DNS primary       : 168.95.192.1
  Subnet Mask       : 255.255.255.255
  Authenticate      : PAP
  Protocol          : TCP/IP
  Device            : PPP/PPPoE
  DHCP server switch : Enable
  NAT function switch : Enable
  LAN port IP (for NAT) : 192.168.123.123
  DNS primary       : 168.95.192.1
  DNS secondary     : 168.95.1.1
  HTTP port         : 80
  SNTP              : mode=1
                   server 168.95.195.12
```

```
time zone : GMT+8
cycle=1024 mins
IPSharing      : no IPSharing device.
IP change      : Disable
PPPoE user name : 84460791@hinet.net
PPPoE password  : *****
PPPoE reboot    : Yes
PPPoE echo      : Enable
EMS IP          : 192.168.10.71
EMS user name    : vwusr
EMS password     : *****
EMS time        : 0

usr/config$
```

Step 3. When Gateway connection succeed. **Setup PC use LAN IP connection Network**

Select [Specify an IP Address] and enter [192.168.123.xxx] in the [IP Address] location (where xxx is a number between 2 and 254 used by the VoIP Gateway to identify each computer), and the default [Subnet Mask 255.255.255.0]. Please notice that two computers on the same LAN cannot have the same IP address. Set Default Gateway value as 192.168.123.123 in the [new gateway] field. Then save your change. PC can also use DHCP mode when DHCP server of IAD is enabled.

3.3 Upgrade Your IAD

3.3.1 Upgrade via Web management interface

3.3.1.1 Before start

Step 1. Please confirm Host PC, which is installed as TFTP / FTP server and is in available network.

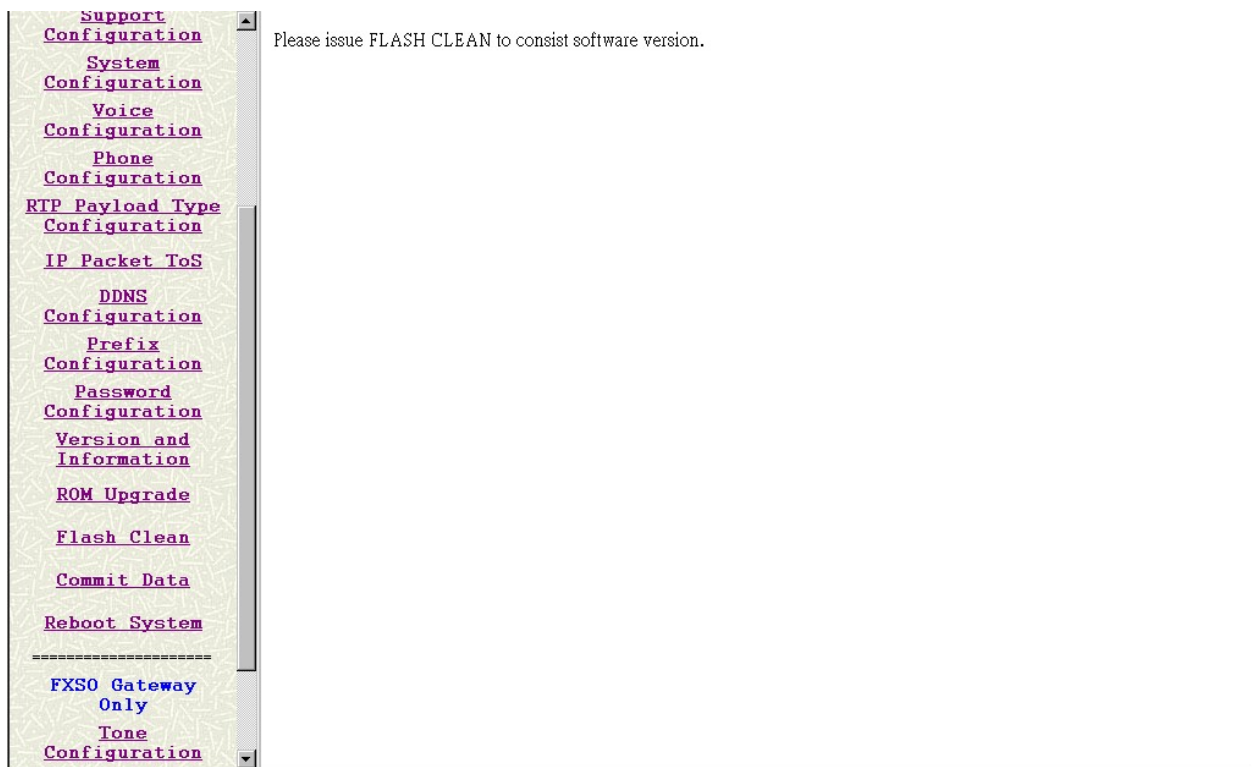
Step 2. Note down your current configurations, such as [H323 Configuration], [Phone Book].

3.3.1.2 Upgrade Version

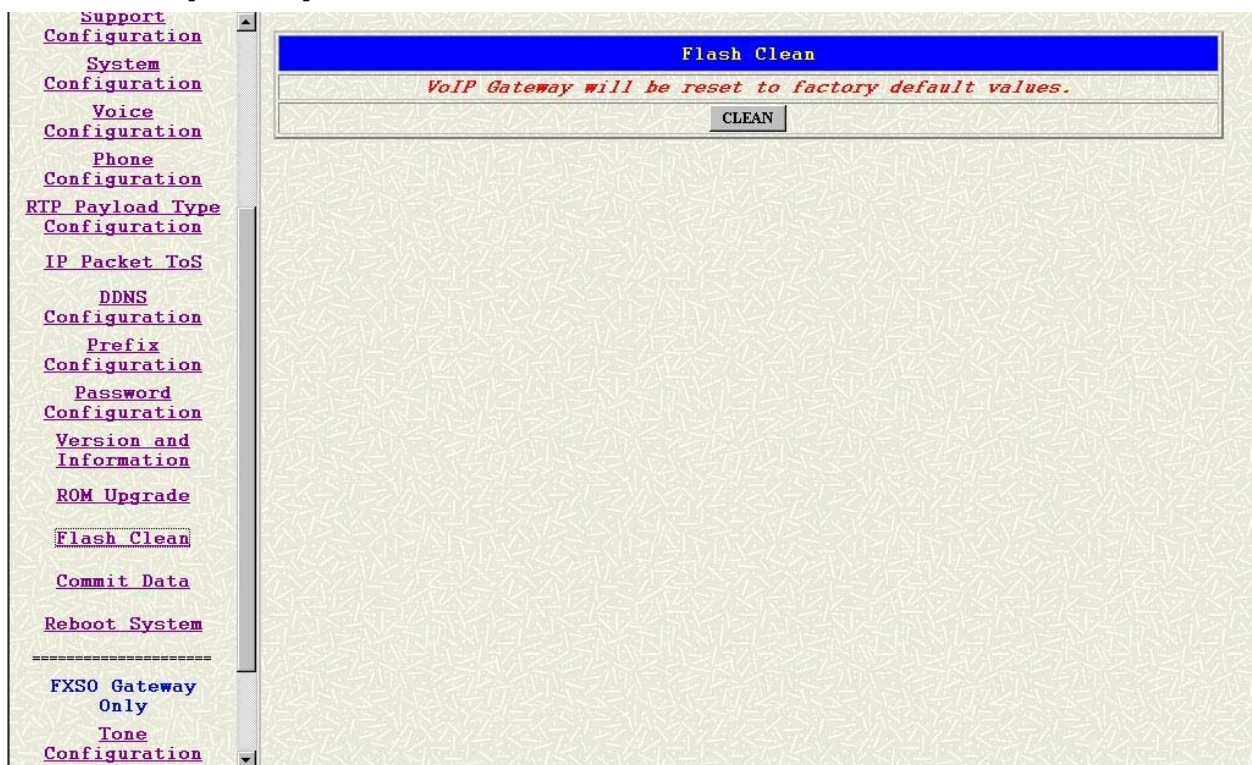
Step 1. To update the Dynamix IAD-162 Gateway ROM Version, please click [ROM Upgrade] on the navigation panel. In the [ROM Configuration] screen, type TFTP/FTP Server IP address, Target File Name, Method, Target File Type (e.g. Server IP Address: 192.168.13.88, Target File Name: IAD162.100, Method: TFTP, Target File Type: Application image) and click the [OK] button.

The screenshot shows the 'ROM Configuration' web interface. On the left is a navigation menu with the following items: VoIP Gateway Configuration Menu (highlighted in green), Network Interface, H323 Configuration, Line Configuration, Phone Book Configuration, Support Configuration, System Configuration, Voice Configuration, Phone Configuration, RTP Payload Type Configuration, IP Packet ToS, DDNS Configuration, Prefix Configuration, Password Configuration, Version and Information, ROM Upgrade (highlighted in red), and Flash Clean. The main content area is titled 'ROM Configuration' and contains the following fields: 'TFTP/FTP server IP Address' with four input boxes containing '192', '168', '13', and '88'; 'Target File name' with a text box containing 'IAD162.100'; 'Method' with a dropdown menu set to 'TFTP'; 'FTP Login' with 'name' and 'passwd' text boxes; 'Target File Type' with a dropdown menu set to 'Application Image'; and an 'OK' button at the bottom right.

Step 2. After upgrade finished, on screen will display [Please issue FLASH CLEAN to consist software version.] information.



Step 3. Click [Flash Clean] on the navigation panel. In the Flash Clean screen, click the [CLEAN] button.



Step 4. In the Flash Clean screen to Display [Flash cleaned!! Please reboot your system!!], when Flash Clean Ok.

Step 5. Click [Reboot System] on the navigation panel. In the Reboot Dynamix IAD-162 Gateway screen, click the [Reboot] button. It will take 40 seconds to reboot.

Step 6. Close the current browser windows and launch your web browser again.
Enter the IP address in the Location or Address field.

3.3.2 Upgrade via Telnet Command interface

Use [rom] command to upgrade software of IAD.

```
usr/config$ rom
```

ROM files updating commands

Usage:

```
rom [-print][-app][-boot][-boot2m][-dsptest][-dspcore][-dspapp]
    [-s TFTP/FTP server ip][-f filename][-method used]
    [-ftp [username][passwd]]
```

```
rom -print      Show versions of rom files (optional).
  -app          Update main application code (optional).
  -boot         Update main boot code (optional).
  -boot2m       Update 2M code (optional).
  -dsptest      Update DSP testing code (optional).
  -dspcore      Update DSP kernel code (optional).
  -dspapp       Update DSP application code (optional).
  -s            IP address of TFTP/FTP server (mandatory).
  -f            File name (mandatory).
  -method       Download via TFTP/FTP (0=TFTP, 1=FTP)
  -ftp          specify username and password for FTP.
```

Note:

This command can run select one option in 'app', 'boot', 'dsptest', 'dspcore', and 'dspapp'.

Example:

```
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
```

```
usr/config$
```

Parameter Usages:

-print: show versions of all rom files.

-app, boot, boot2m, dsptest, dspcore, dspapp, ht: To update main Application program code, Boot code, DSP testing code, DSP kernel code, or DSP application code, and Hold Tone file.

Note:

Most of all, the Rom file needed to get upgrade is App or Boot2m. Please check the exact Rom file before doing download procedure.

-s: To specify TFTP server's IP address when upgrading ROM files.

-f: To specify the target file name, which will replace the old one.

-method: To decide using TFTP or FTP as file transfer server. [0] stands for TFTP, while [1] stands for FTP.

-ftp: If users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

For example:

```
usr/config$ rom -print
```

```
Download Method : TFTP
Boot Rom       : sdboot.200
Application Rom : IAD_1213.bin
DSP App        : 48302ce3.140
DSP Kernel     : 48302ck.140
DSP Test Code  : 483cbit.bin
```

```
usr/config$
```

After software like application has been upgraded, please execute [flash -clean] to clear old configurations and make upgrade complete. This will keep all configurations under [ifaddr].

