2~24-Port H.323/SIP VoIP Gateway

VIP-281/480/880/1680/2480 series

User's manual

Copyright

Disclaimer

PLANET Technology does not warrant that the hardware will work properly in all environments and applications, and makes no warranty and representation, either implied or expressed, with respect to the quality, performance, merchantability, or fitness for a particular purpose.

PLANET has made every effort to ensure that this User's Manual is accurate; PLANET disclaims liability for any inaccuracies or omissions that may have occurred.

Information in this User's Manual is subject to change without notice and does not represent a commitment on the part of PLANET. PLANET assumes no responsibility for any inaccuracies that may be contained in this User's Manual. PLANET makes no commitment to update or keep current the information in this User's Manual, and reserves the right to make improvements to this User's Manual and/or to the products described in this User's Manual, at any time without notice.

If you find information in this manual that is incorrect, misleading, or incomplete, we would appreciate your comments and suggestions.

CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

Trademarks

The PLANET logo is a trademark of PLANET Technology. This documentation may refer to numerous hardware and software products by their trade names. In most, if not all cases, their respective companies claim these designations as trademarks or registered trademarks.

Revision

User's Manual for PLANET 2~24-Port H.323/SIP VoIP Gateway:

Model: VIP-281/VIP-480/VIP-1680/VIP-2480 series

Rev: 2.0 (January, 2007)

Part No. EM-VIP GW_V2

1

Chapter 1 Introduction

Overview

With years of Internet telephony and router manufacturing experience, PLANET proudly introduces the newest member of the PLANET VoIP gateway family: the **VIP-GW**, **VIP-281 / VIP-480 / VIP-880 / VIP-1680 / VIP-2480** series.

The PLANET VoIP Gateway is fully both SIP and H.323 standard compliant residential gateway that provides a total solution for integrating voice-data network and the Public Switched Telephone Network (PSTN), not only provides quality voice communications, but also offers secure, reliable Internet sharing capabilities for daily voice and Internet communications.

With advanced DSP processor (TI) and cutting edge VoIP technology, the PLANET VoIP Gateway is capable of handling both SIP and the H.323 calls. Up to 2/4/8/16/24 registrations to the SIP proxy or H.323 Gatekeeper, the VoIP Gateway are able to make calls to either H.323 or SIP voice communication environment. The VoIP Gateway is equipped with 1/4 LAN port Ethernet switch and built-in NAT router function that provides Internet access using only one IP address; with these features, users may now enjoy high quality voice calls and secure Internet access without interfering with routine activities.

Meanwhile, the PLANET VoIP Gateway is designed for comfort, ease-of-use with a sophisticated, and satisfaction from customers, VoIP Gateway not only inherits traditions of quality voice communications and real-time fax data over IP networks, but VoIP Gateway also eliminates the human resource VoIP network deployment. With optimized H.323/SIP architecture, PLANET VoIP Gateway is the ideal choices for P2P voice chat, ITSP cost-saving solution, but also provide network-converting feature to translate the packet network into traditional PBX system.

With built-in PPPoE/DHCP/DDNS clients, up to 2/4/8/16/24 concurrent connections in VoIP Gateway, voice communications can be established from anywhere around the world. PLANET VoIP Gateway comes with intuitive user-friendly, yet powerful management interface (web/telnet), that can dramatically reduce IT personnel resource, and complete VoIP deployment in a short time, plus remote management capability, VoIP administrators can monitor machine/network status, or proceed maintenance/trouble-shooting service via Internet browser or telnet session.
Besides, it provides voice channels status display and optimized packet voice streaming over managed and public (Internet) IP networks.

There are models for **VIP-281/VIP-480/VIP-880/VIP-1680/VIP-2480** and there are:

2-port model, VIP-28nxx:

VIP-281 equips one FXO and one FXS interfaces to have the great flexibility of PBX connection (FXO), and telephone or FAX machine connection (FXS).

VIP-281FS equips two FXS interfaces telephone set or FAX machine connections (FXS).

4-port model, VIP-48nxx:

VIP-480 equips two FXO and two FXS interfaces to have the great flexibility of PBX connection (FXO), and telephone or FAX machine connection (FXS).

VIP-480FS equips four FXS interfaces telephone set or FAX machine connections (FXS).

VIP-480FO equips four FXO interfaces to have the great flexibility of PBX connection (FXO).

8-port model, VIP-88nxx:

VIP-880 equips four FXO and four FXS interfaces to have the great flexibility of PBX connection (FXO), and telephone or FAX machine connection (FXS).

VIP-880FO equips eight FXO interfaces to have the great flexibility of PBX connection (FXO).

VIP-882 equips six FXS and two FXO interfaces to have the great flexibility of telephone or FAX machine connection (FXS), and PBX connection (FXO).

16-port model, VIP-168nxx:

VIP-1680 equips eight FXO and eight FXS interfaces to have the great flexibility of PBX connection (FXO), and telephone or FAX machine connection (FXS).

VIP-1680FO equips sixteen FXO interfaces to have the great flexibility of PBX connection (FXO).

24-port model, VIP-248nxx:

VIP-2480 equips twelve FXO and twelve FXS interfaces to have the great flexibility of PBX connection (FXO), and telephone or FAX machine connection (FXS).

VIP-2480FO equips twenty-four FXO interfaces to have the great flexibility of PBX connection (FXO).

In the following section, unless specified, **VIP-GW** will represent the family of products.

Network Feature

- **Network Address Translation (NAT):**

NAT allows multiple PCs to connect to an Internet Service Provider (ISP) using a single Internet access account.

- **Point-to-Point Protocol over Ethernet (PPPoE) Client Support:**

If you are a DSL user, the router has a built-in PPPoE client for establishing a DSL link connection with the ISP. There is no need to install a further PPPoE driver on your computers.

- **Smart QoS**

The smart QoS provide stable voice quality while user access internet from private LAN to internet at thesame time. This device would start suppressing throughput automatically when VoIP call proceed and keep full speed access when there is no VoIP traffic.

- **DDNS(Dynamic Domain Name Server)**

DDNS is a service that maps Internet domain names to IP addresses. It allows you to provide Internet users with a domain name (instead of an IP Address) to access your Virtual Servers.

- **Virtual Server**

Remote Users can access services such as the Web or FTP at your local site via public IP addresses can be automatically redirected to local servers configured with private IP addresses.

VoIP Functions

- H.323 / SIP dual mode communication
- SIP 2.0 (RFC3261), H.323v4 compliant
- Peer-to-Peer / H.323 GK / SIP proxy calls
- Voice codec support: G.711(A-law / μ-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)
- Voice processing: Voice Active Detection, DTMF detection, G.165/G.168 compliant echo canceller, silence detection, FAX (T.38 / T.30) Mode Option.
- Built in adaptive buffer that helps to smooth out the variations in delay (jitter) for voice traffic.
- Voice channels status display: This function display each port status likes as on-hook, off-hook, calling number called number, talk duration, codec.
- Life line support for co-existing FXO-FXS port of VIP-281, VIP-480, VIP-880, VIP-882, VIP-1680 and VIP-2480 while power down.

