



# Dynamix DW IAD - 162

## VoIP Gateway (SIP)



## 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 theDynamix IAD – 162 VoIP Gateway.

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## Part I: Dynamix IAD -162 Gateway Overview

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This part introduces the software/hardware specifications and default settings of Dynamix IAD – 162 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 - 161 provides IP telephone number for end users with FreeTalk voice service. User can make phone call via Internet now. No more long distance and international telephony fee! It also connects three computers without another IP sharing as showed as following diagram.

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.
- \* SIP RFC 3261 compliance
- \* Built-in NAT/IP sharing function
- \* Provided call features: Hold, forward and transfer
- \* Automatic FAX detection (Support T.38 protocol)
- \* Codec: G.711 a/μlaw, G.723.1, G.729A
- \* PPPoE connection
- \* VAD (Voice Activity Detection), CNG (Comfort Noise Generate)
- \* G.168/165 echo cancellation
- \* FSK and DTMF Caller ID
- \* Provide both IP telephone number and PSTN telephone number in one device for end users (Dynamix 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
- \* VAD (Voice Activity Detection)
- \* CNG (Comfort Noise Generation)
- \* G.168/165-compliant adaptive echo cancellation
- \* Dynamic Jitter Buffer
- \* Bad Frame Interpolation
- \* Voice/DTMF Gain Settings

### **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 Proxy mode and successfully register to Proxy.
- (2) Light off: Under Peer-to-Peer mode.
- (3) Light Blanking: IAD is under Proxy mode and not successfully register to Proxy.

✱ **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):** (Dynamix 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**

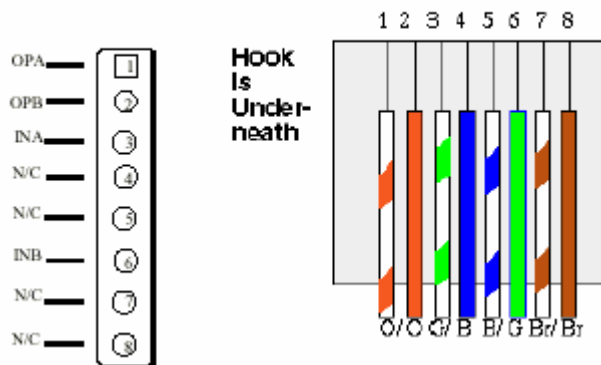


✱ **Reset Button:**

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.

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## Part II: Start-UP

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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  
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: SIP Proxy Server (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 Dynamix IAD - 162 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 IAD 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 IAD Gateway, please make sure you have connected with the correct power adapter.**

### 2.1.3 Essential Configuration via Web Management interface

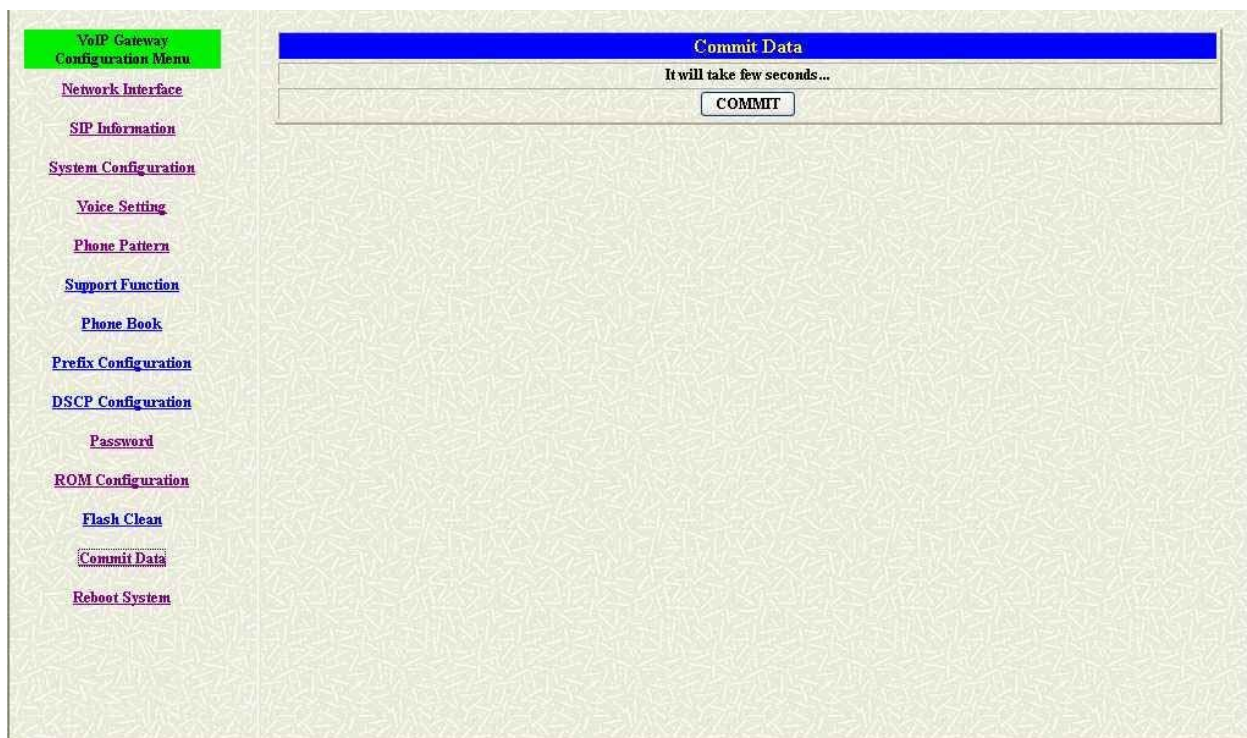
This section describes how to setup IAD via Web management interface. Please follow procedures below to configure essential items before you use IAD gateway.

#### 2.1.3.1 Save Data and Reboot

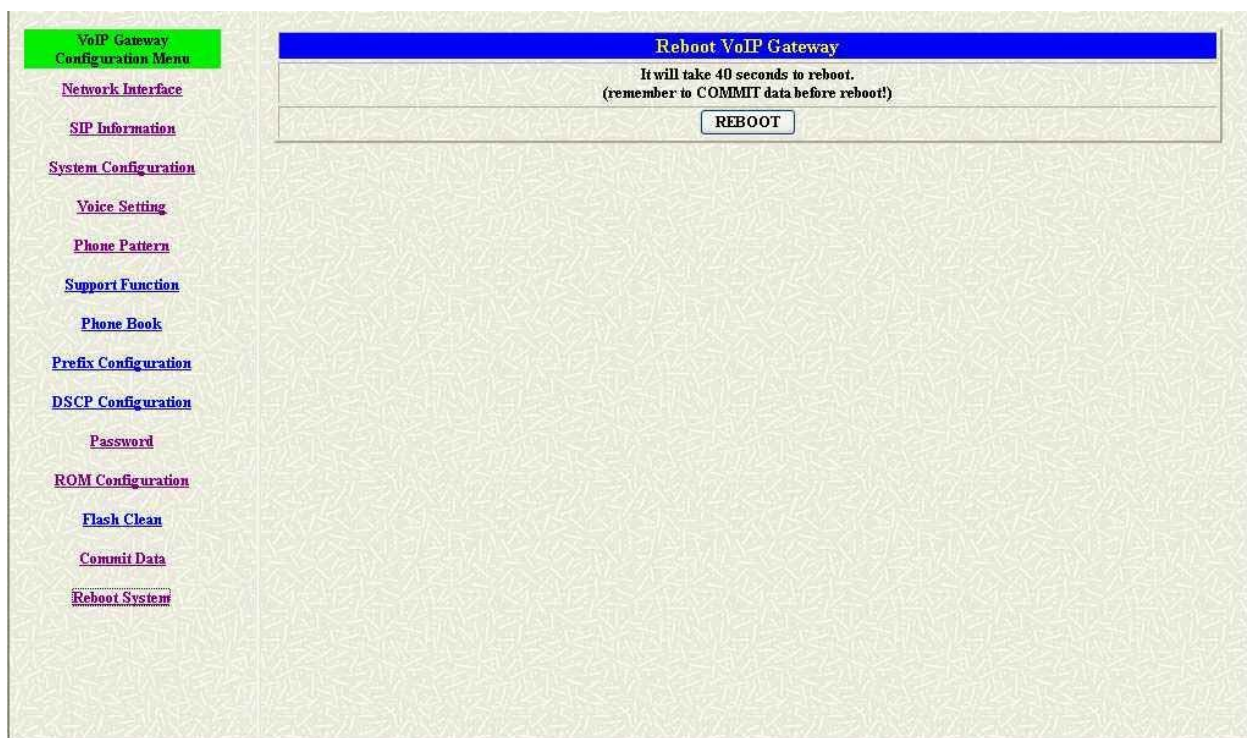
After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

**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, choose the IP mode, 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.