```
usr/config$ flash -clean
```


3.4 Dynamix IAD 162 PSTN Line Application

3.4.1 PSTN Outgoing Call

3.4.1.1 Make PSTN call from TEL Phone set

Default TEL line is intended to make IP call, if user wants to make a call via PSTN line, please dial “* #”, then user will hear dial tone from PSTN side. The dial tone between IP side and PSTN are different. If the PSTN port doesn't connect a PSTN line, press “*#”, the signal will be back to IP side then play busy tone.

3.4.1.2 PSTN Backup

If IAD is working under GK mode, and fail to register to Gatekeeper, TEL line will automatically switch relay to PSTN line, which means when VoIP system is failed, user can still make a call from PSTN line without pressing “*#”.

3.4.2 PSTN Incoming Call

3.4.2.1 VoIP TEL is busy

Caller from PSTN side can still make a call to IAD, PSTN LED will be blanking, when user hangs up IP call, TEL phone set will ring, and user can pick up the call from PSTN side. The Dynamix IAD-162 Gateway has “call waiting tone” function, when IP side is under calling, if PSTN side has an incoming call, the user in IP side will hear a call waiting tone.

3.4.2.2 VoIP TEL is available

When IAD has incoming call from PSTN side, TEL phone set will ring, user can pick the call from PSTN side.

3.4.3 VoIP Outgoing Call

Default TEL line is intended to make IP call, user can just pick up the phone set connected with TEL port and make VoIP call while VoIP network system is available.

3.4.4 VoIP Incoming Call

When user is communicating with PSTN side with TEL Phone set, IAD can't have VoIP incoming call.

3.5 Reset Button Feature



Dynamix IAD-162 has a reset button, it is using software reset. User just need to press this button 3 seconds, all of the configuration, including network interface configuration, will be back to default value.

Warning:

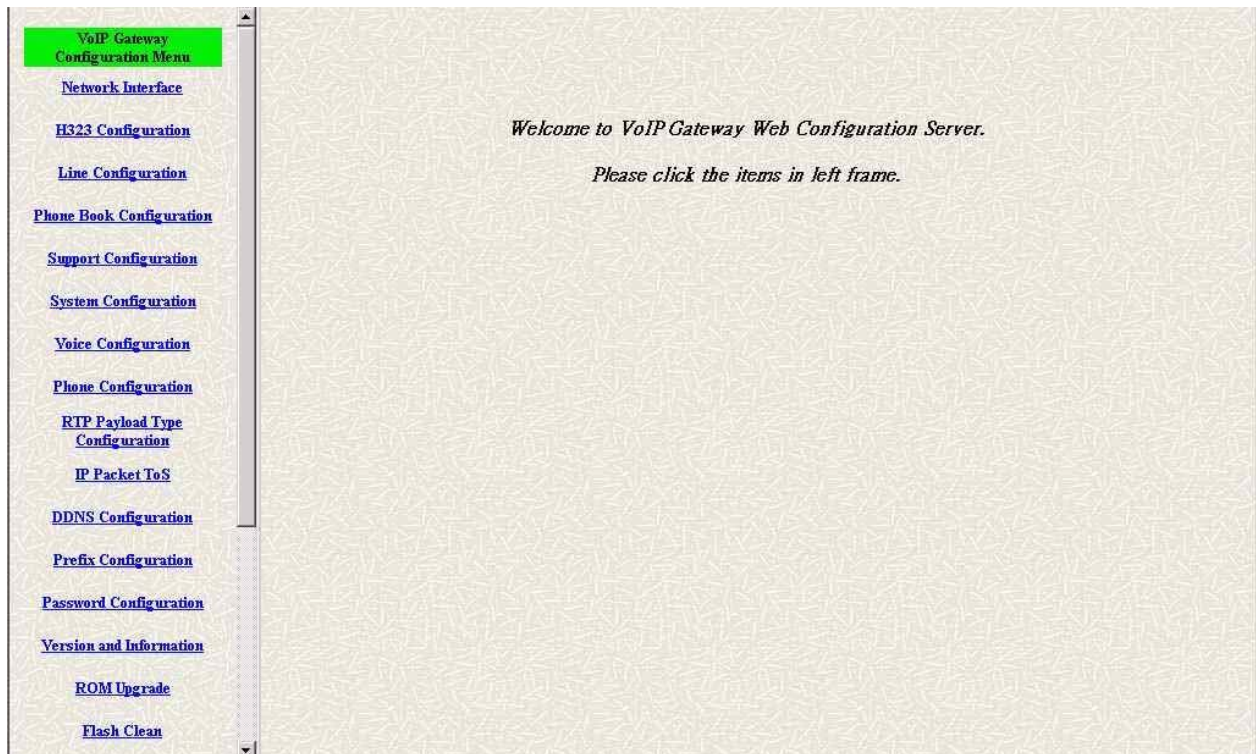
1. The reset feature is using software reset, so user must press this button 3 seconds, and IAD will clean its Flash ROM. After Flash Clean OK, it will reboot automatically.
2. During the period of Flash Clean, user should keep the power supply. If you cut off the power supply during Flash Clean, the IAD may crash and break, caused by the Flash error.

Part IV: Web Management Interface

This part explains how to configure the Dynamix IAD-162 Gateway via WEB

4.1 Login and welcome screen

- Step 1.** Start your web browser.
- Step 2.** Launch your web browser and enter [192.168.123.123] (the default IP address of the LAN port) in the **Location** or **Address** field. Press **Enter**.
- Step 3.** Password request screen will appear as below. Please input “**root**” or “administrator” in the user name field and **no password** in the password field.
- Step 4.** Click **OK**.
- Step 5.** After a successful login, you will see the welcome screen described next. User can click links on the navigation panel at left to go to corresponding configuration screen



4.2 Save and Reboot

Click **OK** at the end of every configuration page to confirm your changes. All configurations will not take effect before reboot system. Please remember to do **[Commit Data]** to save all configuration then **[Reboot System]** to reboot IAD.

4.3 Web Management Configuration

4.3.1 Network Interface

Click [Network Interface] in the navigation panel and open the Network Interface Screen.

Network Interface	
IP Mode:	<input checked="" type="radio"/> Static <input type="radio"/> DHCP <input type="radio"/> PPPoE
IP Address:	192 . 168 . 13 . 6
Subnet Mask:	255 . 255 . 248 . 0
Default Routing Gateway:	192 . 168 . 8 . 254
DHCP server:	
DHCP Server switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
NAT function switch:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
LAN Port IP for NAT:	192 . 168 . 123 . 123
DNS Server Obtained:	<input checked="" type="radio"/> Auto <input type="radio"/> Manual
DNS primary:	168 . 95 . 192 . 1
DNS secondary:	168 . 95 . 1 . 1
HTTP Port:	80
SNTP:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
IP Sharing Server Address:	210 . 59 . 163 . 198
IP Change:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
PPPoE User Name:	123456@hinet.net
PPPoE Password:	*****
PPPoE IP Address:	
PPPoE Destination:	
PPPoE DNS primary:	
Reboot After Remote Host Disconnection:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Send PPPoE Echo Request:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
EMS IP(except 37xx):	1 . 1 . 1 . 1
EMS User Name(except 37xx):	111
EMS Password(except 37xx):	**
EMS Time(except 37xx):	0
OK	

- **IP Mode:** To configure the fixed or dynamic IP address for this unit. Please configure to PPPoE if the ADSL is using the PPPoE type.

- **IP Address:** Set WAN IP Address of IAD.
- **Subnet Mask:** Set the Subnet Mask of IAD
- **Default routing gateway:** Set Default routing gateway of IAD
- **DHCP server:** The DHCP server IP address.
- **DHCP Server switch:** Enable the DHCP function for the LAN interface.
- **NAT function switch:** To enable or disable the NAT function.
- **LAN port IP for NAT:** To configure the LAN port IP for the NAT device.
- **DNS Server Obtained:** If the IP mode is DHCP mode, the DNS is obtained from the DHCP server, user cannot change the DNS IP address. But if users configure this option to manual, they can change it.
- **DNS Primary:** Set Primary Domain Name Server IP address.
User can set Domain Name Server IP address. Once IAD can connect with DNS server, user can specify URL address instead of IP address for GK and phone book IP address.
- **DNS Secondary:** Set Secondary Domain Name Server IP address.
- **HTTP Port:** To configure the HTTP port for access this unit from the remote side.
- **SNTP:** Enable / Disable the Simple Network Time Protocol function
- **SNTP Server Address:** Set SNTP Server Address
When SNTP server is available, enable IAD SNTP function to point to SNTP server IP address so that IAD can get correct current time.
- **GMT:** Set time zone for SNTP Server time
User can set different time zone according to the location of IAD. For example, in Taiwan the time zone should be set as 8, which means GMT+8.
- **IP Sharing:** Enable it if IAD is behind IP Sharing router.
- **IP Sharing Server Address:** Enter the WAN IP address of the IP sharing device if it is the fixed ip.
- **IP Change:** Enable this function if the WAN IP address of the IP sharing device is dynamic address.
- **PPPoE User Name:** To configure the user name for the PPPoE connection.
- **PPPoE Password:** To configure the password for the PPPoE connection.
- **PPPoE IP Address:** In the PPPoE mode, this table will show the ip address that this unit gets from the ISP.
- **PPPoE Destination:** In the PPPoE mode, this table will show the default gateway address that this unit gets from the ISP.
- **PPPoE DNS primary:** In the PPPoE mode, this table will show the DNS ip address that this unit gets from the ISP.
- **After Remote Host Disconnection:** This unit will reboot and re-connect to the ISP

- **Send PPPoE Echo Request:** In the PPPoE mode, if the network connector or the ADSL modem was lost, after the connector and modem connected, it will reboot automatically for the re-connect with the PPPoE server.
- **EMS IP:** Set the EMS server IP address.
- **EMS User name:** Set the EMS login user name.
- **EMS password:** Set the EMS login password.
- **EMS time:** Set the EMS search time. For example, if user configure the EMS time as 2, the IAD will search the EMS server every 2 minutes.

Note:

1. The IP change function could support the GK from us only. Please pay more attentions about this function if your IP sharing device is using the dynamic IP address.
2. The EMS server is a managed center, user can use it to upgrade firmware or restore the configuration. For more information, please refer to the EMS user manual.

4.3.2 H323 Configuration Screen

Click [H323 Configuration] in the navigation panel and open the H323 Configuration Screen.

The H323 Configuration screen is divided into two main sections. The top section contains the following fields:

H323 Configuration	
Mode:	<input checked="" type="radio"/> GK routed <input type="radio"/> Peer-to-Peer
Gatekeeper IP:	192.168.13.77
2nd Gatekeeper IP:	10.1.1.2
Default Gateway IP:	x
Gateway Type:	<input type="radio"/> Terminal <input checked="" type="radio"/> Gateway1 <input type="radio"/> Gateway2
Registered Prefix:	79811
Line1/TEL1 Number:	798111
Line2/LINE1 Number:	
Line3/TEL2 Number:	
Line4/LINE2 Number:	
Line5 Number:	
Line6 Number:	
Registered Alias:	798111234
Display Information:	1FXS
Gatekeeper Discovery:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

The bottom section contains the following fields:

Display Information:	1FXS
Gatekeeper Discovery:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Gatekeeper ID:	GK
Time To Live (TTL):	60
RTP Port:	16384
Gatekeeper finding port:	1718
Gatekeeper RAS Port:	1719
H225 RAS Port:	1024
H225 Call Signal Port:	1720
Destination H225 Call Signal Port:	1720
Allocate Port Range Start:	2000
Allocate Port Range End:	19999
Response Timeout:	15
Connection Timeout:	60
H.235 Security Token:	*
OK	

- **Mode:** Choosing the calling mode for this gateway.
 1. GK routed: Users have to register on the GK if users picked up this option.

2. Peer-to-Peer: It only supports the peer-to-peer and users have to define the phone book for this mode.
- **Gatekeeper IP Address:** Enter the GK ip if users pick up the GK routed mode. **It support on GK mode only.**
 - **2nd Gatekeeper IP:** Support the secondary GK function. **It support on GK mode only.**
 - **Default Gateway IP:** if the number couldn't be found in the phone book configuration, it will be sent to this location.
 - **Gateway Type:** Pick up the type for gateway. **It support on GK mode only.**
 1. **Terminal:** Couldn't support one-stage-dialing function.
 2. **Gateway1:** Could support one-stage-dialing function and the prefix number is necessary for this type.
 3. **Gateway2:** Following the Cisco GK registration function.
 - **Registered Prefix:** The phone number for the GK registering. It support on GK mode only.
 - **Line 1/ TEL 1 Number:** Configure the number of the first port in this unit.
 - **Line 2 / LINE1 Number:** Configure the number of the second port in this unit. **Not support on IAD.**
 - **Line 3 / TEL 2 Number:** Configure the number of the third port in this unit. **Not support on IAD.**
 - **Line 4 / LINE2 Number:** Configure the number of the fourth port in this unit. **Not support on IAD.**
 - **Line 5 Number:** Configure the number of the fifth port in this unit. **Not support on IAD.**
 - **Line 6 Number:** Configure the number of the sixth port in this unit. **Not support on IAD.**
 - **Registered Alias:** The name of this gateway for the GK registering. It support on GK mode only.
 - **Display Information:** This configuration will change the display name for this unit.
 - **Gatekeeper Discovery:** Gateway will send the GRQ message and it will register on the GK if it had received the GCF message.

Note:

When users enable this function, the GK name is necessary. User could enable this function first and define the name of the GK. Gateway will send the GRQ message by broadcast if users define the IP address of GK is 255.255.255.255. If the gateway receive GCF message that's meaning the GK accept the request from gateway, so the gateway could register on that GK successfully who reply the GCF message.

- **Gatekeeper ID:** The name of the GK. It has used with the Gatekeeper Discovery function. **Support the GK mode only.**
- **Time To Live (TTL):** The time for registration confirm. **Support the GK mode only.**