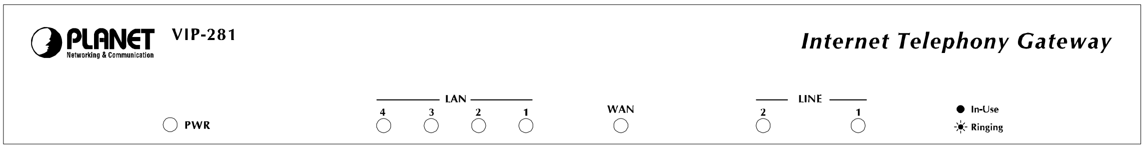
Package Content

The contents of your product should contain the following items:

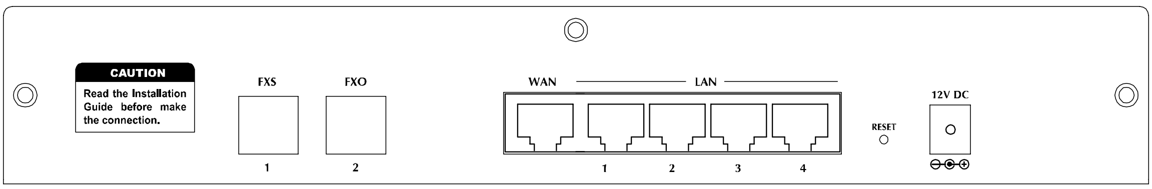
- The VoIP Gateway
- Power adapter (2 / 4 / 8-port model) / Power cord (16 / 24-port model)
- Quick Installation Guide
- User's Manual CD
- RJ-45 cable x 1
- RS-232 cable x 1 (8 / 16 / 24-port model)
- 25 port Telephone Cable x 1 (16 / 24-port model)
- Rack mount brackets x 2 (16 / 24-port model)

Physical Details

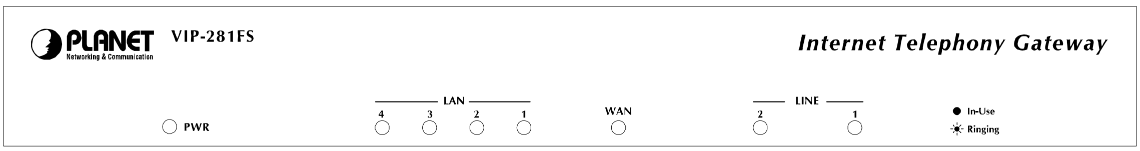
The following figure illustrates the front/rear panel of VIP-GW series:



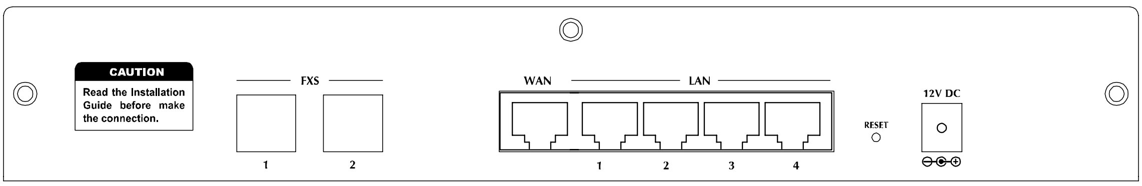
Front Panel of VIP-281



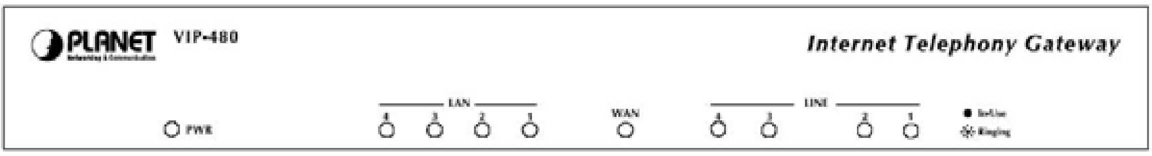
Rear Panel of VIP-281



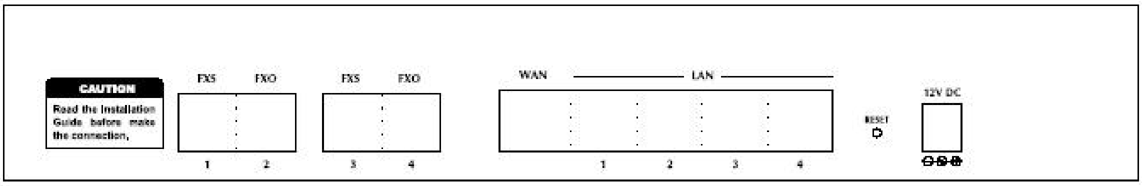
Front Panel of VIP-281FS



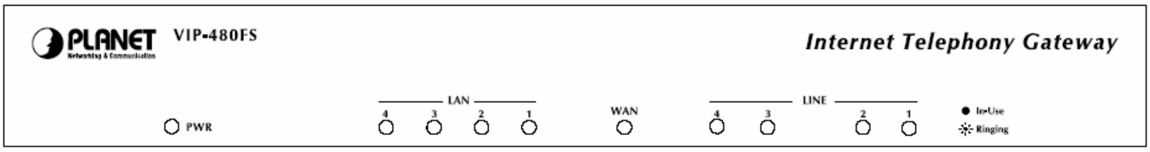
Rear Panel of VIP-281FS



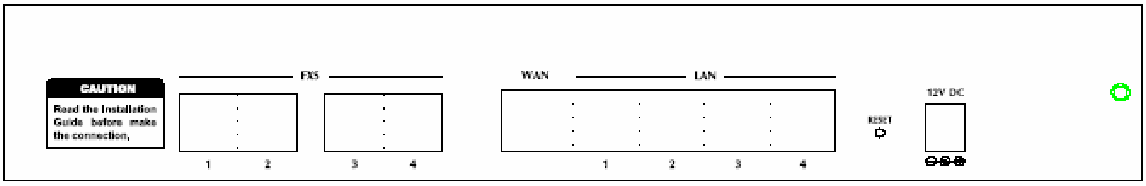
Front Panel of VIP-480



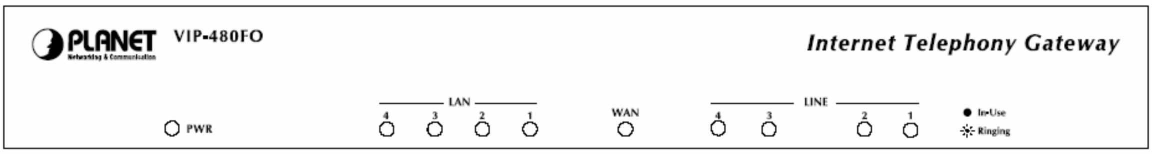
Rear Panel of VIP-480



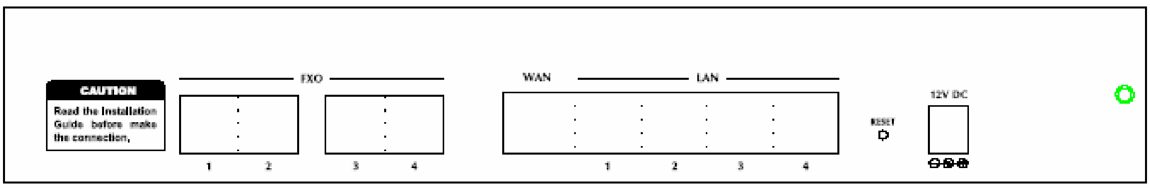
Front Panel of VIP-480FS



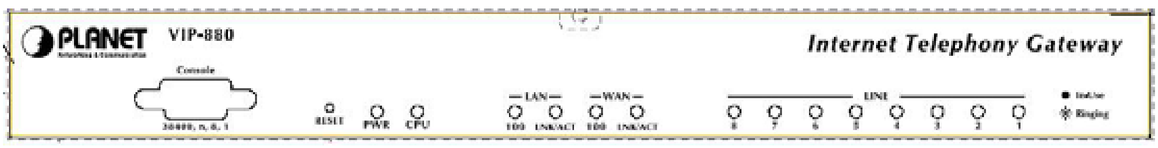
Rear Panel of VIP-480FS



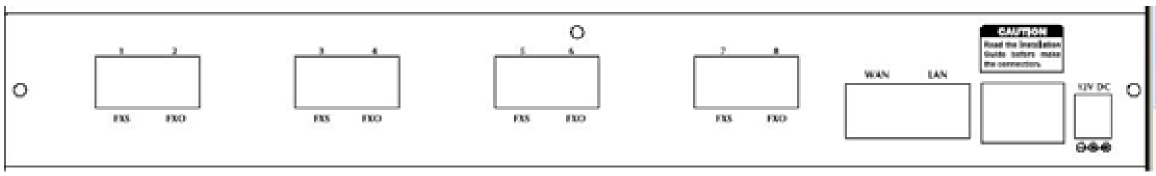
Front Panel of VIP-480FO



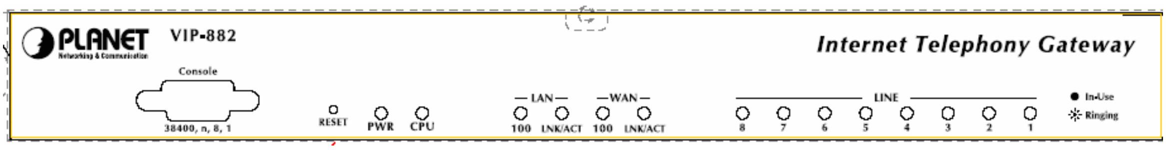
Rear Panel of VIP-480FO



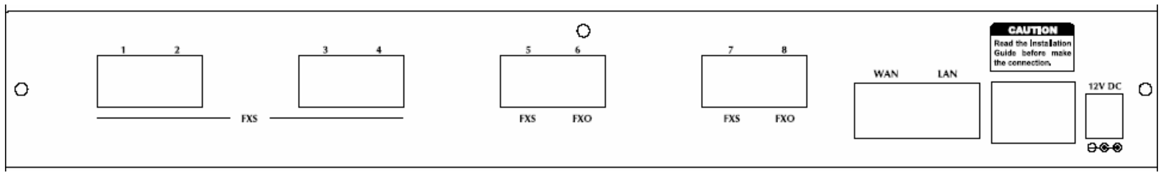
Front Panel of VIP-880



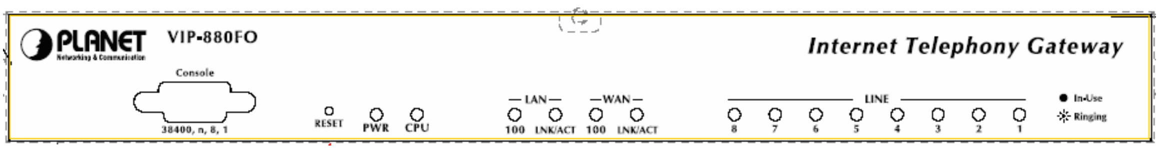
Rear Panel of VIP-880



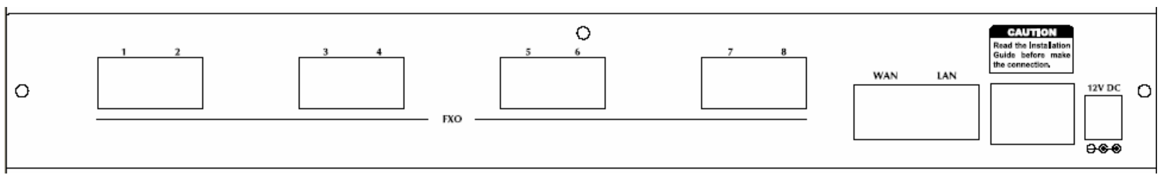
Front Panel of VIP-882



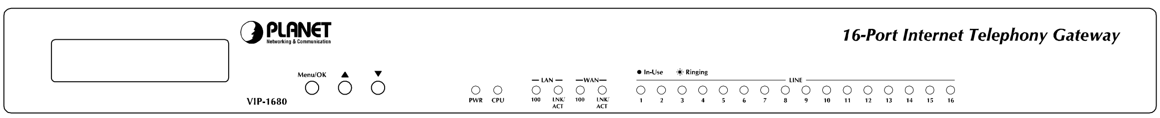
Rear Panel of VIP-882



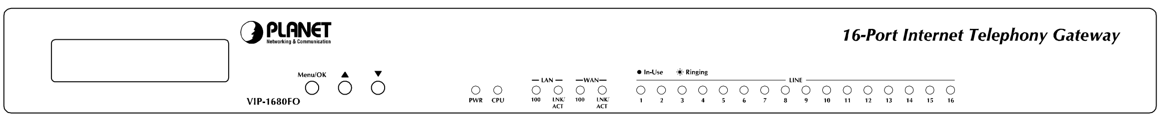
Front Panel of VIP-880FO



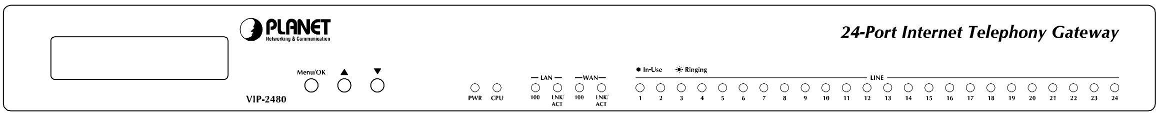
Rear Panel of VIP-880FO



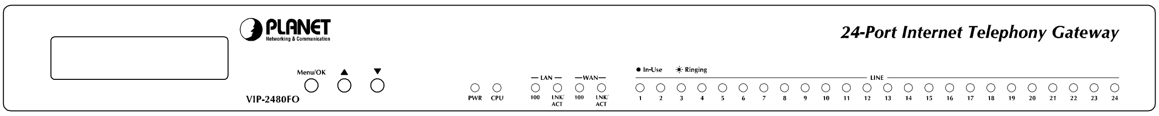
Front Panel of VIP-1680



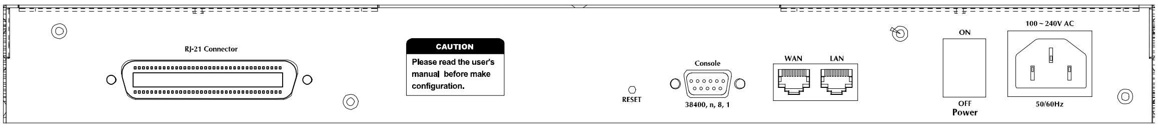
Front Panel of VIP-1680FO



Front Panel of VIP-2480



Front Panel of VIP-2480FO



Rear Panel of VIP-1680/VIP-1680FO/VIP-2480/VIP-2480FO

Front Panel LED Indicators & Rear Panels

Front Panel LED	State	Descriptions
PWR	On Off	GW is powered ON GW is powered Off
CPU (VIP-880 / VIP-1680 / VIP-2480 series)	Flashing	The system is running
WAN Port	ON Flashing Off	GW network connection established Data traffic on cable network Waiting for network connection
LAN Port	ON Flashing Off	LAN is connected successfully Data is transmitting Ethernet not connected to PC
FXS	ON Flashing Off	Telephone Set is Off-Hook Ring Indication Telephone Set is On-Hook
FXO	On Off	Line is busy Line is not enabled
LCD Panel (VIP-1680/ VIP-2840 series)	On Off	System is operation System is Shutdown

NOTE: System initialization will turn some LEDs ON for a few seconds.

Note

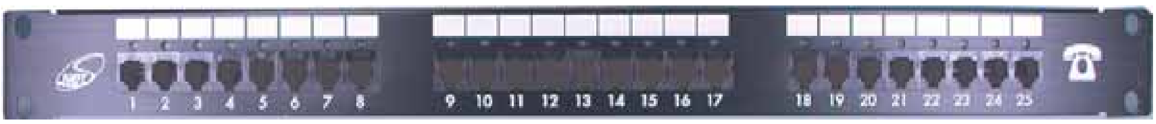
The Default LAN IP is http://192.168.0.1. Press RESET button on rear panel over 5 seconds will reset the VoIP Gateway to this default LAN/WAN IP address and Username/Password function.

Rear Panel	Descriptions
WAN	The WAN port supports auto negotiating Fast Ethernet 10/100Base-T networks. This port allows your voice gateway to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a CAT.5 twisted pair Ethernet cable.
LAN (VIP-880/VIP-1680/ VIP-2480 series)	The LAN port supports 1/4 10/100Base-T switch hub networks. These 1/4 ports allow your PC or Switch/Hub to be connected to the voice gateway through a CAT.5 twisted pair Ethernet cable.
LAN 1 ~ LAN 4 (VIP-281/VIP-480 series)	
Reset	The reset button, when pressed, resets the cable voice gateway without the need to unplug the power cord. **Note: (the VIP-880 series in Front Panel)**