VoIP Gateway Configuration Menu		Network Interface	
<a href="#">Network Interface</a>		IP Mode:	<input checked="" type="radio"/> Static <input type="radio"/> DHCP <input type="radio"/> PPPoE
<a href="#">SIP Information</a>		LAN IP Address:	192 . 168 . 123 . 123
<a href="#">System Configuration</a>		WAN IP Address:	192 . 168 . 13 . 62
<a href="#">Voice Setting</a>		Subnet Mask:	255 . 255 . 248 . 0
<a href="#">Phone Pattern</a>		Default routing gateway:	192 . 168 . 8 . 254
<a href="#">Support Function</a>		NAT:	<input checked="" type="radio"/> enable <input type="radio"/> disable
<a href="#">Phone Book</a>		DNS Server Obtained:	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
<a href="#">Prefix Configuration</a>		Primary DNS Server:	168 . 95 . 192 . 1
<a href="#">DSCP Configuration</a>		Secondary DNS Server:	168 . 95 . 1 . 1
<a href="#">Password</a>		SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
<a href="#">ROM Configuration</a>		SNTP Server Address:	168 . 95 . 195 . 12
<a href="#">Flash Clean</a>		GMT:	8
<a href="#">Commit Data</a>		IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
<a href="#">Reboot System</a>		IP Sharing Server Address:	210 . 59 . 163 . 198
		PPPoE User Name:	84460791@kinet.net

**(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.

VoIP Gateway Configuration Menu	
<a href="#">Network Interface</a>	<b>Network Interface</b>
<a href="#">SIP Information</a>	
<a href="#">System Configuration</a>	
<a href="#">Voice Setting</a>	
<a href="#">Phone Pattern</a>	
<a href="#">Support Function</a>	
<a href="#">Phone Book</a>	
<a href="#">Prefix Configuration</a>	
<a href="#">DSCP Configuration</a>	
<a href="#">Password</a>	
<a href="#">ROM Configuration</a>	
<a href="#">Flash Clean</a>	
<a href="#">Commit Data</a>	
<a href="#">Reboot System</a>	

IP Mode:	<input type="radio"/> Static <input checked="" type="radio"/> DHCP <input type="radio"/> PPPoE
LAN IP Address:	192 . 168 . 123 . 123
WAN IP Address:	192 . 168 . 15 . 211
Subnet Mask:	255 . 255 . 248 . 0
Default routing gateway:	192 . 168 . 8 . 254
NAT:	<input checked="" type="radio"/> enable <input type="radio"/> disable
DNS Server Obtained:	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Primary DNS Server:	168 . 95 . 192 . 1
Secondary DNS Server:	168 . 95 . 1 . 1
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
IP Sharing Server Address:	210 . 59 . 163 . 198
PPPoE User Name:	84460791 @kinet.net

### 2.1.3.3 Application mode-Proxy/Peer-to-Peer Mode

After setting IP address, user must assign IAD to work under Proxy mode or Peer-to-Peer mode. If there is no Gatekeeper, please set your Dynamix IAD 162 as Peer-to-Peer Mode.

#### **2.1.3.3.1 Proxy mode**

Proxy mode means that there will be an intermediate Proxy Server between Dynamix IAD -162 Gateway and the remote entity. While operating at this mode, Dynamix IAD -162 Gateway will first register to the Proxy Server located at the ISP side. For the following operation, it sends the INVITE message to the Proxy Server once you initiate a session. Then the Proxy server will forward the INVITE message to the right place. And the Response message from the remote entity will be forwarded back to you via Proxy server.

**Step 1.** Configure the Dynamix IAD -162 Gateway SIP Configuration. Click SIP Information on the navigation panel. In the SIP Information screen, select Proxy routed Mode function.

**Step 2.** Set the SIP information from your service provider: Proxy IP Address, Line1 Number, Line1 Account, Line1 Password, and click the OK button.

**Note:**

1. Please contact with your Proxy vendor to obtain user account information.
2. If no need to enter password, user also has to set security information, please set "name" the same with line number.
3. Dynamix IAD -162 Gateway uses "line number" to register to proxy server, the "name" is only for authentication.

**(3) PPPoE**

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

VoIP Gateway Configuration Menu		Network Interface	
<a href="#">Network Interface</a>	IP Mode: <input type="radio"/> Static <input type="radio"/> DHCP <input checked="" type="radio"/> PPPoE		
<a href="#">SIP Information</a>	LAN IP Address: 192 . 168 . 123 . 123		
<a href="#">System Configuration</a>	WAN IP Address: 192 . 168 . 13 . 62		
<a href="#">Voice Setting</a>	Subnet Mask: 255 . 255 . 248 . 0		
<a href="#">Phone Pattern</a>	Default routing gateway: 192 . 168 . 8 . 254		
<a href="#">Support Function</a>	NAT: <input checked="" type="radio"/> enable <input type="radio"/> disable		
<a href="#">Phone Book</a>	DNS Server Obtained: <input type="radio"/> Auto <input checked="" type="radio"/> Manual		
<a href="#">Prefix Configuration</a>	Primary DNS Server: 168 . 95 . 192 . 1		
<a href="#">DSCP Configuration</a>	Secondary DNS Server: 168 . 95 . 1 . 1		
<a href="#">Password</a>	SNTP: <input checked="" type="radio"/> enable <input type="radio"/> disable		
<a href="#">ROM Configuration</a>	SNTP Server Address: 168 . 95 . 195 . 12		
<a href="#">Flash Clean</a>	GMT: 8		
<a href="#">Commit Data</a>	IP Sharing: <input type="radio"/> enable <input checked="" type="radio"/> disable		
<a href="#">Reboot System</a>	IP Sharing Server Address: 210 . 59 . 163 . 198		
	PPPoE User Name: 84460791@hinet.net		

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Primary Proxy IP Address:	218.32.223.140
Secondary Proxy IP Address:	null
Outbound Proxy:	null
Proxy port:	5060
Outbound Proxy port:	5060
Prefix String:	null
Line1 Number:	070000123
Line1 Account:	070000123
Line1 Password:	*****
SIP port:	5060
RTP Port:	16384
Expire:	60
OK	

#### 2.1.3.3.2 Peer-to-Peer Mode

Peer-to-Peer Mode allows users to call other VoIP devices without the proxy server. 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 IAD Gateway, follow the steps below:

**Step 1.** Configure the Dynamix IAD -162 Gateway SIP information. Click [SIP information] on the navigation panel. In the SIP information screen, select Peer-to-Peer Mode function, set line number, line account and click the [OK] button. Line account must be the same with Line number.

The screenshot shows the 'SIP Configuration' page. On the left is a navigation menu with options: Network Interface, SIP Information, System Configuration, Voice Setting, Phone Pattern, Support Function, Phone Book, Prefix Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The main area is titled 'SIP Configuration' and contains the following fields:

Run Mode:	<input checked="" type="radio"/> Peer-2-Peer <input type="radio"/> Proxy
Primary Proxy IP Address:	218.32.223.140
Secondary Proxy IP Address:	null
Outbound Proxy:	null
Proxy port:	5060
Outbound Proxy port:	5060
Prefix String:	null
Line1 Number:	070000123
Line1 Account:	070000123
Line1 Password:	.....
SIP port:	5060
RTP Port:	16384
Expire:	60
<input type="button" value="OK"/>	

**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.

The screenshot shows the 'Phone Book' page. On the left is the same navigation menu as in the previous screenshot. The main area is titled 'Phone Book' and contains a table with the following columns: Index, Name, IP\_Address, e164, and Port. Below the table is a 'New Record' form with the following fields:

Index	Name	IP_Address	e164	Port

New Record				
Index: <input type="text"/>	Name: <input type="text"/>	IP Address: <input type="text"/>	E164 No.: <input type="text"/>	Port No.: <input type="text"/>
<input type="button" value="Add Data"/> <input type="button" value="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) Fixed IP**

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

**Step 4.** Commit and reboot IAD.

```
usr/config$ commit
```

```
usr/config$ reboot
```

**Step 5.** When IAD successfully establish PPPoE connection, use command [ifaddr -print] to see detail information.

*For example:*

```
IP mode : PPPoE
PPPoE adapter information
  Status           : Ready
  IP address        : 59.121.139.25
  Destination       : 59.121.128.254
  LAN IP address    : 192.168.123.123
  NAT enabled       : ON
  DNS Server Obtained : Manual
  DNS primary       : 168.95.192.1
  DNS secondary     : 168.95.1.1
  Subnet Mask       : 255.255.255.255
  Authenticate      : PAP
  Protocol          : TCP/IP
  Device            : PPP/PPPoE
  PPPoE user name    : 123@hinet.net
  PPPoE password     : 123
  PPPoE reboot      : Yes
  PPPoE echo        : Enable

  EMS IP Address    : 192.168.1.1
  EMS User ID       : vwusr
  EMS Password      : *****
  EMS cycle time    : 0
usr/config$
```

#### 2.1.4.3 Application mode-Proxy/Peer-to-Peer Mode

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

#### **2.1.4.3.1 Proxy mode**

Proxy mode means that there will be an intermediate Proxy Server between Dynamix IAD -162 Gateway and the remote entity. While operating at this mode, Dynamix IAD -162 Gateway will first register to the Proxy Server located at the ISP side. For the following operation, it sends the INVITE message to the Proxy Server once you initiate a session. Then the Proxy server will forward the INVITE message to the right place. And the Response message from the remote entity will be forwarded back to you via Proxy server.

**Step 1.** Set Proxy Mode, using “sip” command

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

Mode 0 is for Peer-To-Peer mode, while mode 1 is for Proxy mode.

**Step 2.** You must specify Proxy address obtained from your service provider. And the Proxy address can be IPv4 address as well as DNS name.

Several important SIP parameters are listed below when setting proxy mode: “-px”, “-line1”.

For example:

```
usr/config$ sip -px 210.68.222.33 -line1 12345
```

In this case is to set proxy IP address as “210.68.222.23”, line number as “12345”.

**Step 3.** You must configure the accounts using “security” command.

An example is demonstrated below:

```
usr/config$ security -line 1 -name 12345 -password 12345
```

This is to set username (userid) as “12345”, password as “12345” into line1, which means line1 can accept incoming calls after successfully registered to Proxy server.

Note:

1. Please contact with your Proxy vendor to obtain user account information.
2. If no need to enter password, user also has to set security information, please set “name” the same with line number.

#### **2.1.4.3.2 Peer-to-Peer Mode**

Peer-to-Peer Mode allows users to call other VoIP devices without the proxy

server. 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 IAD Gateway, follow the steps below:

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

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

Mode 0 is for Peer-To-Peer mode, while mode 1 is for Proxy 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.

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## Part III: Special Applications and Features

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This part explains how to configure Dynamix IAD – 162 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 Proxy for Dynamix IAD -162 Gateway to get register on is a must. In other word, 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.