- **RTP Port:** The UDP port for the voice sending. RTP ports support a range of the UDP. The line 1 is using UDP (RTP) 16384 and (RTCP) 16385. The line 2 is using UDP (RTP) 16386 and (RTCP) 16387....etc. This configuration is defining the start port for the RTP packets. **Support the GK and Peer-to-Peer mode both.**
- **Gatekeeper finding port:** The port for the Gatekeeper Discovering function of this gateway. **Support the GK mode only.**
- **Gatekeeper RAS Port:** The GK registering port of this gateway. **Support the GK mode only.**
- **H225 RAS Port:** The RAS port of this unit. **Support the GK mode only.**
- **H225 Call Signal Port:** The Call Signal port of this unit. **Support the GK mode only.**
- **Destination H225 Call Signal Port:** To configure the Call Signal port for the destination unit. **Support the P2P mode only.**
- **Allocate Port Range Start:** To configure the beginning of the ports.
- **Allocate Port Range End:** To configure the end of the ports.

Note:

The port range is for the H245 TCP port. Please info your vendor if you want to configure these ports for using.

- **Response Timeout:** The call will be timed out if the call proceeding message didn't received from the remote side. **Support the GK mode only.**
- **Connection Timeout:** The call will be timed out if the connect message didn't received from the remote side. **Support the GK mode only.**
- **H235 Security Token:** Support the H235 security policy for the GK. **Support the GK mode only.**

4.3.3 Line Configuration

Click [System Configuration] in the navigation panel and open the [System Configuration] Screen.

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration.

Line Number	Type	Hunting Group	Hot Line	No Answer Forward (FXS)	Registration	Status
Line 1/ TEL1:	FXS	1	x	x	Not Registered	Ready
Line2 /Line1:						
Line3 /TEL2:						
Line4 /Line2:						
Line5:						
Line6:						

OK

- **Type**: Just show the interface for this port.
- **Hunt Group**: Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group. Only the same type could be configured in the same group.
- **Hotline**: Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- **No Answer Forward**: When the port didn't answer the call, this call will be forwarded to the number you configured. This is only for the E164 number or the phone numbers you want to transfer.
- **Registration**: To show the gateway registered on the GK or not. **Support the GK mode only.**
- **Status**: To show the port is busy or ready.

4.3.4 Phone Book Configuration Screen

Click [PPPoE Configuration] in the navigation panel and open the [PPPoE Configuration] Screen.

The Phone Book configuration is only support the gateway in Peer-to-Peer mode.

Phone Book						
Index	Name	E164	IP Address	Port	Drop	Insert

New Record						
Index	Name	E164	IP Address	Port	Drop Prefix	Insert Prefix
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input checked="" type="radio"/> Disable <input type="radio"/> Enable	<input type="text"/>
Add Data			Delete Data			

- **Add Data:** User can specify 5 sets of phone book via Web Management Interface. Please input index, Name, IP Address and E.164 number of the destination device.

Note:

User only can add 5 phone book items via Web Management Interface, if user wants add more than 5 items, please uses the Command Line Interface to do it.

- **Delete Date:** User can delete any configured phone book data by index.

Note:

The e164 number defined in phone book will be fully sent to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its line number.

4.3.5 Support Configuration Screen

Click [Support Configuration] in the navigation panel and open the [Support Configuration] Screen.

This gateway supports the FAX over IP, fast start function and other functions.

Support Configuration	
T.38 FAX:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
T.38 FAX ECM:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
T.38 FAX ASN.1:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
FAX Redundancy Depth:	0
Fast Start:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
H.245 Tunneling:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
H.245 Message After Fast Start:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Early H.245:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
H.450 (FXS):	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
OK	

- **T.38 FAX:** Enable this function to support the FAX function.
- **T.38 FAX Request Mode:** The switch can control the request message sending or not. This is the special function, please info your reseller before you change this switch.
- **T.38 FAX ECM:** This function could support the error correction mode during the high-speed function.
- **T.38 FAX ASN.1 Support:** Support the ASN.1 function.
- **Fax Redundancy depth:** This support function can make the data for the FAX sending for twice. But this will take more bandwidth.
- **Fast Start:** Enable this function will support the Fast start function.
- **H.245 Tunneling:** Enable or disable the Tunneling support.
- **H.245 Message After Fast Start:** Enable or disable this support function.
- **Early H.245:** Enable or disable the Early H.245 support function.
- **H.450 (FXS):** Support H.450 function.

4.3.6 System Configuration Screen

Click [System Configuration] in the navigation panel and open the [System Configuration] Screen.

The screenshot shows the 'System Configuration' screen with the following settings:

System Configuration	
Inter Digit Time:	3
Forward Time(FXS):	30
Keypad DTMF Type:	<input type="radio"/> In-Band <input type="radio"/> H.245(Alpha) <input checked="" type="radio"/> H.245(Sig) <input type="radio"/> Q.931 <input type="radio"/> RFC2833
User defined Prefix Switch:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
User defined Prefix Disable:	+
User defined Prefix:	0
Codec Select Method:	<input checked="" type="radio"/> Master <input type="radio"/> Caller
Reverse(FXS):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Local Generate Ring Back Tone:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Round Trip:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Gateway Prefix:	<input checked="" type="radio"/> Keep <input type="radio"/> Drop
End of Dial:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Dial to PSTN side(162):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Detect silence voice(FXO):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Ring Time(FXO):	ms

The screenshot shows the 'System Configuration' screen with the following settings:

User defined Prefix:	0
Codec Select Method:	<input checked="" type="radio"/> Master <input type="radio"/> Caller
Reverse(FXS):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Local Generate Ring Back Tone:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Round Trip:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Gateway Prefix:	<input checked="" type="radio"/> Keep <input type="radio"/> Drop
End of Dial:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Dial to PSTN side(162):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Detect silence voice(FXO):	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Ring Time(FXO):	ms
Ring Before Answer(FXO):	
Delay to add DTMF(FXO):	sec
Auto connect time(FXO):	sec
FXO Type(FXO):	<input checked="" type="radio"/> Normal <input type="radio"/> Force 2nd dial
Hardware Type:	<input checked="" type="radio"/> Auto Detect <input type="radio"/> 1FXS + 1FXO <input type="radio"/> 2FXS + 2FXO
OK	

- **Inter Digit Time:** The call will be sent out if user didn't enter the digits after this timer.
- **Forward time:** It supports the No Answer Forward function. If users configure it

for 10, the call will be forwarded when it rings about 10 seconds. From 5 to 65535.

- **Keypad Type:** User could define the keypad for the DTMF sending.
 1. In-Band : The DTMF signal sending by RTP.
 2. Out-Band : The DTMF signal sending not by RTP. Including the H.245(Alpha), H.245(Signal), Q.931 and RFC 2833.
- **User Defined Prefix Switch:** Select on/off for User defined local zone prefix switch. If user enables prefix function, once user dials out, gateway will automatically add prefix number before number user dialed.
- **User Defined prefix Disable:** Disable the defined prefix after press the selected digit (0,1~9, *).
- **User Define prefix:** This will be added in the first digits of the numbers that users had dialed. Can also define IP address here in P2P mode, once user press "#", Gateway will call out this IP address.
- **Codec Select Method:** This could support that the codec will follow the MSD (MasterSlaveDetermination) or the caller side.
 1. Master: Follow the result from the MasterSlaveDetermination.
 2. Caller: Follow the caller side.
- **Reverse (FXS):** Enable or disable the Reverse signal generation function.
- **Local Generate Ring Back Tone (FXS):** To enable or disable the ring back tone generation from the local side.
- **Round Trip:** The call will be disconnected automatically, if the network is failed, default value is 10 seconds.
- **Gateway Prefix:** To keep or drop the prefix number of this gateway. This only support the Gateway type in the GK routed mode. Please get more detail useful as below:
- **End of Dial:** To enable or disable the end of dial function. This function key will be the digit "#".
- **Dial to PSTN side (162):** This function can let user dial out by PSTN. User just need to press "*#", the call signal will relay from IP side to PSTN side.
- **Detect silence voice (FXO):** This function could detect the silence from the PSTN side. If the FXO detect the silence for 40 seconds continuous, it will drop the calls. **Only support FXO.**
- **Ring Time (FXO):** It for the ring detection from the PSTN. The ring detection will be failed if users configure it too long. **Only support FXO.**
- **Ring Before Answer (FXO):** This will help the users to answer the calls from PSTN into this gateway quickly. The call will be connected by one time ring if users configure this for 1. **Only support FXO.**
- **Delay to add DTMF (FXO):** The timer for sending DTMF signal, if the calls are

from the IP to PSTN side. It could only support the one-stage-dialing function.

Only support FXO.

- **Auto connect time (FXO):** The FXO will send the connect message to the IP side is this timer is up in the one-stage-dialing function. It could only support the one-stage-dialing function. Only support FXO.
- **FXO type:** Users could configure the calls need the 2nd stage dialing or not. Only support FXO.
- **Hardware Type:** This will show the hardware detection type. Only support FXO.

Note:

The default value is to auto detect hardware type. Usually it is not necessary to change this setting. Please make sure about your Hardware Type, Gateway may be not functional if set wrong hardware type.

4.3.7 Voice Configuration Screen

Click [Voice configuration] in the navigation panel and open the [Support Configuration] Screen.

Voice Configuration						
Codec Priority	1st G. 723. 1	2nd G. 729	3rd G. 729a	4th G. 729b	5th G. 729ab	6th G. 711mu-L
Frame Size	G.723 60	G.729 60	G.729a 60	G.711u 40	G.711a 40	G.729b 60
Line1/TEL1 Volume:	Voice 28		Input 28		DTMF 23	
Line2/LINE1 Volume:	Voice		Input		DTMF	
Line3/TEL2 Volume:	Voice		Input		DTMF	
Line4/LINE2 Volume:	Voice		Input		DTMF	
Line5 Volume:	Voice		Input		DTMF	
Line6 Volume:	Voice		Input		DTMF	
G723 Silence Suppression:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable					
Echo Canceller:	Line1 <input checked="" type="radio"/> Enable <input type="radio"/> Disable	Line2 <input type="radio"/> Enable <input checked="" type="radio"/> Disable	Line3 <input type="radio"/> Enable <input checked="" type="radio"/> Disable	Line4 <input type="radio"/> Enable <input checked="" type="radio"/> Disable	Line5 <input type="radio"/> Enable <input checked="" type="radio"/> Disable	Line6 <input type="radio"/> Enable <input checked="" type="radio"/> Disable