Power	The supplied power adapter connects here.
FXS	FXS port was connected to your telephone sets or Trunk Line of PBX.
FXO	FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
Standard Telco 50 PIN Connector (RJ-21)	It is a 50 pins RJ-21 connector for connecting to telephone patch panel. **Note: (the VIP-1680/VIP-2480 series only)**
9-pin RS-232 (VIP-880/VIP-1680/ VIP-2840 series)	Connecting VIP to a terminal emulator for configuring VIP **Note: (the VIP-880 series in Front Panel)**

Connecting to the telephone patch panel (16-port/24-port model)

STEP 1: Attach the 25 port patch panel to the gateway through its RJ-21 connector.

STEP 2: The FXS LED indicators at telephone patch panel should be steady ON if the RJ-21 connector is well connected to the gateway and the gateway is powered on.



Patch Panel LED	State	Descriptions
FXS	ON Off	Telephone Set is On-Hook Telephone Set is Off-Hook
FXO	On Off	Line is not enabled Line is In-using

Note: The FXO interface is designed for connecting to PBXs (extension line) or central office switches (CO line), and the FXS interface is designed for connecting to analog telephone sets or fax machines. If the telephone cable connects to VIP-16/2480 series, the FXS interfaces are odd ports i.e. 1, 3, 5, 7, 9, 11, 13, 15, 17, 19, 21, 23, and the FXO interfaces are even ports, i.e. 2, 4, 6, 8, 10, 12, 14, 16, 18, 20, 22, 24.

Warning

Incorrectly connecting telephony devices to the RJ11 port on the Telephony Interface can cause permanent damage to the VoIP Gateway

Chapter 2

Preparations & Installation

2

Physical Installation Requirement

This chapter illustrates basic installation of VIP-GW series

- Network cables. Use standard 10/100Base-TX network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem (for WAN port usage)

Administration Interface

PLANET VIP-GW provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access:

To start VIP-GW web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Default LAN interface IP address of VIP-GW is **192.168.0.1**. You may now open your web browser, and insert **http://192.168.0.1** in the address bar of your web browser to logon VIP-GW web configuration page.

VIP-GW will prompt for logon username/password, please enter: **admin / 123** to continue machine administration.

Note

Please locate your PC in the same network segment (192.168.0.x) of VIP-GW. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

LAN/WAN Interface quick configurations

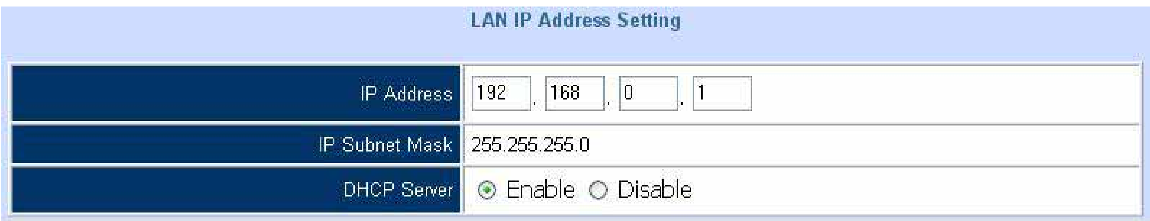
Nature of PLANET VIP-GW is an IP Sharing (NAT) device, it comes with two default IP addresses, and default LAN side IP address is "**192.168.0.1**", default WAN side IP address is "**172.16.0.1**". You may use any PC to connect to the LAN port of VIP-GW to start machine administration.