Network Interface	
IP Mode:	<input checked="" type="radio"/> Static <input type="radio"/> DHCP <input type="radio"/> PPPoE
LAN IP Address:	192 . 168 . 123 . 123
WAN IP Address:	192 . 168 . 13 . 62
Subnet Mask:	255 . 255 . 248 . 0
Default routing gateway:	192 . 168 . 8 . 254
NAT:	<input checked="" type="radio"/> enable <input type="radio"/> disable
DNS Server Obtained:	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Primary DNS Server:	168 . 95 . 192 . 1
Secondary DNS Server:	168 . 95 . 1 . 1
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
IP Sharing Server Address:	210 . 59 . 163 . 198
PPPoE User Name:	84460791@hinet.net

**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.

### 3.1.1.2 More than one IAD behind the same IP Sharing Device

1. Assign an IP address to each IAD using fixed address.
2. Enable the IP Sharing function for each set using following command.  
Fixed IP Address – *usr/config\$ ifaddr -ipsharing 1 "public IP of IP Sharing"*
3. Configure separate SIP port and RTP port for each set to prevent from port conflict. For example, if set A uses the default settings (SIP port: 5060, RTP port: 16384), you must change set B's setting to SIP port equal to 5061 and RTP port equal to 26384 for instance.  
Change SIP port – *usr/config\$ sip -port 5061*  
Change RTP port– *usr/config\$ sip -rtp 26384*
4. Use the Port Forwarding or Port Redirection function provided by IP Sharing device (Router). See following for example.

#### >Advanced setting > NAT setting > Port Redirection

##### Active Configuration

Items	Service name	Protocol	Actual Port	Virtual IP	Virtual Port	Enable
1	1	UDP	5060	192.168.1.10	5060	V
2	2	UDP	16384	192.168.1.10	16384	V
3	3	UDP	16394	192.168.1.10	16394	V
4	4	UDP	5061	192.168.1.11	5061	V
5	5	UDP	26384	192.168.1.11	26384	V
6	6	UDP	26394	192.168.1.11	26394	V
7		---	0		0	X
8		---	0		0	X
9		---	0		0	X
10		---	0		0	X

**Note:**

With Dynamic WAN IP Address, when the WAN IP is changed, we need to change the external IP of Dynamix IAD -162 Gateway using above command.

1. Different Vendor's Router will have different appearance of setting.
2. Once you set the DMZ Host, you don't need to configure the Port Forwarding and vice versa.
3. If there is only one IAD Gateway attached to the IP Sharing device, it is recommended to use DMZ Host setting to enable the NAT traverse and disable the Port Forwarding.
4. If there are two or more sets of IAD Gateway attached to the IP-Sharing device, please configure the Port Redirection (Forwarding) to enable the NAT traverse and disable the DMZ Host.
5. After the IP Sharing configuration of Dynamix IAD -162 Gateway and IP Sharing device is complete, you must reboot the Dynamix IAD -162 Gateway to activate the new settings.

## 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
```

**For example:**

```
usr/config$ ifaddr -print

IP mode : PPPoE
PPPoE adapter information
    Status           : Ready
    IP address        : 59.121.139.25
    Destination       : 59.121.128.254
    LAN IP address     : 192.168.123.123
    NAT enabled        : ON
    DNS Server Obtained : Manual
    DNS primary        : 168.95.192.1
    DNS secondary      : 168.95.1.1
    Subnet Mask        : 255.255.255.255
    Authenticate       : PAP
    Protocol           : TCP/IP
    Device             : PPP/PPPoE
    PPPoE user name    : 84460791@hinet.net
    PPPoE password     : *****
    PPPoE reboot       : Yes
    PPPoE echo         : Enable

    EMS IP Address     : 192.168.1.1
    EMS User ID        : vwusr
```

EMS Password	:	*****
EMS cycle 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 Call Hold, Transfer and Forward

Gateway provides call features including call hold, transfer and forward. Please be noted that both calling and called site have to support this feature. For call forward function, it only works under Proxy mode. Of course, Proxy must support these call features, too.

It is better for user to prepare a telephone set supported [FLASH] function on keypad. If telephone set does not support [FLASH] function on keypad, user can click the Hook quickly by sending FLASH message.

**Note:**

The default FLASH length for Gateway is between 400ms to 800 ms.. This value must be compliant with your phone set, if user press flash but not work, please check the flash time value of your phone set and adjust it on IAD.

#### 3.3.1 Call Hold – press [FLASH]

By pressing the FLASH after making a call, both sites shall hear the 2nd dial tone generated by Gateway. To retrieve the call back, just press the FLASH again.

#### 3.3.2 Call Transfer – press [FLASH], then [transferring number]

A makes a call to B, B press FLASH, A and B hear 2nd dial tone, B presses C's number, C will ring, A will hear Ring Back tone, B Hangs up this call, and A and C can communicate.

#### 3.3.3 Call Forward:

User has to activate/deactivate call forward function via pressing keypad of phone set. This function is only available under Proxy mode, and the Proxy must support Call Forward function. There are three conditions for user to set forward function:

##### 3.3.3.1 No response/ Answer:

While no one answers the call, incoming call will be forwarded to the assigned number.

- (1) Activate: \*75 [Forward No.] #
- (2) Deactivate: #75#

### 3.3.3.2 Busy Forward:

While line is engaged or phone set is been off-hook, incoming call will be forwarded to the assigned number.

- (1) Activate: \*76 [Forward No.] #
- (2) Deactivate: #76#

### 3.3.3.3 Unconditional:

Incoming call will be forwarded to the assigned number unconditionally.

- (1) Activate: \*77 [Forward No.] #
- (2) Deactivate: #77#

## 3.4 Upgrade Your Dynamix IAD – 162 VoIP Gateway

### 3.4.1 Upgrade via Web management interface

#### 3.4.1.1 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 [SIP Information], [Phone Book].

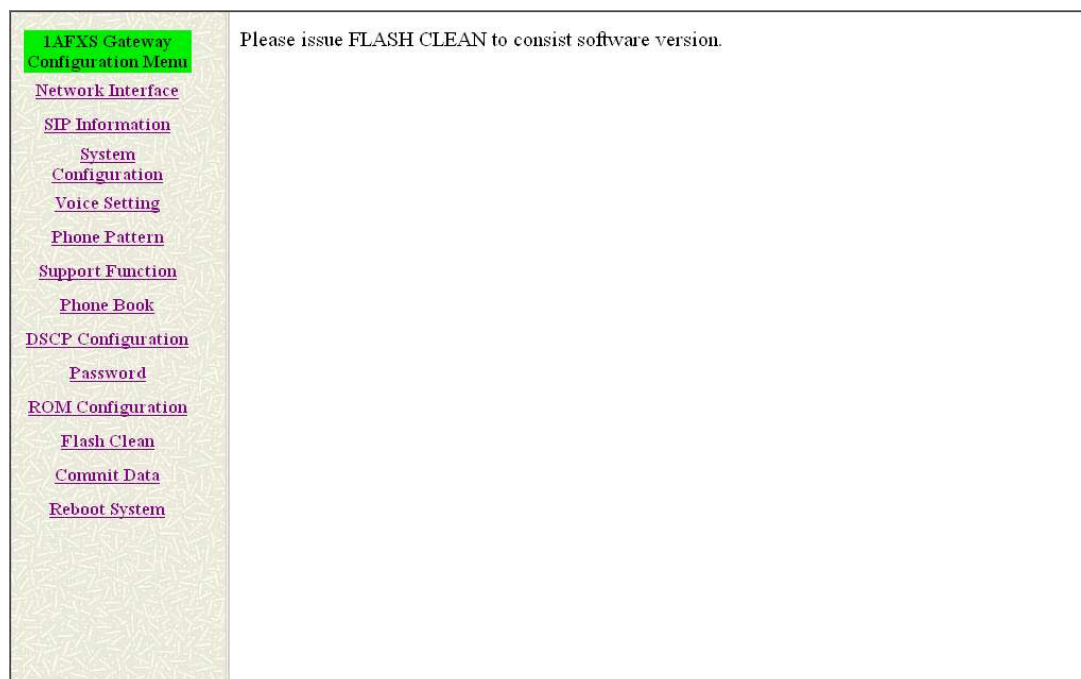
#### 3.4.1.2 Upgrade Version

**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: 192.168.0.100, Target File Name: 162SIP.101, Method: TFTP, Target File Type: Application image) and click the [OK] button.

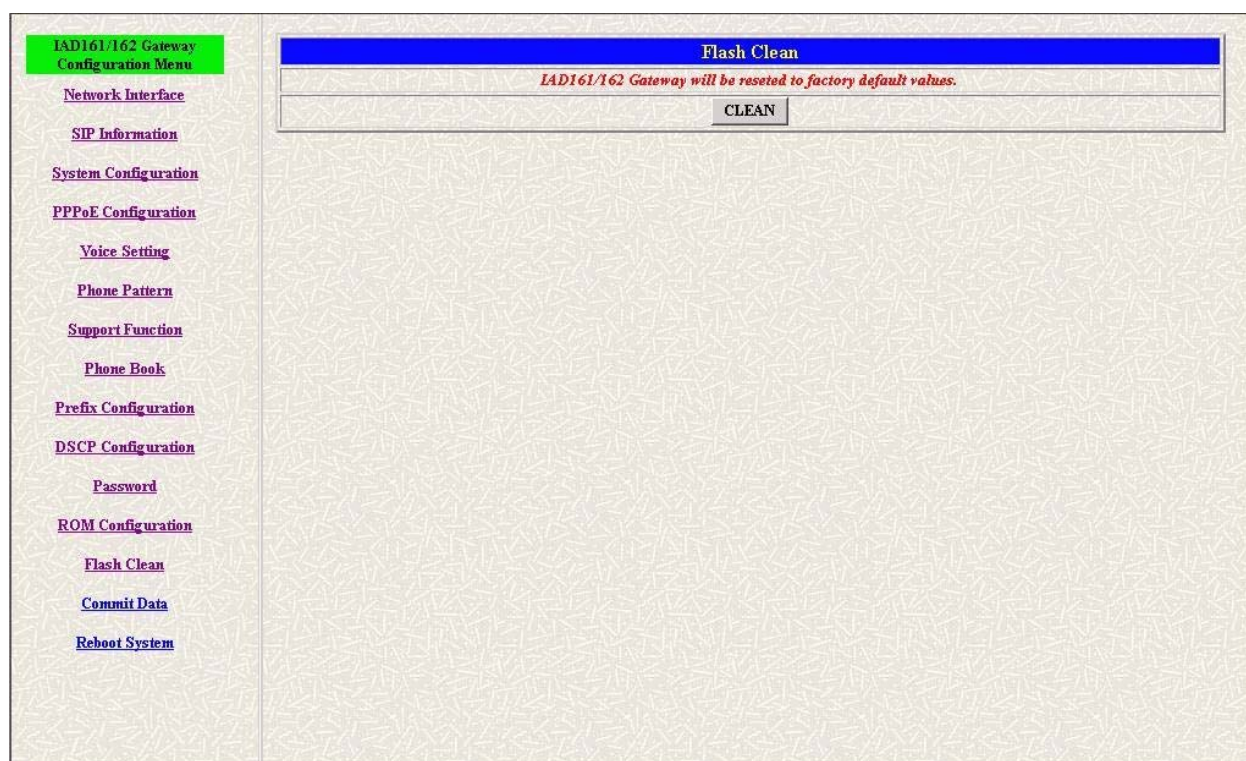
The screenshot shows the 'IAD161/162 Gateway Configuration Web Server' interface in Microsoft Internet Explorer. The left navigation panel lists various configuration options: Network Interface, SIP Information, System Configuration, PPPoE Configuration, Voice Setting, Phone Pattern, Support Function, Phone Book, Prefix Configuration, DSCP Configuration, Password, ROM Configuration (highlighted), Flash Clean, Commit Data, and Reboot System. The main content area displays the 'ROM Configuration' form with the following fields:

FTP/TFTP server IP Address:	192 . 168 . 0 . 100
Target File name:	162SIP.101
Method:	TFTP
FTP Login:	name: [ ] passwd: [ ]
Target File Type:	Application Image
OK	

**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.

### 3.4.2 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] [-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
```