Jitter Buffer:	Min. Delay 90			Max. Delay 150		

- **Codec Priority:** Set priority preference of installed codes, G.723, G.711A, G.711U, G.729, G.729A, G.729B, and G.729AB.
- **Frame Size:** Set Specify sending packet size, G.723: 30/60/90, G.711A, G.711U, G.729, G.729A, G.729B, G.729AB: 20/40/60ms. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended.
- **Line1/ TEL1 Volume:** Set voice volume stands for volume, which can be heard from VoIP Gateway side (0~63, default: 28). Set input gain stands for volume, which the opposite party hears (0~38, default: 28). Set DTMF volume stands for DTMF volume/level (0~31, default: 23).
- **Line2/ Line1 Volume:** IAD does not support.
- **Line3/TEL2 Volume:** IAD does not support.
- **Line4/Line2 Volume:** IAD does not support.
- **Line5 Volume:** IAD does not support.
- **Line6 Volume:** IAD does not support.
- **G723 Silence Suppression:** Select enable/disable for G723 silence suppression and comfort noise generation setting.
- **Echo Canceller:** Setting enable/disable of echo canceller.
- **Jitter Buffer:** Setting of jitter buffer min/max delay.

4.3.8 Phone Configuration Screen

Click [Phone Configuration] in the navigation panel and open the [Phone Configuration] Screen. For tone simulation, IAD Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones.





Phone Configuration								
Ring Cadence:	Frequency	20	On	2000	Off	4000	Level	94
Ring Back Tone:	Low(freq)	440	High(freq)	480	Low(lev)	1	High(lev)	1
					On1	100	Off1	200
					On2	100	Off2	200
Busy Tone:	Low(freq)	480	High(freq)	620	Low(lev)	1	High(lev)	1
					On1	50	Off1	50
					On2	50	Off2	50
Dial Tone:	Low(freq)	350	High(freq)	440	Low(lev)	1	High(lev)	1
					On1	500	Off1	1023
					On2	500	Off2	1023
2nd Dial Tone:	Low(freq)	350	High(freq)	440	Low(lev)	1	High(lev)	1
					On1	25	Off1	25
					On2	25	Off2	25
Flash Timer:	Low	100	High	300				
OK								

- **Ring Cadence:** Setting the played tone type, when VoIP Gateway is receiving a call.
- **Ring Back Tone:** Setting the played tone type, when VoIP Gateway receives a Q.931 Alerting message. In condition that VoIP Gateway is the originate side.
- **Busy Tone:** Setting the played tone type, when destination is busy.
- **Dial tone:** Setting the played tone type, when hook off a phone set of workable VoIP Gateway.
- **2nd Dial Tone:** To configure the value of the local 2nd dial tone (FXO only).
- **Flash Timer:** Setting the detective flash range in ms, for example, 300-500 ms.

Note:

For tone simulation, VoIP Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

 ringing frequency: 15 ~ 100 (Unit: Hz)

-  ringing ring ON/OFF: 0 ~ 8000 (Unit: ms)
-  ringing level: 0 ~ 94 (Unit: V)
-  tone frequency: 0 ~ 65535 (Unit: Hz)
-  tone freqLevel: 0 ~ 65535 (Unit: mVrms)

tone Tone ON/OFF: 0 ~ 8000 (Unit: ms)

4.3.9 RTP Pay Load Type Configuration

Click [RTP Payload Type Configuration] in the navigation panel and open the [RTP Payload Type Configuration] Screen.

RTP Payload Type Configuration	
RFC2833 Payload Type:	<input type="text" value="96"/>
DTMF Payload Type:	<input type="text" value="100"/>
FAX Payload Type:	<input type="text" value="101"/>
FAX ByPass Payload Type:	<input type="text" value="102"/>
MODEM ByPass Payload Type:	<input type="text" value="103"/>
Redundancy Payload Type:	<input type="text" value="104"/>
MODEM Relay Payload Type:	<input type="text" value="105"/>
OK	

- **RFC2833 Payload Type:** To define the payload type for RFC2833 type.
- **DTMF Payload Type:** To define the payload for the DTMF type.
- **FAX Payload Type:** To define the payload for the FAX type.
- **FAXByPass Payload type:** To define the payload for the FAX by Pass type.
- **MODEMByPass Payload Type:** To define the payload for the Modem by Pass type. (This is no use for the hardware as so far.)
- **Redundancy Payload Type:** To define the payload for the Redundancy type.
- **MODEMRelay Payload Type:** To define the payload for the FAX by Pass type. (This is no use for the hardware as so far.)

4.3.10 IP Packets ToS Configuration Screen

The Type of Service should be worked with the network router. The router will check all the packets if it support the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure these values to make the packets with the correct values for the TOS service from the gateway.

The screenshot shows the 'IP Packet TOS (Tapy of Service) Configuration' screen. On the left is a 'VoIP Gateway Configuration Menu' with various options. The main area has a blue header and two input fields for DSCP codes, both set to 0, with an 'OK' button below them.

IP Packet TOS (Tapy of Service) Configuration	
Signalling Packet DSCP Code:	<input type="text" value="0"/>
Media Packet DSCP Code:	<input type="text" value="0"/>
OK	

Set Signal or RTP Packet DSCP value:

- **Default:** Select TOS value as 0
- **User Assign Special DSCP Code:** User can set other unspecified value here.

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Please refer to RFC standard documents for more information about what is DSCP.

4.3.11 DDNS Screen

IAD supports DDNS function. Before using this function, please have a DDNS account and some info from your DDNS server.

Click [Password configuration] in the navigation panel to open the [Password Configuration] screen.

DDNS Device Configuration	
Status:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Server:	<input type="text" value="members.dyndns.org"/>
Host Name:	<input type="text"/>
ID:	<input type="text"/>
Password:	<input type="text"/>
Check IP:	<input type="radio"/> On <input checked="" type="radio"/> Off
IP Check Server 1:	<input type="text" value="checkip.dyndns.org"/>
IP Check Server 2:	<input type="text" value="checkip.dyndns.org"/>
Check every:	<input type="text" value="0"/> <input type="radio"/> minutes <input type="radio"/> hours <input checked="" type="radio"/> Off
OK	

- **Status:** To enable or disable this function.
- **Server:** Enter the server address of your DDNS server.
- **Host Name:** Your DDNS address.
- **ID:** Your account.
- **Password:** Your password.

Note:

Please get this info from your DDNS service provider.

- **Check IP:** Enable or Disable the IP Check function.
- **IP Check Server:** Will check the endpoints IP address.
- **Check Every:** The endpoint will check the server after a period of time.

4.3.12 Prefix Configuration Screen

The Prefix function is using the drop and inserts function.

Click [Prefix Configuration] in the navigation panel and open the [Prefix Configuration] Screen.

VoIP Gateway Configuration Menu

- Network Interface
- H323 Configuration
- Line Configuration
- Phone Book Configuration
- Support Configuration
- System Configuration
- Voice Configuration
- Phone Configuration
- RTP Payload Type Configuration
- IP Packet To S
- DDNS Configuration
- Prefix Configuration**
- Password Configuration
- Version and Information
- ROM Upgrade
- Flash Clean
- Commit Data

Prefix Drop/Insert Configuration

Index	Prefix	Drop	Insert

New Prefix

Index	Prefix	Drop	Insert
<input type="text"/>	<input type="text"/>	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	<input type="text"/>
<input type="button" value="Add Data"/> <input type="button" value="Delete Data"/>			

- **Index:** Setting the index number for prefix record (max 30 record).
- **Prefix:** Setting the prefix number of the whole numbers that could be into this VoIP gateway (1~20 digits).
- **Drop:** Select enable or disable drop prefix function. The function is enabled means to drop prefix number when dialing out. The function is disabled means to keep prefix number.
- **Insert:** Setting the digits that you want to insert in this number (1~30 digits).

4.3.13 Password Configuration Screen

There are two login accounts in this unit. One is the “root” another is “administrator”. The default passwords for these two accounts are null. Users can define the passwords for these two accounts.

Click [Password Configuration] in the navigation panel and open the [Password Configuration] Screen.

Password Configuration	
<div>root ▼</div>	<div>Current Password: <input type="text"/></div> <div>New Password: <input type="text"/></div> <div>Confirm New Password: <input type="text"/></div>
<div>CHANGE ABORT</div>	

- **Root:** The password for the root account.
- **Administrator:** The password for the administrator account. This account couldn't upgrade the 2M and boot rom file.
- **Current Password:** Enter the original password for the account.
- **New Password:** Enter the new password for the account.
- **Confirm New Password:** Enter the new password again.
- **Change:** This button will make the configurations saved and next time login will need the new password.
- **Abort:** Abort the configuration of the password changing.

4.3.14 Version and Information Screen

Users can get more detail about the software version for all the parts in this web page. Click [Version and Information] in the navigation panel and open the [Version and Information] Screen.

Version and Information	
Boot Rom:	sdboot.200
Application Rom:	IAD_1227.bin
DSP Application:	48302ce3.140
DSP Kernel:	48302ck.140
DSP Test Code:	483cbtbin
Greetings:	greeting.100
Ask Pin:	

- **Boot Rom:** The version of the Boot Rom layer.
- **Application Rom:** The version of the Application Rom layer.
- **DSP Application:** The version of the DSP Application Rom layer.
- **DSP Kernel:** The version of the DSP Kernel layer.
- **DSP Test Code:** The version of the DSP Test Code layer.
- **Greeting:** The version of the Greeting file. **For FXO only.**
- **ASK Pin:** The version of the ASK Pin file. **For FXO only.**

4.3.15 ROM Configuration Screen

Click [ROM Upgrade] in the navigation panel and open the [ROM Configuration] Screen.

ROM Configuration	
TFTP/FTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	TFTP <input type="button" value="v"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	Application Image <input type="button" value="v"/>
<input type="button" value="OK"/>	

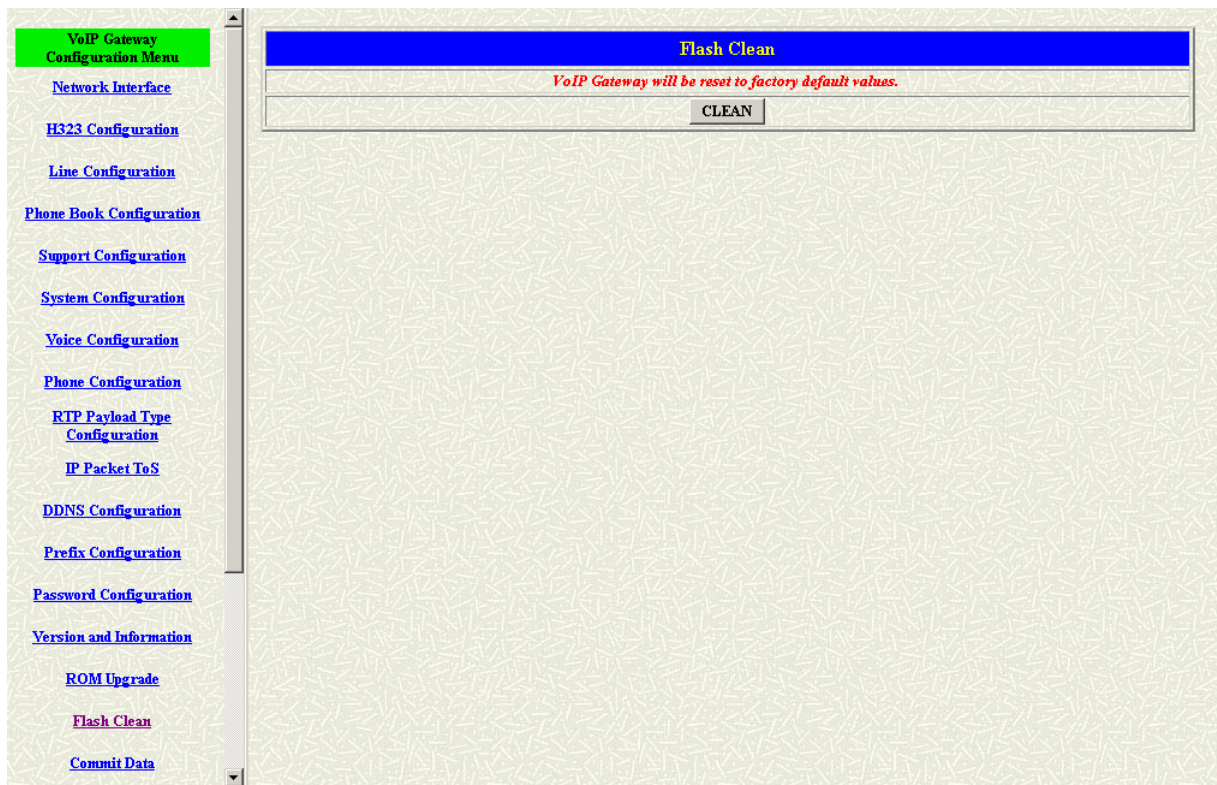
- **FTP/TFTP Server IP Address:** Set TFTP server IP address
- **Target File name:** Set file name prepared to upgrade
- **Method:** Select download method as TFTP or FTP
- **FTP Login:** Set FTP login name and password
- **Target File Type:** Select which sector of IAD to upgrade

Note:

After upgrading 2mb file or Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

4.3.16 Flash Clean Screen

Click [Flash Clean] in the navigation panel and open the [Flash Clean] Screen.

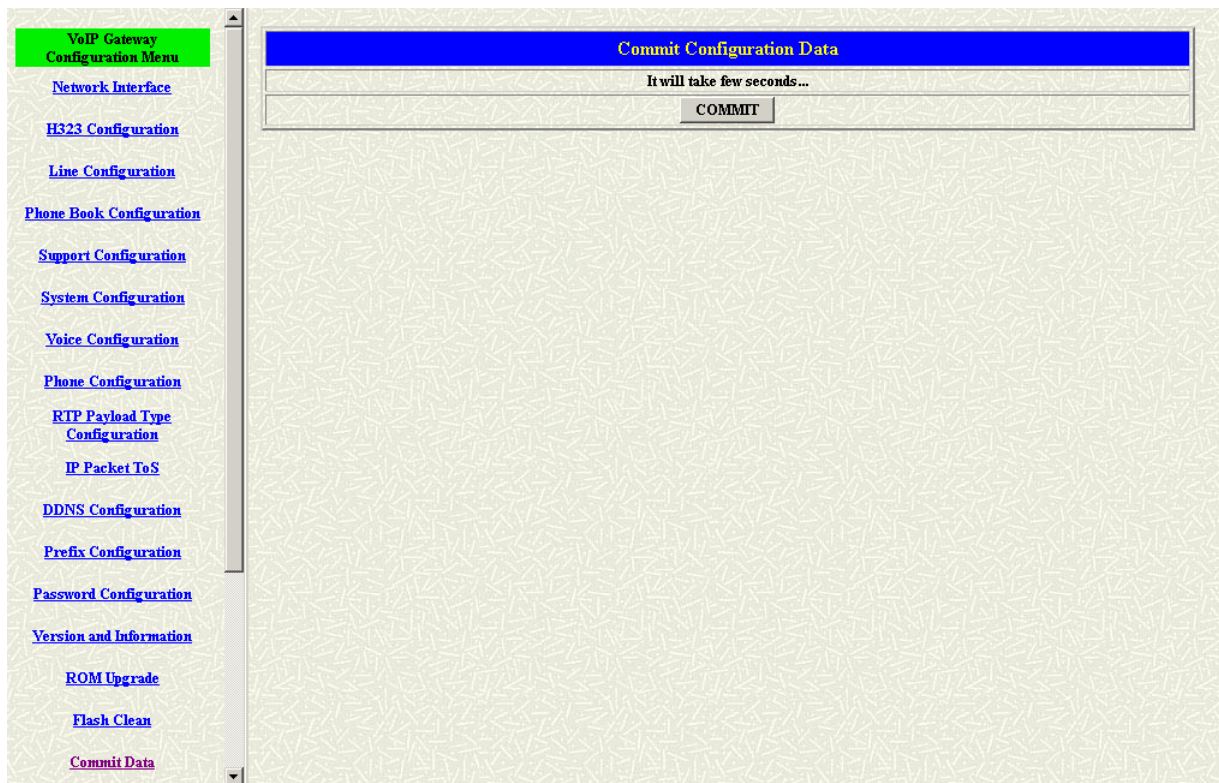


- Press CLEAN will clean all configurations of IAD and reset to factory default value.

Note:
User must re-configure all commands all over again (except Network Configure) once execute this function.

4.3.17 Commit Configuration Data Screen

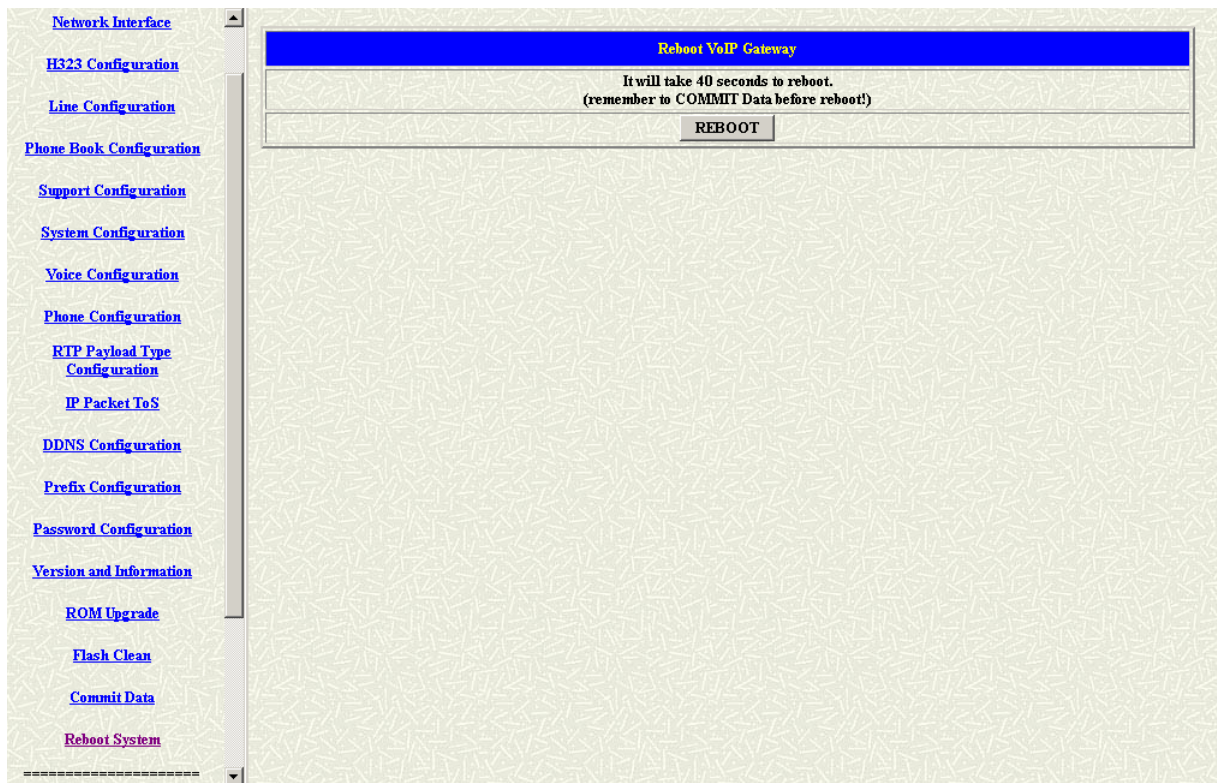
Click [Commit Data] in the navigation panel and open the [Commit Configuration Data] Screen.



- Commit Date to save all configurations. Please remember to commit data before reboot your IAD.

4.3.18 Reboot IAD System screen

Click [Reboot System] in the navigation panel and open the [Reboot Dynamix IAD-162 Gateway] Screen.



- Press reboot will reset IAD.

Note: To execute reboot, please remember to do **Commit Data** before **Reboot System**.

Part V: Telnet Command Interface

This part gives information on how to configure Dynamix IAD-162 gateway via Telnet command line interface.

5.1 Login

For you first login, enter the login: [root] and default no password.