Hint

In general cases, the LAN IP address is the default gateway of LAN side workstations for Internet access, and the WAN IP of VIP-GW are the IP address for remote calling party to connect with.

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **192.168.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **admin** / **123**), browse to "**Advance Setup**" --> "**LAN setting**" configuration menu:



Parameter Description

IP address	LAN IP address of VIP-GW **Default: 192.168.0.1**
Subnet Mask	LAN IP address of VIP-GW **Default: 255.255.255.0**

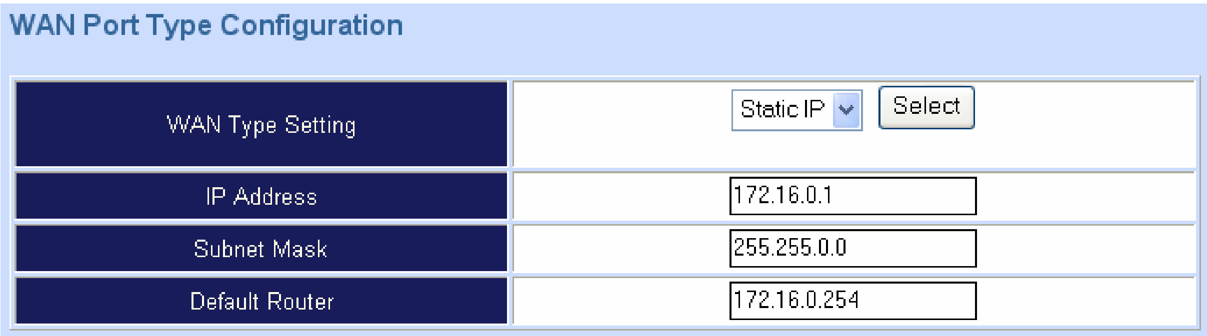
ⓘ Hint

```
It is suggested to keep the DHCP server related parameters
in default state to keep machine in best performance.
```

After confirming the modification you've done, Please click on the **Apply** button to macke the changes effective, and click **"Save Configuration"** to save configuration.

WAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **172.16.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **admin** / **123**), browse to "**WAN Setting**" configuration menu, you will see the configuration screen below:



Connection Type	Data required.
Static IP	The ISP will assign IP Address, and related information.
DHCP	Get WAN IP Address automatically; it is no need to configure the DHCP settings.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

ⓘ Hint

```
Please consult your ISP personnel to obtain proper PPPoE/IP
address related information, and input carefully.
If Internet connection cannot be established, please check
the physical connection or contact the ISP service staff
for support information.
```

3

Chapter 3

Network Service Configurations

Configuring and monitoring your VoIP Gateway from web browser

The VIP-GW integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Overview on the web interface of VoIP Gateway

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

To start VIP-GW web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Manipulation of VoIP Gateway via web browser

Log on VoIP Gateway via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input ***http://192.168.0.1*** to logon VoIP gateway web configuration page.

VoIP gateway will prompt for logon username/password: ***admin*** / ***123***

VIP-GW log in page

VIP-GW main page

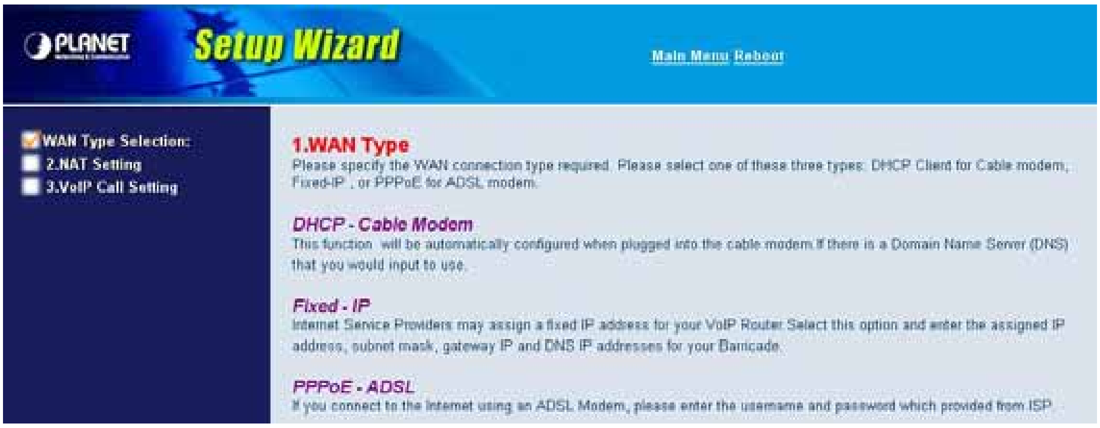
Wizard Setup for Quick Start

Wizard Setup

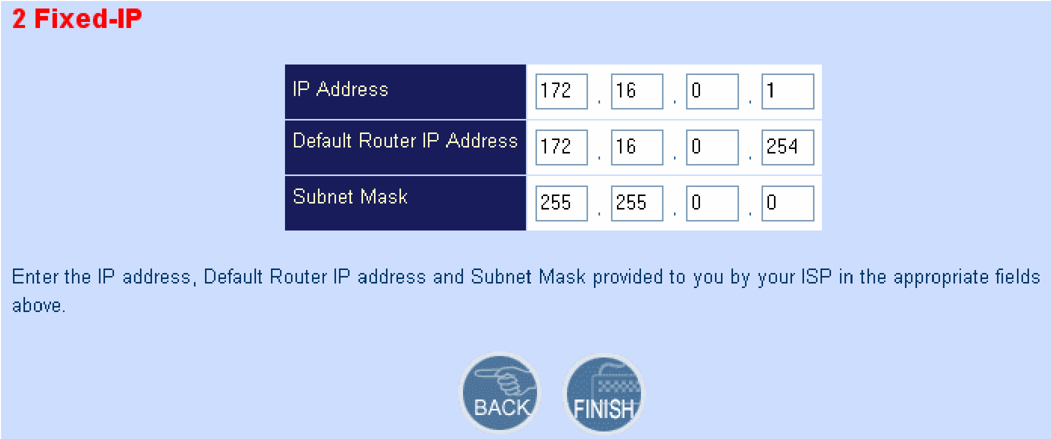
After finishing the authentication, the Main menu will display 3 parts of configuration, please click "**Wizard Setup**" to enter quick start:

1. WAN Port Type Setup (Setup First)

For most users, Internet access is the primary application. The VIP-GW support the WAN interface for Internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click "**WAN Port Type Setup**" from within the **Wizard Setup**, the following setup page will be show:

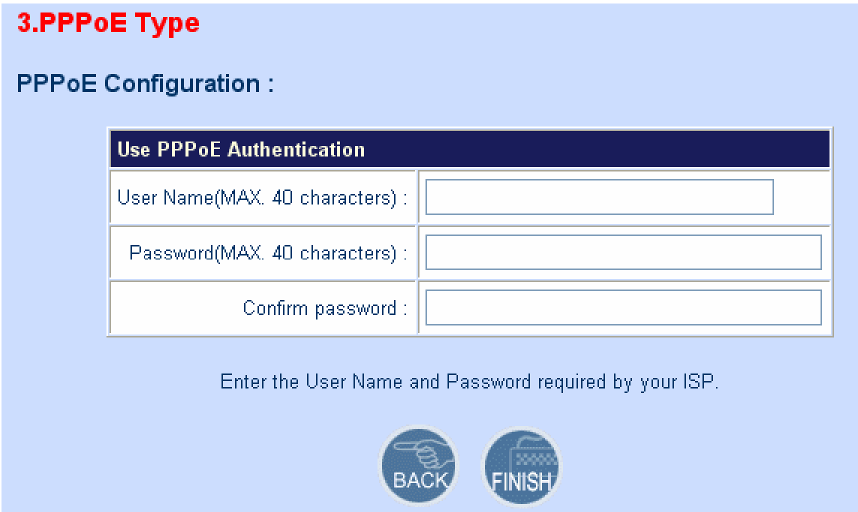


Three methods are available for Internet Access	
Fixed IP User	If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP.
IP Address	check with your ISP provider
Netmask	check with your ISP provider
Default Gateway	check with your ISP provider



ADSL Dial-Up User (PPPoE Enable)

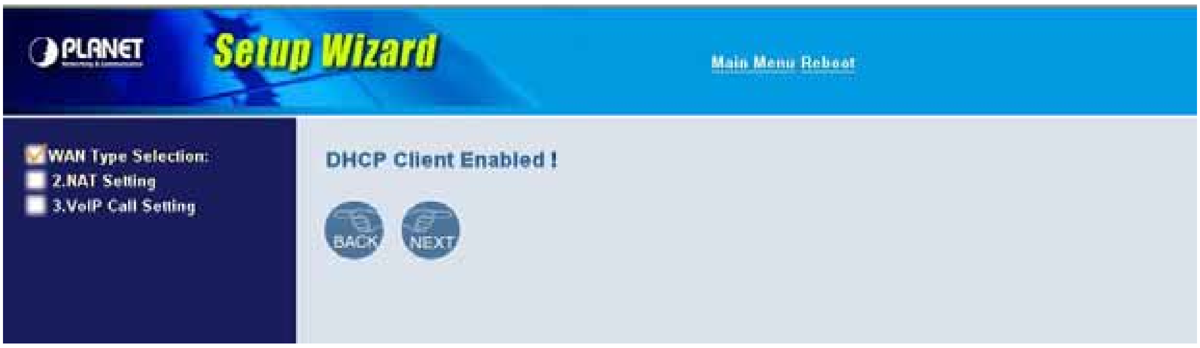
Some ISPs provide DSL-based service and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this item.



Three methods are available for Internet Access	
User Name	Enter User Name provided by your ISP
Password	Enter Password provided by your ISP
Retype Password	Enter Password to confirm again

DHCP Client (Dynamic IP): (Get WAN IP Address automatically)

IP Address: If you are connected to the Internet through a Cable modem line then a dynamic IP address will be assigned.



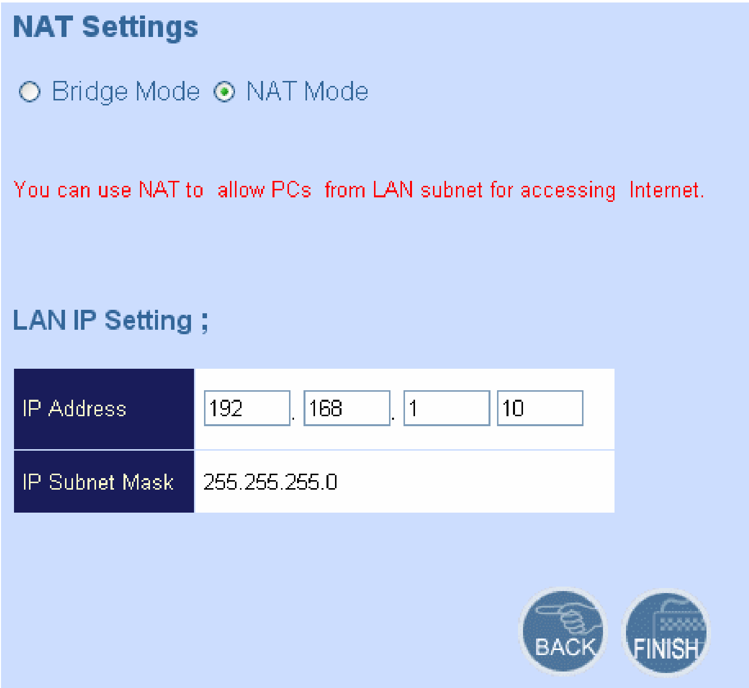
2. Configuring NAT or Bridge setting:

Bridge Mode:

When working on Bride Mode, the VoIP gateway will use only the LAN setting IP, The VIP-GW will use the same LAN IP setting as WAN IP. That means, when Bride mode enable, the WAN connection setting will be ignored.

NAT mode:

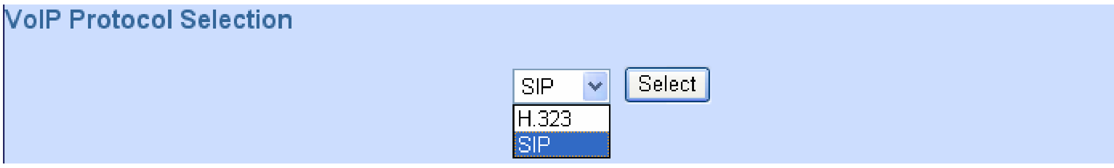
LAN IP Network Configuration	
IP Address	Private IP address for connecting to a local private network **(Default: 192.168.0.1)**
Subnet Mask	Netmask for the local private network **(Default: 255.255.255.0)**