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  : 1asiplAD.107
      DSP App    : 48302ce3.140
      DSP Kernel : 48302ck.140
      DSP Test Code : 483cbit.bin

      Hold Tone  : holdtone.101

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
```

## **3.5 Dynamix IAD - 162 PSTN Line Application**

### **3.5.1 PSTN Outgoing Call**

#### **3.5.1.1 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.

#### **3.5.1.2 PSTN Backup**

If IAD is working under Proxy mode, and fail to register to Proxy server, 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 “\*#”.

### **3.5.2 PSTN Incoming Call**

#### **3.5.2.1 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.

#### **3.5.2.2 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.

### **3.5.3 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.

### **3.5.4 VoIP Incoming Call**

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

## 3.6 Hotline mode

The Hotline Mode is applied in limited two peers. User just picks up the phone set and then hears ring back tone or dial tone depended on configurations of destination device.

**Step 1.** Specify gateway service type as Hotline service.

**Step 2.** Create a Hotline table with [line] command.

```
usr/config$ sysconf -service 1
usr/config$ bureau -hotline 1 10.2.2.2 201
```

This example means that if user picks up phone set of FXS Line1, gateway will automatically dial out IP address of [201].

**Note:**

If this gateway is under P2P mode, please set the phone book firstly.

The IP address of “bureau” command indicates the IP address of called party in P2P mode, or the proxy server IP address in proxy mode.

**Step 4.** After the configuration, [commit] and [reboot] the device.

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

### 3.7 Reset Button Feature



IAD 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 161/162 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 161/162 may crash and break, caused by the Flash error.

---

## Part IV: Web Management Interface

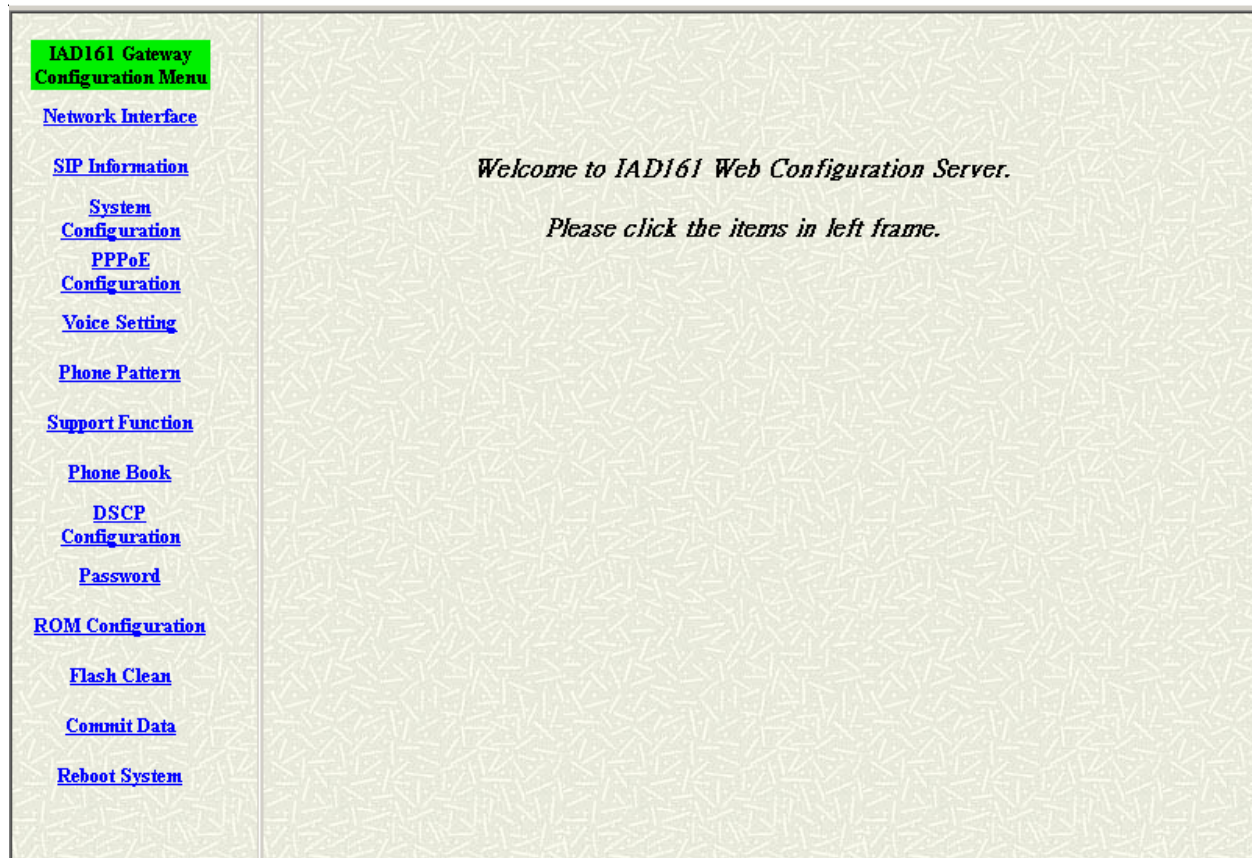
---

---

This part explains how to configure the Dynamix IAD – 162 Gateway via WEB management interface.

## 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**” 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
LAN IP Address:	192 . 168 . 123 . 123
WAN IP Address:	192 . 168 . 10 . 75
Subnet Mask:	255 . 255 . 248 . 0
Default routing gateway:	192 . 168 . 8 . 254
NAT:	<input checked="" type="radio"/> enable <input type="radio"/> disable
DNS Server Obtained:	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Primary DNS Server:	168 . 95 . 192 . 1
Secondary DNS Server:	168 . 95 . 1 . 1
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
IP Sharing Server Address:	210 . 59 . 163 . 198
PPPoE User Name:	pppoe

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
PPPoE User Name:	pppoe
PPPoE Password:	*****
PPPoE IP Address:	
PPPoE Destination:	
Reboot After Remote Host Disconnection:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Send PPPoE Echo Request:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
EMS Address:	192.168.1.1
EMS User Name:	vwusr
EMS Password:	*****
EMS Time:	0
OK	

- **IP Mode:** Select use defines the networking type for this gateway. It could support the Static, DHCP and PPPoE function.
- **LAN IP Address:** Set LAN IP Address of IAD (range: 192.168.1.1-192.168.254.254)

- **WAN 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
- **NAT:** Enable / Disable the Network Address Translation function
- **DNS Server Obtained:** Auto or Manual configure the DNS server, user can manual configure the Gateway's DNS server IP address when it is under PPPoE or DHCP if they set it as "manual". Default is "auto".
- **Primary DNS Server:** 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 Proxy and phone book IP address.
- **Secondary DNS Server:** Set Secondary Domain Name Server IP address.
- **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:** Set Public IP Address of IP Sharing router for IAD to work behind IP sharing.
- **PPPoE User Name:** Set the PPPoE connection account in this table. Please get this info from your ISP.
- **PPPoE Password:** Set the PPPoE connection password in this table. Please get this info from your ISP.
- **PPPoE IP Address:** The field display the IP address. When VoIP Gateway after the connection success, which the gateway got from the ISP.
- **PPPoE Destination:** The field display the default gateway address. When VoIP Gateway after the connection success, which the gateway got from the ISP.
- **Reboot After Remote Host Disconnection:** Select enable or disable this function will make the gateway restart automatically if the PPPoE connection is disconnected or the IP address was taken back by the ISP.
- **Send PPPoE Echo Request:** When the IP mode is in PPPoE mode, select enable or disable this function will make the gateway send echo request packet or not.
- **EMS IP:** Set EMS server IP address.
- **EMS User Name:** Set EMS authentication user name.
- **EMS Password:** Set EMS authentication password
- **EMS Time:** Set EMS refresh time.

---

**Note:**

1. When IAD is behind IP sharing device, if Proxy support behind NAT function, both IAD and IP sharing don't need to do any configuration. Please contact with your proxy vendor more correct information before configuring IAD.
2. About the EMS function, please refer to the EMS user's manual for more detail information.

### 4.3.2 SIP Information Screen

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

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Primary Proxy IP Address:	10.1.1.2
Secondary Proxy IP Address:	null
Outbound Proxy:	null
Proxy port:	5060
Outbound Proxy port:	5060
Prefix String:	null
Line1 Number:	1001
Line1 Account:	1001
Line1 Password:	****
SIP port:	5060
RTP Port:	16384
Expire:	60
OK	

- **Run Mode:** Select IAD to work under Peer-to-Peer mode or Proxy mode.
- **Primary Proxy IP Address:** Set primary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Interface**).
- **Secondary Proxy IP Address:** Set secondary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Interface**). When IAD fail to register to primary Proxy, it will try to register to secondary Proxy, when it fails again, it will retry to register to Primary Proxy.
- **Outbound Proxy:** Set IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Configure**) of outbound Proxy server.
- **Proxy port:** Set Proxy port for IAD to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- **Outbound proxy port:** Set outbound Proxy port for IAD to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- **Prefix String:** set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, IAD will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, IAD will set out as abc123@"IP address of Proxy". This function is for special proxy server.
- **Line Number:** identify one number for the IAD to register to the Proxy.

- **Line Account:** set user name of IAD for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or IAD won't register successfully.
- **Line Password:** set password for registering.
- **SIP Port:** set SIP UDP port.
- **RTP Port:** set RTP port for sending voice data.
- **Expire:** set expire time of registration. IAD will keep re-registering to proxy server before expire timed out

### 4.3.3 System Configuration

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

System Configuration	
Keypad DTMF Type:	<input checked="" type="radio"/> In-Band <input type="radio"/> RFC2833 <input type="radio"/> INFO
RFC2833 Payload Type:	<input type="text" value="96"/>
FAX Payload Type:	<input type="text" value="101"/>
Inter Digit Time:	<input type="text" value="3"/>
CallerID Type:	<input checked="" type="radio"/> disable <input type="radio"/> FSK(BELL CORE) <input type="radio"/> DTMF <input type="radio"/> NTT
Busy Forward:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
End of Dial Digit:	<input type="radio"/> NONE <input type="radio"/> * <input checked="" type="radio"/> #
<input type="button" value="OK"/>	

- **Keypad DTMF Type:** set DTMF type. User can select DTMF type IAD transmits.
- **RFC2833 Payload Type:** change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.
- **FAX Payload Type:** Change FAX payload type of IAD.
- **Inter Digit Time:** Set the DTMF inter digit time (second)  
To 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, IAD will dial out all number pressed (The inter digit time range is 1~10 seconds).
- **Caller ID Type:** Set Caller ID function. If user set disable, IAD won't display caller ID on Phone set when if receive caller ID information from remote site. User can also select caller ID type to be FSK, DTMF or NTT according on which type your phone set supports.
- **Busy Forward:** Set enable or disable to route the call to the next line (While the line number is not matched or engaged).
- **End of Dial Digit:** select end of dialing key, e.g. set end of dial key as \* button, after finished pressing dialing number then press \* will dial out.

### 4.3.4 Voice Configuration Screen

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