```
login: root
password:
Welcome to Terminal Configuration Mode
Please enter your configuration item

usr/config$
```

Note:

Login account [root] or [administrator] is the default login account and there is no password needed.

5.2 Save and Reboot

After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Confirm the changed configurations, input [commit] and press [enter] key to save it.

Step 2. Input [reboot] then press [enter] key to restart Gateway.

Step 3. After around 40 seconds, Gateway will take effect in new configurations.

Do not turn off your Gateway or remove the Gateway while saving your configuration.

```
usr/config$ passwd -set root voip

Setting
Login: root
Password: voip
OK

usr/config$
```

5.3 System Commands Overview

5.3.1 [help]

Press help/man/? will display all command list of IAD. The following table lists all of the commands that you can use with the Gateway. Refer to the following chapters for descriptions of commonly used commands.

This user's guide describes commands that are helpful for configuring the Gateway. Using commands not documented in the user's guide can damage the unit and possibly render it unusable.

Commands with Dynamix IAD-162 Gateway

Command	Description
help	help/man/? [command]
quit	quit/exit/close
debug	Show debug message
reboot	Reboot local machine
flash	Clean configuration from flash rom
commit	Commit flash rom data
ifaddr	Internet address manipulation
time	Show current time
ping	Test that a remote host is reachable
sysconf	System information manipulation
h323	H.323 information manipulation
line	Gateway line information and configuration
prefix	Prefix drop/insert information manipulation
pbook	Phonebook information and configuration
voice	Voice information manipulation
support	Special Voice function support manipulation
sysinfo	System information
phone	Setup of call progress tones and ringing (SLIC control)
tos	IP Packet ToS (Type of Service) values
ddns	Dynamic DNS update manipulation
pt	RTP payload type configuration and information
rom	ROM file update
Auth	Auth to customized the WEB configuration item for administrator.
passwd	Password setting information and configuration

5.3.2 [quit]

Type [quit] will quit the Gateway configuration mode. And turn back to login prompt.

```
usr/config$ quit

Disconnecting..
login: root
Welcome to Terminal Configuration Mode
Please enter your configuration item

usr/config$
```

Note:

It is recommended that type the [quit] command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

5.3.3 [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command [debug -open] as well.

```
usr/config$ debug

Debug message information and configuration
Usage:
debug [-add type1 [[type2]..]] | -open | -close | -status
    -status    Display the enabled debug flags.
    -add       Add debug flag.
    -delete    Remove specified debug flag.
    -open      Start to show debug messages.
    -close     Stop showing debug messages.
Example:
    debug -add h323 h323vp
    debug -open

usr/config$
```

Parameter Usages:

- status: Display the enabled debug flags.

- **add**: Add debug flag.

h323: h323 related information

vp: voice related information

- **delete**: Remove specified debug flag.
- **open**: Start to show debug messages.
- **close**: Stop showing debug messages.

In this example, user open debug flags including h323, vp, h323vp.

```
usr/config$ debug -add h323 vp h323vp
usr/config$ debug -open
```

For example:

```
usr/config$ debug -status

Current debug type enabled :
Debug Mode is open
DEBUG-> H323VP  H323
usr/config$
```

5.3.4 [reboot]

After [commit], type [reboot] to reload Gateway in new configuration. The procedure is as below:

```
usr/config$ reboot

....Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done
  Hardware Type : 1FXS
  Set RTL8305SB
no answer from 192.168.4.114

HTTPD initialized...
cmInitialize succeed!
Ras port:1024
CallSignal port:1720

AC4804[0] is ok
successful 1 2
Initialize OSS libraries...OK!
VP v1.44 stack open sucessfully.
```



```
login:
```

5.3.5 [flash]

Clean the configuration stored in flash.

```
usr/config$ flash
```

Flash memory information and configuration

Usage:

```
flash [-clean]
```

```
flash -clean  Clean the configuration stored.
```

Note:

This command will clean the configuration stored in the flash and reboot it.

```
usr/config$
```

Parameter Usages:

- **clean**: clean all the user defined value, and reboot Gateway in factory default mode.

Note:

It is recommended that use [flash –clean] after application firmware id upgraded.

Warning:

User whose login name is root only executes it. All configurations in command [ifaddr] will be kept.

For example:

```
usr/config$ flash -clean
```

```
Flash clean ok!! Rebooting ...  
Attached TCP/IP interface to cpm unit 0  
Attaching interface lo0...done  
Hardware Type : 1FXS  
Set RTL8305SB  
no answer from 192.168.4.114
```

```
HTTPD initialized...  
cmInitialize succeed!  
Ras port:1024  
CallSignal port:1720
```

```
AC4804[0] is ok  
successful 1 2  
Initialize OSS libraries...OK!  
VP v1.44 stack open sucessfully.
```

```
login:
```

5.3.6 [commit]

Save changes after configuring Gateway.

```
usr/config$ commit
```

```
This may take a few seconds, please wait..  
Commit to flash memory ok!
```

```
usr/config$
```

Note:

Users shall use [commit] to save modified value, or they will not be activated after system reboot.

5.3.7 [ifaddr]

Configure and display Gateway network information.

```
usr/config$ ifaddr
```

```
LAN information and configuration
```

Usage:

```
ifaddr [-print][-mode used][-ip IP Address][-mask Subnet Mask]
        [-gate Default Gateway][-dns 1 IP Address][-http portnumber]
        [-dhcpsv used][-nat used][-lanip IP Address][-sntp mode [server]]
        [-timezone GMT][-ipsharing used IP address][-ipchange used]
        [-id PPPoE username][-pwd PPPoE password][-reboot used][-emsip
EMS IP]
        [-emsid EMS username][-emspwd EMS password][-emstime EMS digit]
```

- print Display network information and configuration.
- mode Specify WAN IP mode (0=Static, 1=DHCP, 2=PPPoE).
- ip Specify WAN port static IP address.
- mask Set WAN port static IP subnet mask.
- gate Specify WAN port static IP default gateway IP address
- autodns Specify the way to obtain DNS Server (0=Manual, 1=Auto).
- dns Set DNS server IP address. Provide set DNS primary and secondary IP address (1=primary IP address, 2=secondary).
- http Specify http port number.
- dhcpsv DHCP server switch (0=disable, 1=enable).
- nat NAT function switch (0=disable, 1=enable).
- lanip Specify LAN port IP address (For NAT function).
- sntp Set SNTP server mode and specify IP address.
SNTP mode (0=No update, 1=Specify server IP, 2=broadcast mode).
- timezone Set local timezone.
- ipsharing Specify usage of an IP sharing device and specify IP address (0=Disable, 1=Enable).
- ipchange Replace IP address if the shared IP is changed (0=Disable, 1=Enable).
- id PPPoE connection user name.
- pwd PPPoE connection password.
- reboot Reboot after remote host disconnection in PPPoE mode (0=No Reboot, 1=Yes Reboot).
- echo PPPoE Echo Request (0=disable, 1=enable).
- emsip EMS server IP(x=IP null).
- emsid EMS user name.
- emspwd EMS user password.
- emstime EMS refresh time(0~1024 min).

Note:

Range of IP address setting (0.0.0.0 ~ 255.255.255.255).

LAN IP can not support subnet x.x.0.x

Example:

```
ifaddr -mode 0
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -dns 1 168.92.1.1 -dns 2 168.95.1.2
ifaddr -http 80
ifaddr -lanip 192.168.123.123
ifaddr -sntp 1 210.59.163.254
ifaddr -ipsharing 1 210.59.163.254 -ipchange 1
ifaddr -id 123456@hinet.net -pwd 123456 -reboot 1
```

Parameter Usages:

- **print**: Print current IP setting and status
- **mode**: Specify WAN IP mode (0=Static, 1=DHCP, 2=PPPoE), DHCP mode: Dynamic Host Configuration, PPPoE mode: PPPoE Dial-up function.
- **ip**: Assigned IP address for Gateway
- **mask**: Assigned internet subnet mask
- **gate**: Assigned IP default gateway
- **autodns**: Specify the way to obtain DNS Server (0=Manual, 1=Auto), this function is only usable in DHCP or PPPoE mode.
- **dns**: Setup DNS Server IP Address.
- **http**: Assigned Gateway web browser connection port number, default port number 80.
- **dhcpsv**: Using this command to let IAD as a DHCP server for LAN port.
- **nat**: Enable or Disable NAT function
- **lanip**: Specify LAN port IP address (For NAT function), use this command setup LAN IP address assigned to PC or other machine.