3. VoIP Call Protocol Setup

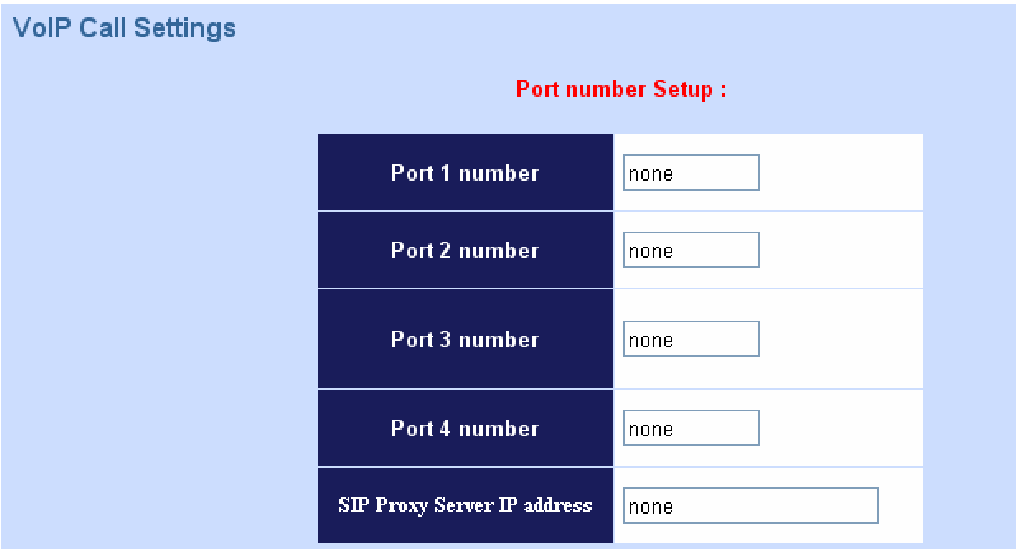
STEP 1 : Configure VoIP Call Signal Protocols :

User could select either H.323 or SIP Protocol, and click “**select**”



STEP 2 : configure the numbering with phone/line ports

Phone Number	The representation number is the phone number of the telephone that is connected to phone port
Line Number	Line ports are connected to the extension ports of the PBX system or the PSTN line. They have a common Line Hunting Group Number. When this number is dialed, the VIP-GW system will find a free FXO line connected to PBX. This hunting will skip all busy lines and absent lines and find only the idle line to the PBX. After the available line is found, you can hear the dial tone from PBX. After that, you can dial the needed phone number out through PBX



STEP 3: Let VIP-GW register to Gatekeeper/SIP Proxy Server

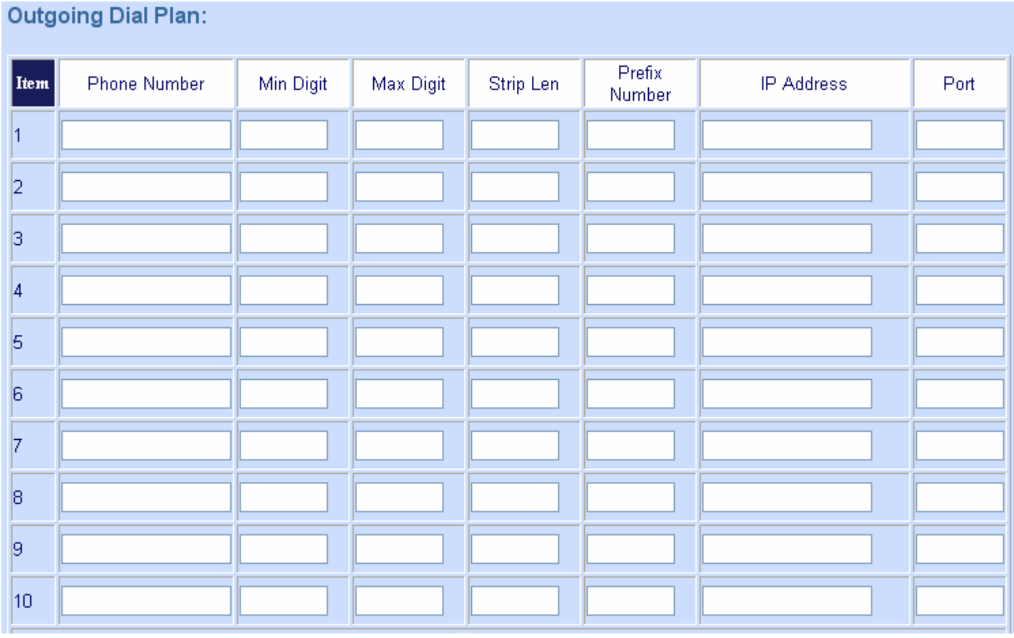
Note: If user does not have Gatekeeper/SIP proxy server, please go to STEP 4: Outgoing Dialing Plan

Gatekeeper IP address	There is a gatekeeper address fields. If this gateway does not want to register to any gatekeeper, just set value 0.0.0.0 to the primary Gatekeeper address.
SIP Proxy Server IP addresses	There is a SIP proxy server address fields. If this VIP-GW does not want to register to any SIP proxy server, just set value 0.0.0.0 to the sip proxy server address.

STEP 4: Outgoing Dialing Plan

The purpose of “Outgoing Direct Call” setting is to let user create a proprietary dialing plan when this Gateway is not registered to any H.323 Gatekeeper or any SIP proxy server. This setting can also assign some dialing plan to local ports (including prefix strip, prefix addition).

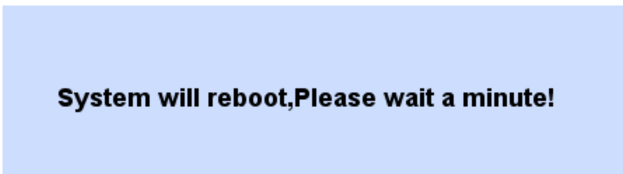
Through this setting, user can directly map a number to a specific gateway (IP address).



In the “Outgoing Dial Plan” settings: Maximum Entries : 50
“**Leading Number**” is the leading digits of the dialing number.
“**Min Length**” and “**Max Length**” is the min/max allowed length you can dial.
“**Strip Length**” is the number of digits that will be stripped from beginning of the dialed number.
“**Prefix Number**” is the digits that will be added to the beginning of the dialed number.
“**Destination**” is the IP address of the destination gateway that owns this phone number

STEP 5: Finishing the Wizard Setup

After completing the Wizard Setup, please click “**Finish**” bottom. The VIP-GW will save the configuration and rebooting gateway automatically. After 20 Seconds, you could re-login the machine.