The screenshot displays the 'Voice Configuration' screen. On the left is a navigation menu with options: Network Interface, SIP Information, System Configuration, Voice Setting (highlighted), Phone Pattern, Support Function, Phone Book, Prefix Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The main area is titled 'Voice Setting' and contains the following configuration options:

Codec Priority	1st	2nd	3rd	4th	5th
	G.723.1	G.729a	G.711mu-Law	G.711A-Law	G.729
Frame Size	G.723.1 60ms	G.729a 40ms	G.729 40ms	G.711mu 40ms	G.711A 40ms
G.723 Silence Suppression:	<input type="radio"/> enable <input checked="" type="radio"/> disable				
Volume:	voice 28	input 28	DTMF 23		
Echo Cancelor:	<input checked="" type="radio"/> enable <input type="radio"/> disable				
Jitter Buffer:	Min. Delay 90		Max. Delay 150		
OK					

- **Codec Priority:** set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.
- **Frame Size:** User can set different packet size for each codec.
- **G.723 Silence Suppression:** Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1
- **Volume:** Adjust the volume in "Voice" (sending out); "Input" (receiving); "DTMF" (DTMF sending out).
- **Echo Cancelor:** Enable / Disable (suggested always Enable this function).
- **Jitter Buffer:** Set Min. Delay and Max. Delay of Jitter Buffer for voice packets.

**Note:**

Well the application before you change voice parameters, because this might cause incompatibility.

### 4.3.5 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 Pattern						
Ring Tone:	Frequency		On	Off		
	High(freq)	Low(freq)	High(lev)	Low(lev)	On	Off
Ring Back Tone:	480	440	155	155	2000	4000
Busy Tone:	620	480	155	155	500	500
Dial Tone:	400	0	155	0	8000	0
2nd Dial Tone:	440	350	19	19	25	25
OK						

- **Ring Tone:** Set Ring frequency, on time, off time, voltage level. IAD will give ring to phone set to trigger ring. If user found that phone set cannot ring when having incoming call, please try to increase ring frequency here.

- 🕒 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)

- **Ring Back Tone:** Set ring back tone parameters.
- **Busy Tone:** Set busy tone parameters.
- **Dial tone:** Set Dial tone parameters.

- 🕒 Low(freq) : Frequency value of Low frequency
- 🕒 High(freq) : Frequency value of High frequency
- 🕒 Low(lev) : Level (volume) of Low frequency
- 🕒 High(lev) : Level (volume) of High frequency

- 🕒 On1 : On cadence of first cycle
- 🕒 Off1 : Off cadence of first cycle
- 🕒 On2 : On cadence of second cycle
- 🕒 Off2 : Off cadence of second cycle

Note:

1. If disconnect tone is single-frequency, user has to configure the same frequency value of “Low frequency” and “High frequency”; the same level of “Low frequency” and “High frequency”
2. For On/Off cadence, user must set “1023” instead of “0”, if there is only one set of cycle, please as in second set columns

### 4.3.6 Support Configuration Screen

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

The screenshot shows the 'Support Function' configuration screen. On the left is a navigation menu with the following items: VoIP Gateway Configuration Menu (highlighted in green), Network Interface, SIP Information, System Configuration, Voice Setting, Phone Pattern, Support Function (highlighted in blue), Phone Book, Prefix Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The main area has a blue header 'Support Function'. Below it are two rows: the first row contains 'T.38 FAX:' with radio buttons for 'enable' and 'disable' (where 'disable' is selected); the second row contains 'FAX Redundancy Depth:' with a dropdown menu showing '0'. An 'OK' button is located at the bottom right of the configuration area.

- **T.38 FAX:** enable/disable FAX function. If user wants to fax with IAD gateway, this function must be enabled.
- **FAX Redundancy Depth:** Set the FAX redundancy.

### 4.3.7 Phone Book Configuration

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

**VoIP Gateway Configuration Menu**

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [Voice Setting](#)
- [Phone Pattern](#)
- [Support Function](#)
- [Phone Book](#)
- [Prefix Configuration](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Phone Book				
Index	Name	IP_Address	e164	Port

**New Record**

Index  Name  IP Address  E164 No.  Port No.

- **Add Data:** User can specify 10 sets of phone book via Web Management Interface. Please input index, Name, IP Address and E.164 number of the destination device.
- **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.

### 4.3.8 Prefix Configuration Screen

Click [Prefix configuration] in the navigation panel and open the [Prefix Drop/Insert Configuration] Screen.

**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).

### 4.3.9 DSCP Configuration Screen

Click **DSCP Configuration** in the navigation panel and open the **DSCP Configuration** Screen.

Set Signal or RTP Packet DSCP value:

- **Assured Forwarding (AF) PHB:** Select Delay priority and Drop Precedence
- **Expedited Forwarding (EF) PHB:** Select TOS value as EF
- **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)	(AF21)	(AF31)	(AF41)
	001010	010010	011010	100010

<b>Medium Drop Precedence</b>	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
<b>High Drop Precedence</b>	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

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

### 4.3.10 Password Configuration Screen

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

**It is highly recommended that you change the default password ([Null]).**

The screenshot shows the 'Password' configuration screen. On the left is a navigation menu with the following items: VoIP Gateway Configuration Menu (highlighted in green), Network Interface, SIP Information, System Configuration, Voice Setting, Phone Pattern, Support Function, Phone Book, Prefix Configuration, DSCP Configuration, Password (highlighted in blue), ROM Configuration, Flash Clean, Commit Data, and Reboot System. The main area has a blue header titled 'Password'. Below the header is a form with a dropdown menu on the left showing 'root'. To the right of the dropdown are three input fields labeled 'Current Password:', 'New Password:', and 'Confirm New Password:'. At the bottom of the form are two buttons: 'CHANGE' and 'ABORT'.

- **Change:** First select login name as root or administrator, then enter current password, new password and confirm new password again to set new password.
- **Abort:** Press abort will clean all inputs.

### 4.3.11 ROM Configuration Screen

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

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

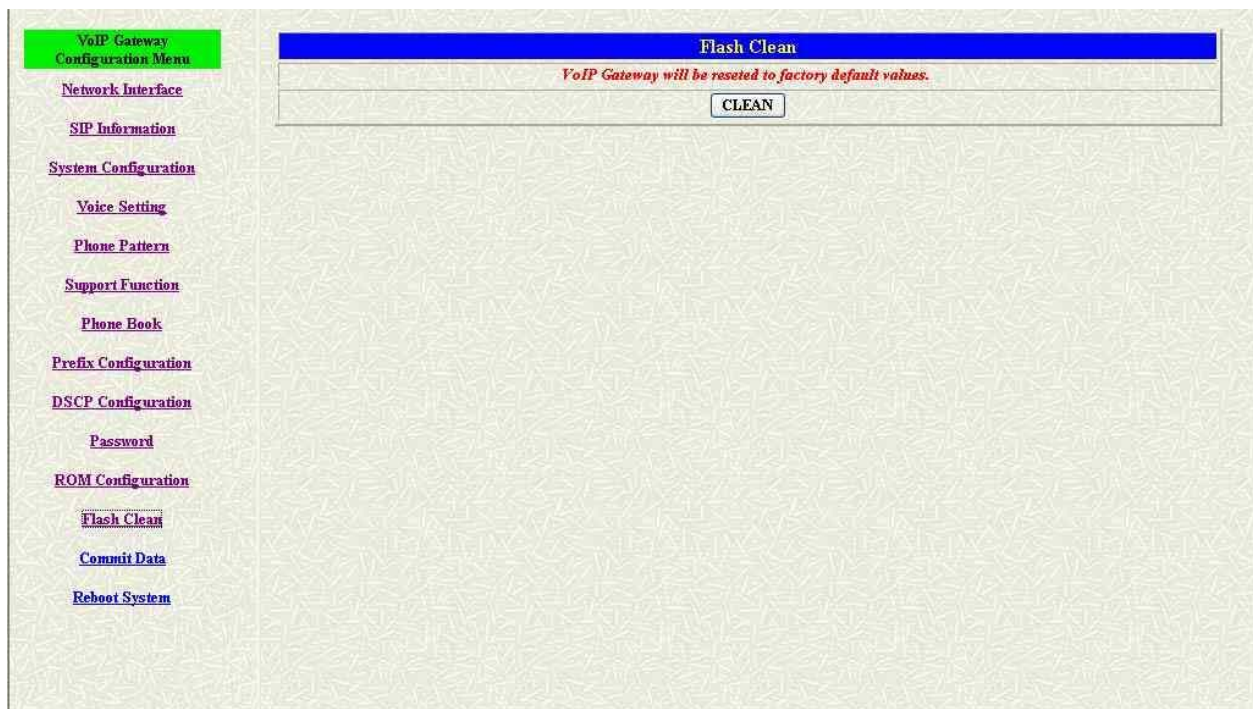
- **FTP/TFTP Server IP Address:** Set TFTP server IP address
- **Target File name:** Set file name prepared to upgrade
- **Method:** Select download method as TFTP or FTP
- **FTP Login:** Set FTP login name and password
- **Target File Type:** Select which sector of IAD to upgrade

**Note:**

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

### 4.3.12 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,

### 4.3.13 Commit Configuration Data Screen

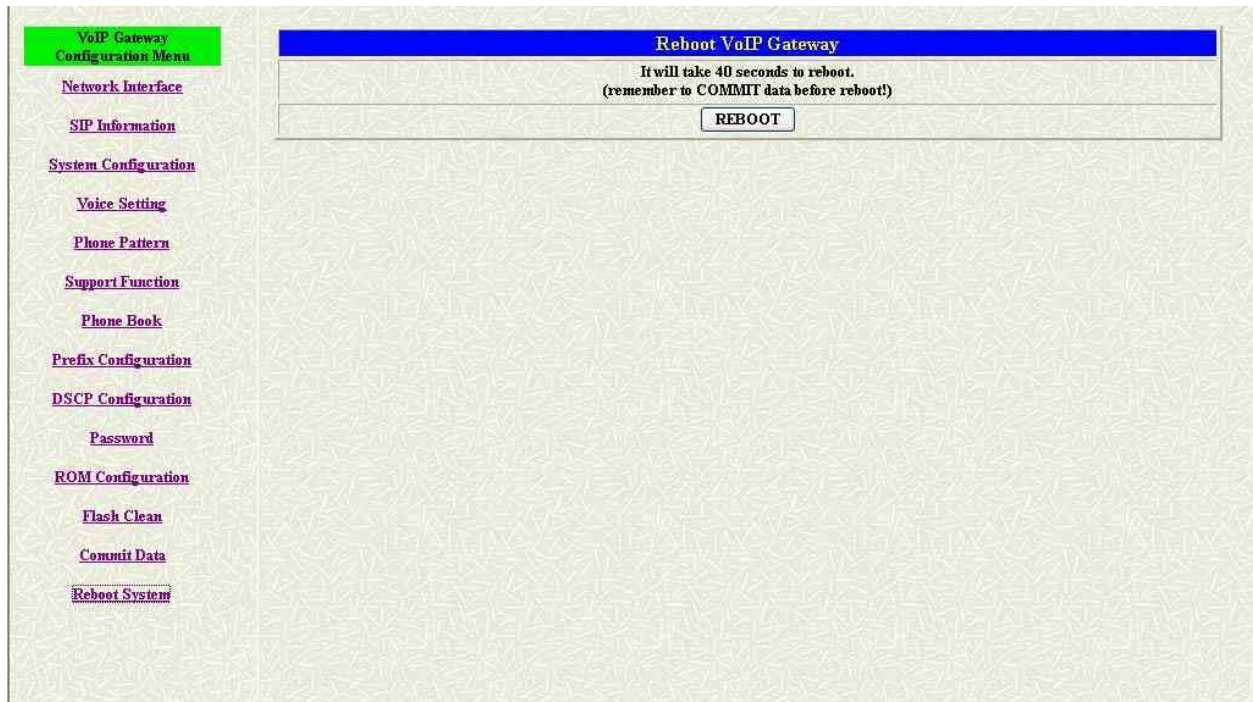
Click [Commit Data] in the navigation panel and open the [Commit Configuration Data] Screen.

The screenshot shows the 'Commit Data' screen. On the left is a navigation menu with the following items: VoIP Gateway Configuration Menu (highlighted in green), Network Interface, SIP Information, System Configuration, Voice Setting, Phone Pattern, Support Function, Phone Book, Prefix Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data (highlighted in red), and Reboot System. The main content area has a blue header bar labeled 'Commit Data'. Below the header, it says 'It will take few seconds...' and there is a 'COMMIT' button.

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

### 4.3.14 Reboot IAD System screen

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



- Press reboot will reset IAD.

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

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## Part V: Telnet Command Interface

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---

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	Input help/man/? to list all command list.
quit	Input quit/exit/close to exit telnet connection.
debug	Add debug flag and display debug messages.
reboot	Reboot local machine.
commit	Save all data in IAD.
ifaddr	Internet address manipulation.
time	Show current time.
ping	Test if an IP address is reachable.
pbook	Phone book information and configuration.
flash	Clean all configuration from flash rom.
sysconf	System information manipulation.
sip	Configure SIP related parameters.
security	This command is used to configure the account information included username and password obtained from the service provider.
voice	Voice information manipulation.
support	Special functions support manipulation.
tos	Set DSCP values for QOS.
phone	Setup of call progress tones and ring (SLIC control).
bureau	To set Hotline function which must be under Peer-to-Peer mode and switch to hotline service.
rom	ROM file update.
auth	Set configuration items for "administrator" user.
passwd	Password setting information and configuration.

Command	DESCRIPTION
prefix	Prefix drop/insert information manipulation
line	Line information manipulation

### 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 sip msg
```

```
debug -open
```

```
usr/config$
```