```
usr/config$ ifaddr -lanip 192.168.XXX.YYY
(The range of LAN IP is XXX: 1-254, YYY: 1-254)
```

- **sntp**: Simple Network Time Protocol (0=No update, 1=Specify server IP, 2=broadcast mode). When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:

```
usr/config$ ifaddr -sntp 1 10.1.1.1
```

Note:

The 10.1.1.1 stands for SNTP server's IP address.

- **timezone**: set local time zone according to GMT

- **ipsharing**: To specify a global fixed IP address, user can add this IP address in the command.

```
usr/config$ ifaddr -ipsharing 1 210.11.22.33
```

Note:

If the IP address is not a fixed one, the dedicated IP address is not necessary in the command. However, dynamic IP Address is not working in Peer-to-Peer mode.

- **ipchange**: If the unit is behind the IP sharing device and the IP address for the WAN port of that IP sharing is using the dynamic IP address. This function has to be enabled.

```
- usr/config$ ifaddr -ipchange 1
```

- **id**: This id is for the user name of the PPPoE usage.
- **pwd**: This password is for the user name of the PPPoE usage.
- **reboot**: If the connection disconnected by the ISP, the unit will reboot and get the ip again.
- **echo**: In the PPPoE mode, if the network connector or the ADSL modem was lost, after the connector and modem connected, it will reboot automatically for the re-connect with the PPPoE server.
- **emsip**: Set the EMS server IP address.
- **emsid**: Set the EMS login user name.
- **emspwd**: Set the EMS login password.
- **emstime**: Set the EMS search time. For example, if user configures the EMS time as 2, the IAD will search the EMS server every 2 minutes.

Note:

The EMS server is a managed center, user can use it to upgrade firmware or restore the configuration. For more information, please refer to the EMS user manual.

5.3.8 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type [time] command to show current network time.

```
usr/config$ time
Current time is WED SEP 17 12:36:49 2003

usr/config$
```

5.3.9 [ping]

Use [ping] to test whether a specific IP is reachable or not.

For example: if 192.168.1.2 is not existing while 210.63.15.32 exists. Users will have the following results:

```
usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping 192.168.1.254
PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
210.63.15.32 is alive
usr/config$
```

5.3.10 [sysconf]

This command displays system information and configurations.

```
usr/config$ sysconf
```

System information and configuration

Usage:

```
sysconf [-print][-idtime digit][-forwardtime digit][-keypad used]
        [-prefixsw used][-prefixdisab used][-usrdefprefix digits]
        [-codec used][-reverse used][-localrpt used][-roundtrip used]
        [-gwpprefix used][-eod used]
```

sysconf

```
-print          Display system overall information and configuration.
-idtime         Inter-Digits time (1~10 sec).
-forwardtime    Forward time for FXS line if no answer (5~65535 sec).
-keypad         Select DTMF type: 0=In-band, 1=H.245 Alphanumeric,
                2=H.245 SignalType, 3=Q.931 UserInfo, 4=RFC2833.
-prefixsw       User defined local zone prefix switch (0=OFF, 1=ON).
-prefixdisab    Local zone prefix disable character (one character
                from 0~9, *, or NONE('-' key)).
-usrdefprefix   User defined local zone prefix (0 ~ 20 digits).
```


-codec	Codec select method (0=Caller, 1=Master).
-reverse	Reverse.(0=Disable, 1=Enable).
-callerid	caller ID.(0=Disable, 1=Enable).
-localrbt	Local ring back tone (0=Disable, 1=Enable).
-roundtrip	Disconnect no connect in line busy.(0=Disable, 1=Enable).
-gwprefix	Drop gateway prefix when call from IP (0=Keep, 1=Drop).
-eod	End of dial (0=Disable, 1=Enable).
-dPSTN	FXS dial *# and change to PSTN side(0=Disable,1=Enable).
Example: sysconf -idtime 5	

Parameter Usages:

- **print**: Print current sysconf settings.
- **idtime**: Set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed (1-10 seconds).
- **forwardtime**: Set forward time (5-65535 seconds) for FXS Line. If call hasn't answered the call in this time, call will be forward to assigned number in [line] command. (Please refer to [line] command for forward setting.)
- **keypad**: Select DTMF replay type (0=In-band, 1=H.245 Alphanumeric, 2=H.245 SignalType, 3=Q.931 UserInfo, 4=RFC2833), Users can adjust the value according to various applications. In-band: Gateway will transfer DTMF signal via RTP payload. H.245 Alphanumeric via H.245 UII Alphanumeric. H.245 SignalType FXS (VoIP) via H.245 UII Singal Type. Q.931 UserInfo via Q.931 UserInfo. RFC2833 via RFC2833.
- **prefixsw**: Switch on/off prefix function. If user enables prefix function, once user dials out, gateway will automatically add prefix number before number user dialed.
- **prefixdisab**: Set disable key (0,1~9, *) to disable the prefix function in this current call. For example, if user has set prefix as 100, and wants to dial out 100123, user can only press 123, to dial out 100123. However, if user wants to dial 123 without prefix, user can press prefix disable key, for example [*], user can press [*], then dial 123, gateway will dial out 123 without adding prefix.
- **usrdefprefix**: Define prefix number.

Note:

The above three commands [prefixsw], [prefixdisab] and [usrdefprefix] have to work together. If user would like to dial 9 to replace 123456789, he can firstly switch on the command [sysconf -prefixsw 1], secondly to define the prefix digits as 12345678 [sysconf -usrdefprefix 12345678]. While user would like to dial other

numbers, he can dial a defined digit – [*] to disable the function

[sysconf –prefixdisab *], then dial to the number he wants to dial.

- **codec**: Set who is the one to determine voice codec during negotiation. 0 is caller will determine the codec, and 1 is master will determine the codec. (During negotiation 2 endpoints will compare gateway type level to determine who is master.)
- **reverse**: If the FXS is the calling party, it will generate the reverse signal to the analog side when it got the connect message from the IP side. It could be used in the pay phone application
- **callerid**: Enable or Disable the caller-id function.
- **localrbt**: Enable or Disable locally generate ring back tone. Disable means gateway will receive ring back tone from remote callee, Enable means gateway will generate ring back tone locally.
- **roundtrip**: The call will be disconnected automatically, if the network is failed, default value is 10 seconds.
- **gwprefix**: Drop or keep gateway prefix . Keep Means when gateway has incoming call from IP side, it will not drop prefix before searching for call number. Drop Means when gateway has incoming call from IP side, it will drop prefix before searching for called number.
- **eod**: It will transfer the DTMF in [#] if users disable the end of dial function. Users have to press the keypad in [#] if the end of dial function is enable.

Note:

User can also define IP address here in P2P mode, once user press “#”, Gateway will call out this IP address.

- **dPSTN**: This function can let user dial out by PSTN. User just need to press “*#”, the call signal will relay from IP side to PSTN side.

For example:

```
usr/config$ sysconf -print
```

System information

```
Inter-Digits time      : 3
Forward time           : 30
Keypad DTMF type       : H.245 SignalType
User defined prefix switch : OFF
User defined prefix disable : *
User defined prefix     : 0
Codec select method     : Master
Reverse                 : Enable
```

```

Caller ID                : Disable
Local generate ring back tone : Enable
Round Trip               : Disable
Gateway prefix           : Keep
End of dial              : Enable
Dial to PSTN side        : Enable

```

```
usr/config$
```

5.3.11 [h323]

This command is to configure H.323 related parameters.

```
usr/config$ h323
```

H.323 stack information and configuration

Usage:

```
h323 [-print][-mode used][-gk IP address][-algk IP address][-gwtype used]
      [-dfgw IP address][-prefix number][-line number][-passwd number]
      [-alias h323id][-display information][-gkid ID][-gkdis used]
```

```

-print      Display H.323 stack information and configuration.
-mode       Configure as Gatekeeper mode or Peer-to-Peer mode
            (0=Gatekeeper, 1=Peer-to-Peer).
-gk         Gatekeeper IP address.
-algk       Second Gatekeeper IP address.
-gwtype     Gateway Type (0=Terminal, 1=Gateway1, 2=Gateway2).
-dfgw       Default Gateway IP address.
-prefix     Prefix number.
-line1      Line 1 E.164 number.
-passwd     H.235 security password.
-alias      IP side registered H323 ID.
-display    String representing display information for reporting
            to the called party.
-gkid       Gatekeeper ID.
-gkdis      Gatekeeper discovery (0=Off, 1=On).
-ttl        RAS TTL time (0~3600 second).
-rtp        RTP port number (1~65535).
-gkfind     Gatekeeper finding port (1~65535).
-gkras      Gatekeeper RAS port (1~65535).

```

- h225 H225 ras port (1~65535).
- q931 H225 call signal port (1~65535).
- dstq931 Destination H225 call signal port (1~65535).
- range Dynamically allocated port range (1~65535).
- respto Max waiting time for 1st response to a new call (1~200).
- connto Max waiting time for call establishment after receiving 1st response of a new call (1~20000).

Note:

Range of IP address setting (0.0.0.0 ~ 255.255.255.255).

Example:

```
h323 -gk 210.59.163.171 -line1 70 -line2 71
h323 -alias Your_Alias_Name -gkid GK -gkdis 1 -passwd 1234
h323 -range start 1024 end 19999
```

Parameter Usages:

- **print**: Print current h323 related settings
- **mode**: Alternatives for gatekeeper or peer-to-peer mode (0=gatekeeper mode; 1=peer-to-peer mode). If users select gatekeeper mode, an extra gatekeeper is needed when Gateway is in operation.

```
- usr/config$ h323 -mode 1 (peer to peer mode)
```

- **gk**: Assign gatekeeper's IP address when Gateway is in gatekeeper mode.
- **algk**: Assign second gatekeeper's IP address as redundancy. If Gateway fails to register to main GK for 10 times, it will try to register to alternative GK.
- **gwtype**: Gateway type has two kinds, gateway and terminal. Gateway type: will register as H.323 defined Gateway; user has to define [prefix] in next command. Terminal type: will register as H.323 defined Terminal, [prefix] command is not necessary.
- **dfgw**: Default gateway is applied under Peer-to-Peer mode. User defines a constant default gateway IP address, then any number dialed will pass forward to this IP Address.
- **prefix**: Assign VoIP Gateway prefix number, as well as the registered number on the Gatekeeper.
- **line1**: Assign line 1 number.
- **passwd**: Set H.235 security password. If user's GK need H.235 security token password to authenticate, user have to input open password in this command.
- **alias**: H.323 ID. If in gatekeeper mode, this h.323 ID must be different from others who are registering to the same gatekeeper.

- **display**: An addition name for user to recognize in called site.
- **gkid**: Set GateKeeper name for GateKeeper discovery. When Gateway send out GateKeeper discovery message will search GateKeeper with this GateKeeper name.
- **gkdis**: Set auto discovery function on or off. If this function is enabled and IP address of GateKeeper is set as 255.255.255.255, Gateway will multicast to search a GateKeeper on network with configured GateKeeper name; if IP address of GateKeeper is set, before Gateway register to the assigned GateKeeper, it will send out GRQ (GateKeeper Request) message with configured GateKeeper name to GateKeeper first.
- **ttl**: To set timer for TTL (Time To Live). Gateway would send RRQ, with keep Alive, to gatekeeper periodically according to TTL timer, default:60 (0~3600 second).
- **rtp**: To allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy (1~65532).
- **gkfind**: Gatekeeper finding port. Which Gateway uses it to discover a gatekeeper. Default value is 1718 (1~65535).
- **gkras**: To set default gatekeeper RAS port number. Default value, is 1719 well-known port for RAS communication.
- **h225**: To set the ras port.(1~65535)
- **q931**: To set the call signal port.(1~65535)
- **dstq931**: To specify destination H.225 signal port number.
- **range**: To allocate port range (1-65535) Gateway may use it.
- **respto**: Maximum response time out
- **connto**: Maximum connection time out.

For example:

```
usr/config$ h323 -print
```

H.323 stack relate information

```

RAS mode           : GK mode
Gatekeeper IP address : 10.1.1.2
Second Gatekeeper IP : 10.1.1.2
Gateway Type       : Gateway1
Registered prefix number : 0
Line1              : 000
H.235 security token : ***
Registered alias   : 1FXS-01778c
Display Information : 1FXS

```

```

Gatekeeper discovery      : Off
Gatekeeper ID             : GK
RAS TTL time              : 60
RTP port                  : 16384
Gatekeeper finding port   : 1718
GK RAS port               : 1719
H225 RAS port             : 1024
H225 Call signal port     : 1720
Allocated port range      :
    start port            : 2000
    end port              : 19999
Response timeOut          : 15
Connect timeOut           : 60

```

```
usr/config$
```

5.3.12 [line]

Line information and configuration.