**Parameter Usages:**

-status: Display the enabled debug flags.

-add: Add debug flag.

-- sip: sip related information

-- msg: 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 sip, vp, msg.

```
usr/config$ debug -add sip msg
```

```
usr/config$ debug -open
```

**For example:**

```
usr/config$ debug -status
```

```
Current debug type enabled :
```

```
Debug Mode is open
```

```
DEBUG-> SIP MSG
```

```
usr/config$
```

**5.3.4 [reboot]**

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

```
usr/config$ reboot
```

```
Start to Unregister ...
```

```
Unregister complete...
```

```
. Rebooting...It will take 40 seconds....Attached TCP/IP interface to cpm unit 0
```

```
Attaching interface lo0...done
```

```
HTTPD initialized...
```

```
Flash Check
```

```
WorkMode : PROXY_MODE
```

```
Start registering to Proxy server
```

```

AC4804[0] is ok
AC4804[1] is ok
successful 2 2
Initialize OSS libraries...OK!
VP v1.42 stack open sucessfully.

login:

```

### 5.3.5 [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.

### 5.3.6 [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]
        [-nat used][-lanip IP Address][-sntp mode [server]]
        [-timezone GMT][-ipsharing used IP address][-autodns used]
        [-id PPPoE username][-pwd PPPoE password][-reboot used][-emsip
EMS IP]
        [-emsid EMS username][-emspwd EMS password][-emstime EMS digit]

```

```

-print      Display LAN information and configuration.
-mode       Specify WAN IP mode (0=Static, 1=DHCP, 2=PPPoE).
-ip         Specify WAN ip address.
-lanip      Specify LAN ip address.
-mask       Set Internet subnet mask.
-gate       Specify default gateway ip address
-nat        Set NAT service flag (On/Off).
-sntp       Set SNTP server mode and specify IP address.
-autodns    Specify the way to obtain DNS Server (0:Manual/1:Auto).
-dns        Specify IP address of DNS Server.
-timezone   Set local timezone.
-ipsharing  Specify usage of an IP sharing device and specify IP address.
-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 address.
-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).

SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).

**Example:**

```

ifaddr -mode 0
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -lanip 192.168.123.123
ifaddr -nat 1
ifaddr -sntp 1 210.59.163.254
ifaddr -ipsharing 1 210.59.163.254
ifaddr -autodns 1
ifaddr -dns 1 168.95.1.1
ifaddr -id 123456@hinet.net -pwd 123456 -reboot 1

```

usr/config\$

**Parameter Usages:**

-print: Print current IP setting and status

- mode: Set IP mode as static IP, DHCP or PPPoE.
- ip: Assigned IP address for Gateway
- 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)
```

- mask: Assigned internet subnet mask
- gate: Assigned IP default gateway
- nat: Provide DHCP Server and NAT function.
- 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 while 10.1.1.1 stands for SNTP server's IP address:
- autodns: Auto or manual configures the DNS IP address when gateway device is under DHCP and PPPoE mode.
- dns: Setup DNS Server IP Address.

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

- timezone: set local time zone according to GMT
- ipsharing: To enable or disable IP sharing function. When this function is enabled, user must specify a public fixed IP address.

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

**Note:**

If the public IP address is not a fixed one, IAD cannot work behind NAT with peer-to-peer mode.

- 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.

**For example:**

```
usr/config$ ifaddr -print

IP mode : Static
Internet address information
  LAN IP address      : 192.168.123.123
  WAN IP address      : 192.168.10.75
  Subnet mask         : 255.255.248.0
  Default gateway     : 192.168.8.254
  NAT enabled         : ON
  SNTP                : mode=1
                      server 168.95.195.12
                      time zone : GMT+8
                      cycle=1024 mins

  IPSharing           : no IPSharing device.

  Primary DNS Server   : 168.95.192.1
  Secondary DNS Server : 168.95.1.1
  PPPoE user name      : pppoe
  PPPoE password       : *****
  PPPoE reboot        : Yes
  PPPoE echo          : Enable

  EMS IP Address       : 192.168.1.1
  EMS User ID          : vwusr
  EMS Password         : *****
  EMS cycle time       : 0
usr/config$
```

### 5.3.7 [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$
```

### 5.3.8 [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 210.63.15.32
PING 210.63.15.32: 56 data bytes
64 bytes from 210.63.15.32: icmp_seq=0. time=5. ms
64 bytes from 210.63.15.32: icmp_seq=1. time=0. ms
64 bytes from 210.63.15.32: icmp_seq=2. time=0. ms
64 bytes from 210.63.15.32: icmp_seq=3. time=0. ms
----210.63.15.32 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$
```

### 5.3.9 [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

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
```

```
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port
numb
er]]
```

```
pbook [-delete index]
```

```
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port
numb
er]]
```

- print      Display phonebook data.
- add        Add an record to phonebook.
- search     Search an record in phonebook.
- delete     Delete an record from phonebook.
- insert     Insert an record to phonebook in specified position.
- modify     Modify an exist record.

**Note:**

If parameter 'end\_record' is omitted, only record 'start\_record' will be display.

If both parameters 'end\_record' and 'start\_record' are omitted, all records will be display.

Range of ip address setting (0.0.0.0 ~ 255.255.255.255).

Range of index setting value (1~100),

**Example:**

```
pbook -print 1 10
```

```
pbook -print 1
```

```
pbook -print
```

```
pbook -add name Test ip 210.59.163.202 e164 1001
```

```
pbook -insert 3 name Test ip 210.59.163.202 e164 1001
```

```
pbook -delete 3
```

```
pbook -search ip 192.168.4.99
```

```
pbook -modify 3 name Test ip 210.59.163.202 e164 1001
```

```
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
```

index	Name	IP	E164	Port
=====				
1	74	192.168.13.74	74	
-----				

```
usr/config$
```

### **5.3.10 [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:**

1. It is recommended to execute [flash -clean] after application firmware been upgraded.
2. Only User who login with root can execute it. After flash clean, all configurations in command [ifaddr] and [pppoe] will still be kept.

**For example:**

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

Flash clean start

Flash clean success!!

!! rebooting ...

Attached TCP/IP interface to cpm unit 0

Attaching interface lo0...done

HTTPD initialized...

Flash Check

```

WorkMode : PROXY_MODE
Start registering to Proxy server

AC4804[0] is ok
AC4804[1] is ok
successful 2 2
Initialize OSS libraries...OK!
VP v1.42 stack open successfully.

login:

```

### 5.3.11 [sysconf]

This command displays system information and configurations.

```

usr/config$ sysconf

System information and configuration
Usage:
sysconf [-print] [-idtime digit] [-bf digit] [-keypad dtmf]
        [-faxtype type][[-2833type type][[-lcdrop ON/OFF]
        [-droptime digit][-eod digit] [-callerid type]
        [-service used][[-dtmfstart digits] [-dtmfend digits]
sysconf -print

-print          Display system overall information and configuration.
-idtime         Inter-Digits time.(1~10 sec)
-service        Specify gateway service type. (0: Dial in service,
                1: HotLine service.)
-keypad         Select DTMF type: 0=In-band,
                1=RFC2833.
                2=INFO.
-faxtype         FAX    Payload    Type          (range:96~128
inter-used:100,102~105)
-2833type        RFC2833  Payload    Type          (range:96~128
inter-used:100,102~105)
-lcdrop          Disconnect Supervision(Loop Current Drop) (ON:1 / OFF:0)
-droptime        Period of Loop Current Drop (ms)
-eod             End of Dial Digit setting(0: none, 1: *, 2: #)
-callerid        Caller ID Type setting, 0: Disable,

```

```

1: FSK(BELLCORE),
2: DTMF,
3: NTT.

-dtmfstart      DTMF CallerID Start Symbol.
-dtmfend        DTMF CallerID End Symbol.
Example:
  sysconf -keypad 0 -eod 2 -callerid 1

usr/config$

```

**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).

-service: set SIP Phone to be normal mode or under hotline mode. (sysconf -service 0/1, 0 for normal service, 1 for hotline service.)

-keypad: DTMF replay type. When value is "1", Dynamix IAD -162 Gateway will transfer DTMF signal via RTP payload as defined in RFC2833. When the value is set to "0", the DTMF type is set as In-band, and 2 for SIP info DTMF.

-faxtype: FAX Payload Type. range: 96~128 inter-used:100,102~105.

-2833type: RFC2833 Payload Type. Range: 96~128 inter-used: 100, 102~105.

-lcdrop: Disconnect Supervision (Loop Current Drop) (ON:1 / OFF:0).

-droptime: Period of Loop Current Drop (ms).

-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.

-callerid: Support Bell Core, DTMF caller ID and NTT caller ID function. After the first ring at destination site, device will send line number as caller ID to called site.

-dtmfstart: DTMF Caller ID Start Symbol.

-dtmfstart: DTMF Caller ID End Symbol.

**For example:**

```

usr/config$ sysconf -print

System information
Gateway Service      : 0

```

```

Inter-Digits time           : 3
BusyForward                 : OFF
Keypad DTMF type           : In-band
End of Dial Digit          : #
Caller ID Type              : x
DTMF Caller ID Start Symbol : D
DTMF Caller ID End Symbol   : C
RFC2833 Payload Type       : 96
FAX Payload Type            : 101
Disconnect Supervision      : OFF
Loop Current Drop Time(ms) : 500

```

```
usr/config$
```

### 5.3.12 [sip]

This command is to configure SIP related parameters.

```
usr/config$ sip
```

SIP stack information and configuration

Usage:

```
sip [-print] [-mode pxmode] [-outpx IPaddress] [-transport type]
```

```
sip [-px address] [-px2 address] [-pxport number] [-outpxport number]
```

```
[-line1 number]
```

```
[-prefix prefixstring] [-expire t1] [-port udpPort] [-rtp rtpPort]
```

```
sip -print
```

```

-print      Display SIP stack information and configuration.
-mode       Configure as Peer-to-Peer mode:0/Proxy mode:1.
-px         Primary Proxy server address. (IPv4 address or dns name)
-px2        Secondary Proxy server address. (IPv4 address or dns name)
-pxport     Proxy server port.      (the port of proxy)
-outpx      OutBound Proxy server address. (IPv4 address or dns name)
-outpxport  OutBound Proxy server port. (the port of OutBound proxy)
-prefix     Specify prefix string, use it when UserID contains alphabets
            (if UserID uses numerals, specify as null)
-line1      TEL1 Phone number.
-pbsearch   Search phone book      0:off/1:on.
-expire     The relative time after which the message expires(0 ~ (2^31-1))
-port       SIP local UDP port number (5060~5070), Default: 5060
-rtp        RTP port number (2326~65534), Default: 16384

```

**Example:**

```

sip -mode 1
sip -px 210.59.163.171 -line1 70

```

```
usr/config$
```

**Parameter Usages:**

-mode: Configure as Proxy mode or Peer-to-Peer mode (0: Peer-to-Peer mode, 1: Proxy mode).

-px: to specify Proxy address when Dynamix IAD -162 Gateway is in proxy mode. Proxy address can be IPv4 address or DNS name.

-px2: to setting Secondary Proxy server address. Proxy address can be IPv4 address or DNS name.

-pxport: Set Proxy port for IAD to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.

-outpx: Set IP Address or URL address (Domain Name Server must be configured. Please refer to Network Configure) of outbound Proxy server.

-outpxport: Set outbound Proxy port for SIP-Phone to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.

-prefix: when your username contains alphabets, for example sip1123, then specify the prefix string as "sip".

-line1: assign line 1 number.

-pbsearch: enable/disable phone book search function under Proxy Mode. If user enabled this function, IAD will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, IAD will make call to related IP address.

-expire: this parameter is used to inform proxy server the valid duration of the registration information.

-port: SIP local UDP port which uses to listen incoming SIP Messages.

-rtp: Specify the RTP received port number.

**Note:** One will need to configure port and rtp parameters only when you deploy two or more sets behind the IP sharing device (Router).

**For example:**

```
usr/config$ sip -print
```

```
Run Mode                : PEER-2-PEER MODE
```

```
Prefix string           : null
```

```

Line1                : 1001
pbook search         : OFF
SIP listen port      : 5060
RTP receive port     : 16384
usr/config$

```

```
usr/config$ sip -print
```

```

Run Mode              : PROXY MODE
Primary Proxy address : 10.1.1.2
Secondary Proxy address : null
Proxy port            : 5060
OutBound Proxy address : null
Transport Type (TCP/UDP) : UDP
Prefix string         : null
Line1                 : 1001
pbook search          : OFF
SIP listen port       : 5060
RTP receive port      : 16384
Expire                : 3600
usr/config$

```

### 5.3.13 [security]

This command is used to configure the account information included username and password obtained from the service provider

```
usr/config$ security
```

Security information and configuration

Usage:

```

security [-line number][-name username] [-pwd password]
security [-print]

```

```

-print          Display system account information and configuration.
-line           Specify which line number you want to set the account.
-name           Specify user name.
-pwd            Specify password.