```
usr/config$ line
```

Gateway line information and configuration

Usage:

```
line [-print][-config number [hunt number][hotline number]
    [forward number]]
```

```
line -print    Gateway line information.
```

```
    -config    Set Gateway line information.
```

```
                hunt: Hunting group.
```

```
                hotline: Hot line configuration.
```


```
                forward: No answer forward for FXS line.
```

Example:

```
    line -config 1 hunt 1 hotline 1003 forward 1002
```

```
usr/config$
```

Parameter Usages:

- **print**: print out all current settings of line
- **config**: determine which line to configure
 -  **hunt**: Set hunting group flag of each line. User can assign different hunt

group number represent different hunt group.

- 🕒 **hotline:** Set hotline table. The Hotline Mode is applied in limited two channels. User just picks up the phone set of one FXS TEL or calls in one FXO line, and gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device.
- 🕒 **forward:** Set no answer forward table.

For example:

```
usr/config$ line -print
```

Line information and configuration

Index Type Hunt Hotline	No Answer Forward	Registration
Status		

=====

1	FXS	1	x	x	No
---	-----	---	---	---	----

Ready

usr/config\$

5.3.13 [prefix]

This command is for make rules for drop or insert prefix digits.

```
usr/config$ prefix
```

Prefix drop/insert information and configuration

Usage:

prefix [-print]

```
[-add [prefix number][drop used][insert digits]]
```

```
[-modify index [prefix number][drop used][insert digits]]
```

```
[-delete index number]
```

```
prefix -print    Display drop/insert information.
```

-add Add new prefix drop/insert information

prefix : The prefix of dialed number.

drop : Drop prefix (0=Disable, 1=Enable).

insert : 1~10 digits.

- modify Modify prefix drop/insert information
 - index : The prefix index number record.
 - prefix : The prefix of dialed number.
 - drop : Drop prefix (0=Disable, 1=Enable).
 - insert : 1~10 digits.
- delete Delete prefix index number record.

Example:




```

prefix -print
prefix -add prefix 100 drop 1 insert 2000
prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -modify 1 prefix 100 drop 0 insert 300
prefix -delete 1

```

usr/config\$

Parameter Usages:

- add: Add a rule to drop or insert prefix digits of incoming call.
 -  prefix: Set which prefix number to implement prefix rule.
 -  drop: Enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call.
 -  insert: Set which digit to insert on incoming call.

```
usr/config$ prefix -add prefix 100 drop 1 insert 2000
```

- modify: Modify a rule to drop or insert prefix digits of incoming call.

```
usr/config$ prefix -modify 100 drop 0 insert 200
```

- delete: Delete a rule to drop or insert prefix digits of incoming call.

```
usr/config$ prefix -delte modify 100 drop 0 insert 200
```

For example:

```
usr/config$ prefix -print
```

Prefix drop/insert information and configuration

Index	Prefix	Drop	Insert
1	100	Enable	2000

usr/config\$

5.3.14 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users do not have to reboot the machine, and the record will be effective immediately.

usr/config\$ pbook

Phone book information and configuration

Usage:

pbook [-print]

[-add [name string][e164 number][IP address][port number]

[drop used][insert digits]]

[-modify number [name string][e164 number][IP address][port number]

[drop used][insert digits]]

[-delete number]

pbook -print Display phone book information and configuration.

-add Add new phone book record.

name : 1 ~ 20 characters.

e164 : 1 ~ 20 digits.

ip : IP address.

port : 1024 ~ 65535.

drop : Drop prefix (0=Disable, 1=Enable).

insert : 1 ~ 30 digits.

-modify Modify phone book record.

name : 1 ~ 20 characters.

e164 : 1 ~ 20 digits.

ip : IP address.

port : 1024 ~ 65535.

drop : Drop prefix (0=Disable, 1=Enable).

insert : 1 ~ 30 digits.

-delete Delete phone book index record.

Note:

Range of IP address setting (0.0.0.0~255.255.255.255).

Range of index setting value (1~50).

Example:

pbook -print

```

pbook -add name test e164 1234 ip 192.168.1.10
pbook -add name test e164 1234 ip 192.168.1.10 port 1720 drop 1 insert
5678
pbook -modify 1 name test e164 5678 ip 192.168.1.10 port 1720 drop 0
pbook -delete 1

usr/config$

```

Parameter Usages:

- print: Print out current contents of Phone Book. Users can also add index number, from 1 to 100, to the parameter to show specific phone number.

Note:

Index number: means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

- add: add a new record to phone book. When adding a record, users have to specify name, IP, and e164 number to complete the command.
 - 🕒 name: Name to represent caller.
 - 🕒 e164: e.164 number for mapping with IP address of caller
 - 🕒 ip: IP address of caller
 - 🕒 port: Call signal port number of caller
 - 🕒 drop : Drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
 - 🕒 inert: Insert digits.(1~10 digits)

```
usr/config$ pbook -add name test e164 100 ip 192.168.13.78
```

- modify: modify an existing record. When using this command, users have to specify the record's index number, and then make the change.

```
usr/config$ pbook -modify 1 name test e164 5678 ip 192.168.1.10 port 1730
drop 0
```

- delete: delete a specific record. [pbook -delete 3] means delete index 3 record.

```
usr/config$ pbook -delete 3
```

PhoneBook Rules:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it

is matched to its e164, including Line number in some particular device.

For example:

```
usr/config$ pbook -print

Phone book information and configuration
Index  Name      E.164      IP          Port  Drop   Insert
=====
1      test       100        192.168.13.78  1720  Disable
usr/config$
```

5.3.15 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Gateway.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-print]
      [-send [G723 ms] [G729 ms] [G729A ms] [G729B ms] [G729AB ms]
[G711U ms] [G711A ms] ]
voice [-priority [G723] [G729] [G729A] [G729B] [G729AB] [G711U] [G711A] ]
      [-volume line [voice level][input level][dtmf level]]
      [-nscng [G711U used1][G711A used2][G723 used3]]
      [-echo used][mindelay t1][maxdelay t2]


-print      Display voice codec information and configuration.
-send       Specify sending packet size.
            G.723   (30/60/90 ms)
            G.729   (20/40/60/80 ms)
            G.729A   (20/40/60/80
ms) G.729B
            (20/40/60/80 ms) G.729AB
            (20/40/60/80 ms) G.711U
                        (20/40/60 ms)
            G.711A   (20/40/60 ms)
-priority    Priority preference of installed codecs.
            G.723
            G.729
```


	G.729A
	G.729B
	G.729AB
	G.711U
	G.711A
-volume	Specify the following levels: voice volume (0~63, default: 30), input gain (0~63, default: 30), dtmf volume (0~31, default: 23),
-nscng	No sound compression and CNG (G.723.1 only, 0=Off, 1=On).
-echo	Setting of echo canceller. (0=Off, 1=On, per port basis).
-mindelay	Setting of jitter buffer min delay. (0~150, default: 90).
-maxdelay	Setting of jitter buffer max delay. (0~150, default: 150).
Example:	
voice -send g723 60 g729 60 g729a 60 g729b 60 g729ab 60 g711u 60 g711a 60	
voice -volume 1 voice 20 input 32 dtmf 27	
voice -echo 1	
usr/config\$	

Parameter Usages:

- **print**: Print current voice information and configurations.
- **send**: To define packet size for each codec. 20/40/60ms means to send a voice packet per 20/40/60 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729 codec, while 30/60ms is applicable to G.723.1 codec.
- **priority**: Codec priority while negotiating with other h323 device. This parameter determines the listed sequence in h.245 TCS message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the particular codec without others.

```
usr/config$ voice -priority g723 (only select this codec)
usr/config$ voice -priority g723 g729 g711u g711a (select four codecs, and
g723 is the first choice)
```

- **volume**: There are three adjustable value.
 voice volume stands for volume, which can be heard from Gateway side(range 0~63, default: 28).

- 🕒 input gain stands for volume, which the opposite party hears (range 0~38, default: 28).
- 🕒 dtmf volume stands for DTMF volume/level, which sends to its own Line (range 0~31, default: 23).
- nscng: Silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only.

```
usr/config$ voice -nscng g723 1
```

- echo: On or Off the activate each canceler.
- mindelay: The minimum jitter buffer size (Default value= 90 ms).
- maxdelay: The minimum jitter buffer size (Default value= 150 ms).

```
usr/config$ voice -mindelay 90 -maxdelay 150
```

Note:

Be sure to know well the application before you change voice parameters because this might cause incompatibility.

For example:

```
usr/config$ voice -print
```

Voice codec setting relate information

Sending packet size :

G.723.1	: 60 ms
G.729	: 60 ms
G.729A	: 60 ms
G.729B	: 60 ms
G.729AB	: 60 ms
G.711U	: 40 ms
G.711A	: 40 ms

Priority order codec :

g7231 g729 g729a g729b g729ab g711u g711a

Volume levels :

voice volume : 28
input gain : 28
dtmf volume : 23

No sound compress & CNG :

G.723.1	: Off
G.729	: There is no setting
G.729A	: There is no setting

```

G.729B      : There is no setting
G.729AB     : There is no setting
G.711(U-Law) : There is no setting
G.711(A-Law) : There is no setting
Echo canceller : On
Jitter buffer :
Min Delay    : 90
Max Delay    : 150

usr/config$

```

5.3.16 [support]

This command provides some extra functions that might be needed by users.

```
usr/config$ support
```

Special Voice function support manipulation

Usage:

```
support [-print][-t38 used][-t38ecm used][-t38asn1 used][-faxrdd digits]
        [-fstart used][-tunnel used][-h245fs used][-earlyh245 used]
        [-h450 used]
```

support -print Display support information and configuration.

```

-t38      T.38(FAX) (0=Disabled, 1=Enabled).
-t38rq    T.38(FAX) Send request Mode.(0=Disabled, 1=Enabled).
-t38ecm   T.38(FAX) ECM (0=Disabled, 1=Enabled).
-t38asn1  T.38(FAX) ASN.1 support (0=Disabled, 1=Enabled).
-faxrdd   FAX redundancy depth (0 ~ 2).
-fstart   Fast start (0=Disabled, 1=Enabled).
-tunnel   H245 Tunneling (0=Disabled, 1=Enabled).
-h245fs   H245 message after FastStart support (0=Disabled,
1=Enabled).
-earlyh245 EarlyH245 support (0=Disabled, 1=Enabled).
-h450     H450 support (0=Disabled, 1=Enabled).

```

Example:

```

support -t38 1
support -t38rq 1
support -t38ecm 1
support -t38asn1 1

```

```
support -faxrdd 1
support -fstart 1
support -tunnel 1
support -h245fs 1
support -earlyh245 1
support -h450 1
```

```
usr/config$
```

Parameter Usages:

- **print**: print current settings in support command.
- **t38**: Enable or disable T.38 fax ability. The function is will automatically defer codec (G.723 or G.729a) to T.38 when FAX signal is detected.
- **t38rq**: The T.38 could support the request mode message sending or not. 0 for disable the request mode sending and 1 for request mode sending with the T.38.
- **t38ecm**: Enable or disable t38ecm function. The function is support the error correction in the high-speed fax mode.
- **t38asn1**: Enable or disable Enable ASN.1 function. The function is support with the FAX.
- **faxrdd**: Set fax redundancy depth. User can increase FAX redundancy depth when network traffic is heavy. For example, if user set fax redundancy as 2, Gateway will resend fax packets every 2 packets.
- **fstart**: Enable or disable fstartStart function. The function is can shorten the connection time if the opposite party also supports fastStart.
- **tunnel**: Enable or disable H.245 tunneling function. The function is send H.245 (Call Control messages) via H.225's (Call Signal messages) link. The function is effective only when both side support h245 tunnel.
- **h245fs**: Enable or disable Gateway H.245 FastStart. The function is send h.245 messages after FastStart.
- **earlyh245**: Enable or disable early H.245 function. The function is effective only when both sides support early H.245.
- **h450**: Enable or disable H.450 related features, which include transfer, hold and forward.