```

Example:

```
security -line 1 -name 1001 -pwd 1001
```

usr/config\$

**Parameter Usages:**

-print: print current setting in security command.

-line: Specify which line number you want to set into the account

**Note:** *If you have only one account, you can set into line1 or line2 using this parameter. For example, if you set the account into line1, line1 can accept incoming calls.*

-name: Specify the username of your account information.

-pwd: Specify the password of your account information.

**For example:**

usr/config\$ security -print

Line1 account information

Username : 1001

Password : \*\*\*

usr/config\$

**5.3.14 [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 [-send [G723 ms] [G711U ms] [G711A ms] [G729 ms] ]

[-volume [voice level] [input level] [dtmf level]]

[-nscng [G711U used1] [G711A used2] [G723 used3]]

[-echo used] [-mindelay t1] [-maxdelay t2]

voice -print

voice -priority [G723] [G711U] [G711A] [G729]

-print Display voice codec information and configuration.

-send Specify sending packet size.

G.723 (30/60 ms)

G.711U (20/40/60 ms)

G.711A (20/40/60 ms)

```

G.729 (20/40/60/80 ms)
-priority Priority preference of installed codecs.
G.723
G.711U
G.711A
G.729
-volume Specify the following levels:
voice volume (0~63, default: 25),
input gain (0~38, default: 25),
dtmf volume (0~31, default: 23),
-nscng No sound compression and CNG. (G.723.1 only, On=1, Off=0).
-echo Setting of echo canceller. (On=1, Off=0, 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 g711u 60 g711a 60 g729 60
voice -volume voice 20 input 32 dtmf 27
voice -echo 1 1 1 1

```

usr/config\$

### **Parameter Usages:**

-print: Print current voice information and configurations.

-send: To define packet size for each codec. 20/40/60/80 ms mean to send a voice packet per 20/40/60/80 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, 20/40/60/80 ms is applicable to G.729 codec, while 30/60ms is applicable to G.723.1 codec.

-priority: Codec priority while negotiating with other SIP device. The codec listed in left side has the highest priority when both parties determining final codec.

```

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.729A            : 40 ms
      G.723.1           : 60 ms
      G.711U            : 40 ms
      G.711A            : 40 ms
  Priority order codec  :
      g729a g7231 g711u g711a
  Volume levels         :
      voice volume      : 25
      input gain        : 25
      dtmf volume       : 23
  No sound compress & CNG :
      G.729A            : There is no setting
      G.723.1           : Off
      G.711(U-Law)      : Off
      G.711(A-Law)      : Off
  Echo canceller        : On On On On
  Jitter buffer         :
      Min Delay         : 90
      Max Delay         : 150

usr/config$
```

### 5.3.15 [support]

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

```
usr/config$ support
```

Special Voice function support manipulation

Usage:

```
support [-t38 enable]
```

```
        [-busy number] [-noanswer number] [-uncon number]
```

```
support -print
```

```
-t38      T.38(FAX) enabled/disabled.
```

```
-busy     Busy Forward number.(if empty, please fill "null")
```

```
-noanswer No Anser Forward number.(if empty, please fill "null")
```

```
-uncon    Unconditional Forward number.(if empty, please fill "null")
```

Example:

```
support -t38 1
```

```
support -busy 1001
```

```
support -uncon null
```

```
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.

-busy: Provide setting busy forwrd to other number, when you setting this function. Then this channel busy, auto forward to setting phone number.

-noanswer: Provide setting noanser forwrd to other number, when you setting this function. Then this channel not answer, auto forward to setting phone number.

-uncon: Provide setting noanser forwrd to other number, when you setting this function. Then all call this channel number, will all auto forward to setting phone number.

#### **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

Forward Numbers

Busy Forward number: 0123456789

NoAnswer Forward number: 0212345678

Uncondition Forward number:

usr/config\$

### 5.3.16 [tos]

This command is for setting Differentiated Service Code Point configuration.

usr/config\$ tos

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

Usage:

tos [-rtptype dscp]

tos [-sigtype dscp]

tos -print

[-rtpreliab mode]

tos -print

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\$

### 5.3.17 [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 [-ring [freq ] [ringON ] [ringOFF ] [ringLevel]]
      [-rbt [freqHi ] [freqLo ] [freqHiLev] [freqLoLev]
          [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
      [-bt [freqHi ] [freqLo ] [freqHiLev] [freqLoLev]
          [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
      [-dt [freqHi ] [freqLo ] [freqHiLev] [freqLoLev]
          [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
      [-flash [freqLo ] [freqHi ]]
      [-level [loopCurrentLevel] [onhookLineVoltageLevel ]]
phone [-print [ring]][rbt][bt][dt][flash]]
```

-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 .

-level Loop Current and On-Hook Line Voltage level 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)
level  loopCurrent  : 0 ~ 7     (20mA ~ 41mA, Step : 3mA)
level  OnHookVol    : 0 ~ 63    ( 0V ~ 94.5V, Step : 1.5V)

```

Example:

```

phone -print rbt
phone -ring 20 2000 4000 94
phone -rbt 480 440 125 105 2000 4000 2000 4000
phone -bt 620 480 125 105 500 500 500 500
phone -dt 440 350 96 96 8000 0 8000 0
phone -flash 400 800
phone -level 1 32

```

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

usr/config\$

usr/config\$ phone -print rbt

## Phone ring back tone paramter

Ringback Tone frequency high : 440  
Ringback Tone frequency low : 480  
Ringback Tone frequency high level : 13  
Ringback Tone frequency low level : 13  
Ringback Tone tone1 on : 100  
Ringback Tone tone1 off : 200  
Ringback Tone tone2 on : 100  
Ringback Tone tone2 off : 200

usr/config\$

usr/config\$ phone -print bt

## Phone busy tone paramter

Busy Tone frequency high : 620  
Busy Tone frequency low : 480  
Busy Tone frequency high level : 155  
Busy Tone frequency low level : 155  
Busy Tone tone1 on : 500  
Busy Tone tone1 off : 500  
Busy Tone tone2 on : 500  
Busy Tone tone2 off : 500

usr/config\$

```
usr/config$ phone -print dt
```

Phone dial tone paramter

```
Dial Tone frequency high      : 440
Dial Tone frequency low       : 350
Dial Tone frequency high level : 155
Dial Tone frequency low level  : 155
Dial Tone tone1 on            : 8000
Dial Tone tone1 off           : 0
Dial Tone tone2 on            : 8000
Dial Tone tone2 off           : 0
```

```
usr/config$
```

```
usr/config$
```

```
usr/config$ phone -print flash
```

Phone flash paramter

```
Flash frequency high : 800
Flash frequency low   : 400
```

```
usr/config$
```

### 5.3.18 [bureau]

To set Hotline function.

```
usr/config$ bureau
```

Bureau line setting information and configuration

Usage:

```
bureau [-hotline [Port DestIP TELnum]]
```

```
bureau -print
```

```
-print    Display Bureau line information and configuration.
-hotline   Set Hot line information. (Port range: 1~6)
```

Note:

```
Hotline feature should be used together with:
    $sysconf -service 1 (HotLine service)
```

Example:

```
bureau -hotline 1 192.168.4.69 628 2 192.168.4.200 999
```

```
usr/config$
```

### **Parameter Usages:**

-print: Display current Hotline table.

-hotline: Define Line1 Hotline table respectively. The table is included [Line number], [destination IP Address] and [destination Port or Number].

### **For example**

1. Destination is a FXS device, 628 is its Line1 number

```
usr/config$ bureau -hotline 1 200.168.4.69 628
```

User picks up the Line1, and then hears the ring back tone generated from destination. Of course, 628 are ringing simultaneously.

### **For example:**

```
usr/config$ bureau -print
```

Bureau line setting relate information

Hot line table

```
=====
Port      Destination Address      Remote TEL
-----
1          192.168.13.78            629
=====
```

```
usr/config$
```

### **5.3.19 [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.

- server: specify EMS Server IP address. Provide auto upgrade rom application verion, but you must use EMS Server it work.

**For example:**

```
usr/config$ rom -print

Download Method : TFTP
    Boot Rom    : sdboot.200
Application Rom  : 1asipIAD.107
    DSP App     : 48302ce3.140
    DSP Kernel  : 48302ck.140
    DSP Test Code : 483cbit.bin

    Hold Tone   : holdtone.101

usr/config$
```

**5.3.20 [auth]**

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

```
usr/config$ auth

Root control what command administrator can use.
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
```

Now the Administrator user can use the command which Root user assigned.

-print: Display the configurable items for “administrator” user.

**For example:**

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

Root can control what command administrator can use.

```
ifaddr    : Enable
sip        : Disable
line       : Disable
pbook     : Enable
support    : Disable
sysconf    : Disable
voice      : Disable
phone      : Disable
rtp        : Disable
tos        : Disable
prefix     : Disable
passwd     : Enable
rom        : Disable
flash      : Disable
bureau     : Enable
pppoe      : Enable
```

usr/config\$

### 5.3.21 [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 executed 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$
```

```
usr/config$ passwd -clean
```

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

```
Clean password OK.
```

```
usr/config$
```

**5.3.22 [prefix]**

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

```
usr/config$ prefix
```

```
Prefix drop/insert information and configuration
```

```
Usage:
```

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

```
prefix -delete index
```

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

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

prefix     The prefix of dialed number.  
 drop       Drop prefix(Enable:1/Disable:0).  
 insert     Insert digits.

Example:

```

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

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\$

### **5.3.23 [line]**

This command displays the line status.

```
usr/config$ line
```

---

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

```
Example:
```

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

***Parameter Usages:***

-print: display line status

***For example:***

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

```
Line information and configuration
```

```
Line  Reg. Status
```

```
=====
```

```
 1    No   Ready
```

```
usr/config$
```