Note:

It is not recommended to change the value in this command, only if users do know well the application. This might cause incompatibility with other devices.

For example:

```
usr/config$ support -print

Special Voice function support manipulation
  T.38(FAX) support      : Disabled
  T.38(FAX) Request Mode : Enabled
  T.38(FAX) ECM          : Disabled
  T.38(FAX) ASN.1        : Disabled
  FAX redundancy depth   : 0
  FastStart support      : Disabled
  Tunneling support      : Disabled
  H.245 message after FastStart support : Enabled
  EarlyH245 support      : Disabled
  H450 support           : Disabled

usr/config$
```

5.3.17 [sysinfo]

This command could show up the line is busy or not.

```
usr/config$ sysinfo

System information
Index Registration Status
=====
1      No      Ready

usr/config$
```

Parameter Usages:

- **Index:** Line channel.
- **Reg:** Line registration Gatekeeper status. If you use registration Gatekeeper server, then display registration status [No] or [OK].
- **Status:** Line status in use display [busy] or not use display [Ready]. When you set h323 line number [x] disable, the status field not check h323 line number setup.

5.3.18 [phone]

Gateway progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone

specification or users-defined tone specification.

```
usr/config$ phone
```

Phone ringing , ringback tone , busy tone , dial tone setting and notes

Usage:

```
phone [-print [ring][rbt][bt][dt][flash]]
```

```
    [-ring [freq  ][ringON  ][ringOFF ][ringLevel]]
```

```
    [-rbt  [freqLo ][freqHi  ][freqLoLev][freqHiLev]
```

```
          [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
```

```
    [-bt   [freqLo ][freqHi  ][freqLoLev][freqHiLev]
```

```
          [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
```

```
    [-dt   [freqLo ][freqHi  ][freqLoLev][freqHiLev]
```

```
          [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
```

```
    [-flash [freqLo ][freqHi ]]
```

```
phone -print  Display phone ringing/tone configuration.
```

```
    ring  : ringing
```

```
    rbt   : ringback tone
```

```
    bt    : busy tone
```

```
    dt    : dial tone
```

```
    flash : flash tone
```

```
-ring  ringing configuration set.
```

```
-rbt   ringback tone configuration set.
```

```
-bt    busy tone configuration set.
```

```
-dt    dial tone configuration set.
```

```
-flash flash configuration set.
```

Note:

```
ringing frequency   : 15 ~ 100  (Unit : Hz)
```

```
ringing ring ON/OFF : 0  ~ 8000 (Unit : ms)
```

```
ringing level       : 0  ~ 94   (Unit : V) tone
```

```
    frequency       : 0  ~ 65535 (Unit : Hz)
```

```
tone   freqLevel    : 0  ~ 65535 (Unit : mVrms)
```

```
tone   Tone ON/OFF  : 0  ~ 8000  (Unit : ms)
```

Example:

```
phone -print rbt
```

```
phone -ring 20 2000 4000 94
```

```
phone -rbt 440 480 8 8 2000 4000 2000 4000
```

```
phone -bt 620 480 8 8 500 500 1023 1023
```

```
phone -dt 440 350 8 8 500 1023 500 1023
```



```
phone -flash 100 300
```

```
usr/config$
```

Parameter Usages:

- **print**: Print current call progress tone configurations (ring: ring tone, rbt: ring back tone, bt: busy tone, dt: dial tone). This parameter should be accompanied with tone type.
- **ring**: To set RING tone value. The played tone type, when Gateway is receiving a call.
- **rbt**: To set RingBackTone value. The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.
- **bt**: To set BusyTone value. The played tone type, when destination is busy.
- **dt**: To set DialTone value. The played tone type, when hook off a phone set of workable Gateway.
- **flash**: Set the detective flash range in ms, for example, 300-500 ms.

Note:

For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

For example:

```
usr/config$ phone -print rbt
Phone ringback tone paramter
  Ringback Tone frequency high      : 480
  Ringback Tone frequency low       : 440
  Ringback Tone frequency high level : 155
  Ringback Tone frequency low level  : 155
  Ringback Tone tone1 on            : 2000
  Ringback Tone tone1 off           : 4000
  Ringback Tone tone2 on            : 2000
  Ringback Tone tone2 off           : 4000
```

5.3.19 [tos]

Define IP Packet ToS (Type of Service)/ Differentiated Service configuration.

```
Usr/config$ tos
```

IP Packet ToS (Type of Service)/Differentiated Service configuration

Usage:

```
tos [-print][-rtptype dscp][-sigtype dscp]
```

tos -print Display IP packet Tos configuration.

 -rtptype The packages of voice (0~63).

 -sigtype The package of call signal (0~63).

Example:

```
tos -rtptype 7 -sigtype 0
```

```
usr/config$
```

Parameter Usages:

- **rtptype**: the packages of voice (0~63).
- **sigtype**: the package of call signal (0~63).

Note:

The value of rtptype and sigtype is from 0 to 63. It's working if it supported by your network.

For example:

```
usr/config$ tos -print
```

IP Packet ToS information:

 Signalling Packet:

 DSCP Code : 0

 Media Packet :

 DSCP Code : 0

```
usr/config$
```

5.3.20 [ddns]

The dynamic DNS service information and configuration

```
usr/config$ ddns
```

The dynamic DNS service information and configuration

Usage:

```
ddns [-print][-enable used][-server Address][-hostname Name][-id Account]
```

```
[-passwd Password][-checkip used][-checkipsvr Address][-delay time]
[-force IP]
```

```
ddns -print      Display Dynamic DNS information and configuration.
-ddns -enable    Use the dynamic DNS service(0=Disable, 1=Enabled).
-ddns -server    Specify DDNS server address.
-ddns -hostname  Registered domain name.
-ddns -id        Registered account ID.
-ddns -passwd    Registered account password.
-ddns -checkip   Check the host current IP address(0=Disable, 1=Enabled).
-ddns -checkipsvr1 Specify IP address check server.
-ddns -checkipsvr2 Specify secondary IP address check server.
-ddns -delay     Setting the service delay time (1~59 minutes or 1~24
horus).
-ddns -force     Force execute the dynamic DNS service.
```

Example:

```
ddns -print
ddns -enable 1
ddns -server member.dyndns.org -hostname ipphone.dyndns.org
ddns -delay 30 m (30 minutes)
ddns -delay 12 h (12 hours)
ddns -force 11.22.33.44
```

```
usr/config$
```

Parameter Usages:

- **enable**: Enable or disable the DDNS function.
- **server**: Enter the server address of the DDNS server you use.
- **hostname**: Enter the domain name address which you get from the DDNS server.
- **id**: Enter your login DDNS server id.
- **passwd**: Enter your login DDNS server password.
- **checkIP**: Enable or disable the check current user's IP address.
- **checkIPsvr1**: Enter the IP address of the check server.
- **checkIPsvr2**: Enter the secondary IP address of the check server.
- **delay**: Enable or disable the service delay time.
- **force**: Execute the DDNS function all the times.

Note:

Support DDNS Server: www.dyndns.org.

For example:

```
usr/config$ ddns -print

Dynamic DNS service information
  Status           : Enable
  Server           : www.dyndns.org
  Host Name        : totoro609.dnsalias.net
  ID               : totoro609
  Password         : 123456789
  Check IP         : Disable
  IP Check Server1 : checkip.dyndns.org
  IP Check Server2 : checkip.dyndns.org
  Delay            : Off

usr/config$
```

5.3.21 [pt]

RTP payload type configuration and information

```
usr/config$ pt

RTP payload type configuration and information
Usage:
pt [-print][-rfc2833 type][-dtmf type][-fax type][-faxbypass type]
  [-modembypass type][-redundancy type][-modemrelayp type]

pt -print          Display the RTP payload type information.
  -rfc2833         Configure the DTMF RFC2833 payload type.
  -dtmf            Configure the DTMF payload type.
  -fax             Configure the FAX payload type.
  -faxbypass       Configure the FAX ByPass payload type.
  -modembypass     Configure the MODEM ByPass payload type.
  -redundancy      Configure the Redundancy payload type.
  -modemrelay      Configure the MODEM Relay payload type.

Example:
  pt -rfc2833 96 -fax 101

usr/config$
```

For example:

```
usr/config$ pt -print

RTP payload type information
  RFC2833 payload type      : 96
  DTMF payload type        : 100
  FAX payload type         : 101
  FAX ByPass payload type   : 102
  MODEM ByPass payload type : 103
  Redundancy payload type   : 104
  MODEM Relay payload type  : 105

usr/config$
```

5.3.22 [rom]

ROM file information and firmware upgrade function.

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print] [-app] [-boot] [-dsptest] [-dspcore] [-dspapp]
    [-ht] [-method used] [-boot2m]
    -s TFTP/FTP server ip -f filename

rom -print
  -print      show versions of rom files. (optional)
  -app        update main application code(optional)
  -boot       update main boot code(optional)
  -boot2m     update 2M code(optional)
  -ht         updata Hold Tone PCM file(optional)
  -dsptest    update DSP testing code(optional)
  -dspcore    update DSP kernel code(optional)
  -dspapp     update DSP application code(optional)
  -s          IP address of TFTP/FTP server (mandatory)
  -f          file name(mandatory)
  -method     download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
  -ftp        specify username and password for FTP
  -server     specify EMS Server IP address

Note:
```

This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.

Example:

```
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
```

usr/config

Parameter Usages:

- **print**: show versions of all rom files.
- **app, boot, boot2m, dsptest, dspcore, dspapp, ht**: To update main Application program code, Boot code, DSP testing code, DSP kernel code, or DSP application code, and Hold Tone file.

Note:

Most of all, the Rom file needed to get upgrade is App or Boot2m. Please check the exactly Rom file before doing download procedure.

- **s**: To specify TFTP server's IP address when upgrading ROM files.
- **f**: To specify the target file name, which will replace the old one.
- **method**: To decide using TFTP or FTP as file transfer server. [0] stands for TFTP, while [1] stands for FTP.
- **ftp**: If users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

For example:

```
usr/config$ rom -print
```

```
Download Method : TFTP
Boot Rom       : sdboot.200
Application Rom : IAD161_162.100
DSP App        : 48302ce3.140
DSP Kernel    : 48302ck.140
DSP Test Code  : 483cbit.bin
```

```
usr/config$
```

5.3.23 [auth]

For security concern, the "root" user can customize some configurable items for "administrator" user.


```
usr/config$ auth
```

Auth to customized the WEB configuration item for administrator.

Usage:

auth -print Display auth switch configuration.

Use item name to do config name (0=Disable, 1=Enabled).

Example: auth -ifaddr 1

```
usr/config$
```

Parameter Usages:

- "item name": Assign the configurable item for "administrator" user.

```
usr/config$ auth -ifaddr 1
```

```
usr/config$ auth -h323 1
```

```
usr/config$ auth -voice 1
```

- print: Display the configurable items for "administrator" user.

For example:

```
usr/config$ auth -print
```

Auth to customized the WEB configuration item for administrator.

```
ifaddr   : Enable
h323     : Enable
line     : Disable
pbook    : Enable
support  : Disable
sysconf  : Disable
voice    : Enable
phone    : Disable
rtp      : Disable
tos      : Disable
ddns     : Disable
prefix   : Disable
passwd   : Enable
version  : Enable
rom      : Disable
flash    : Disable
```

```
usr/config$
```

5.3.24 [passwd]

For security concern, users have to input the password before entering configuration mode. [passwd] command is for password setting purpose.

```
usr/config$ passwd
```

Password setting information and configuration

Usage:

```
passwd [-set [Login name] [Password]][-clean]
```

```
passwd -set      Loginname Password.
```

```
      -clean    Clear all password stored in flash.
```

Note:

1. Loginname can be only 'root' or 'administrator'
2. passwd -clean will clear all passwd stored in flash, please use it with care.

Example:

```
passwd -set root Your_Passwd_Setting
```

```
passwd -clean
```

```
usr/config$
```

Parameter Usages:

- **set**: Set login name and password, input login name then input new password.
- **clean**: Will clear all password setup, and change null.

Note:

Gateway Login name only use [root] or [administrator]. [root] and [administrator] have the same authorization, except commands that can be excuted by [Login name: root] only [passwd –set root], [rom –boot], [room-boot2m] and [flash –clean].

For example:

```
usr/config$ passwd -set root root1234
```

Setting

login: root

Password: root1234

OK

```
usr/config$
```

```
sr/config$ passwd -clean
```

```
Please wait a moment!!
```

```
Clean password OK.
```

```
usr/config$
```