4

Chapter 4

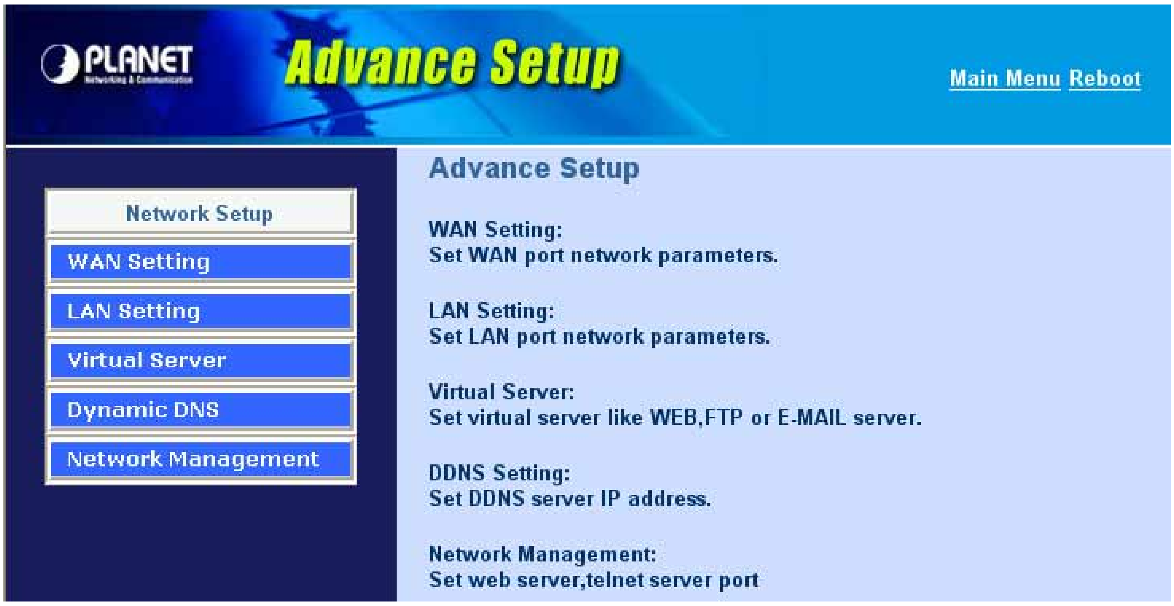
System Configurations

Advance Setup of Network Setup

In Advanced Setup, VIP-GW provides user two major parts function to configure:

One is “**Network Setup**”, the other one is “**VoIP Call Setup**”

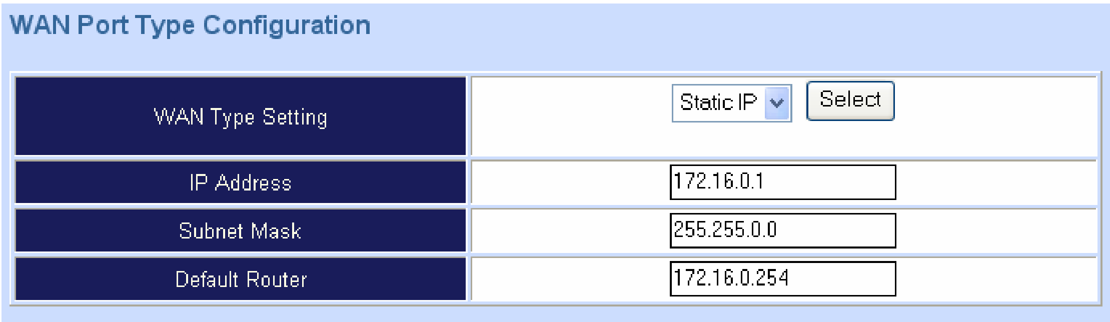
Network Setup Label	
WAN Setting	Sets/changes the WAN port type like “Fixed IP”, “DHCP Client” or ”PPPoE”.
LAN Setting	Modifies the IP address of the LAN port and setting DHCP server parameters.
Virtual Server	Remote user can access server such as Web or FTP at you local site vi。 pa public IP address can be automatically redirected to local servers configured with private IP address.
Dynamic DNS	Dynamitic DNS allows you to provide Internet users with a domain name to access your server.
Network Parameters	Network parameter allows you to modify the access port of gateway.



WAN Setting

For most users, Internet access is the primary application. The VIP-GW series support the WAN interface for Internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click "**WAN Setting**", the following setup page will be shown.Three methods are available for Internet access.

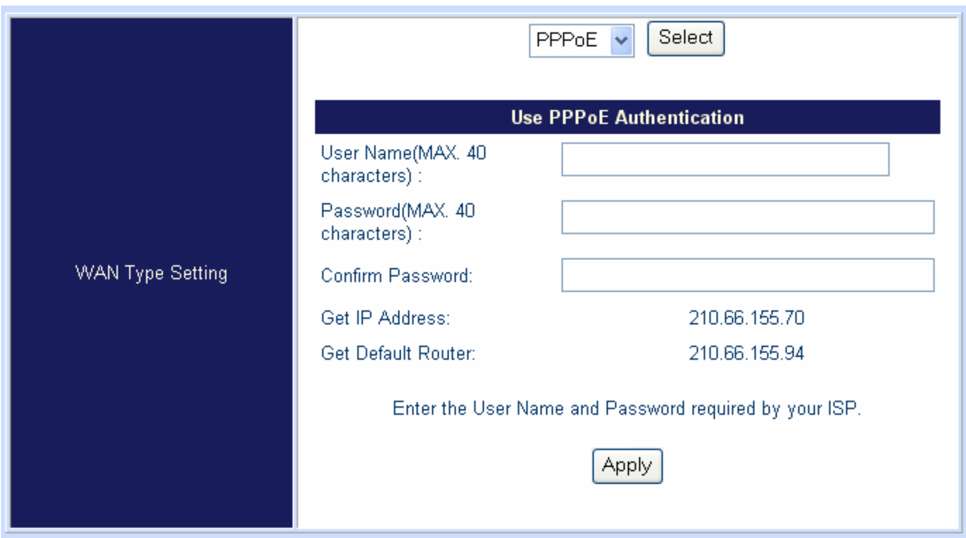
Static IP	You are a leased line user with a fixed IP address; fill out the following items with the information provided by your ISP
IP Address	Kindly please check with your ISP provider
Netmask	Kindly please check with your ISP provider
Default Gateway	Kindly please check with your ISP provider



PPPoE for ADSL

Some ISPs provide DSL-based service and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this item.

User Name	Enter User Name provided by your ISP
Password	Enter Password provided by your ISP
Retype Password	Enter Password to confirm again

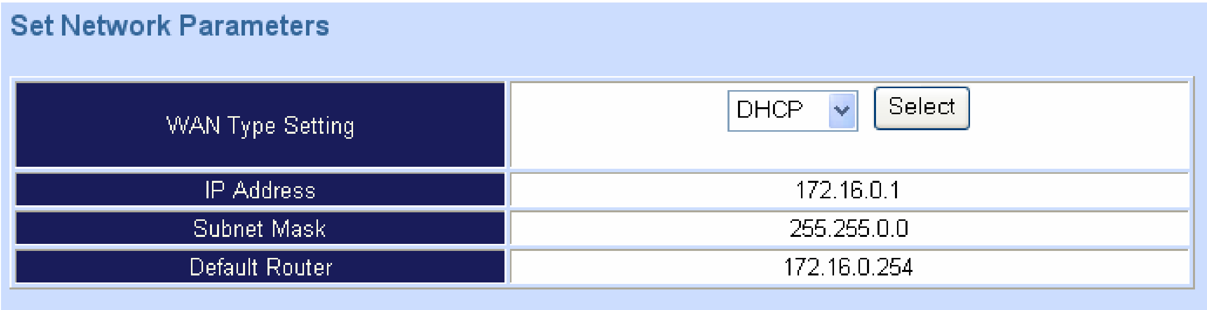


DHCP client (Dynamic IP): (get WAN IP address automatically)

IP Address: If you are connected to the Internet through a Cable modem line then a dynamic IP address will be assigned.

ⓘ Note

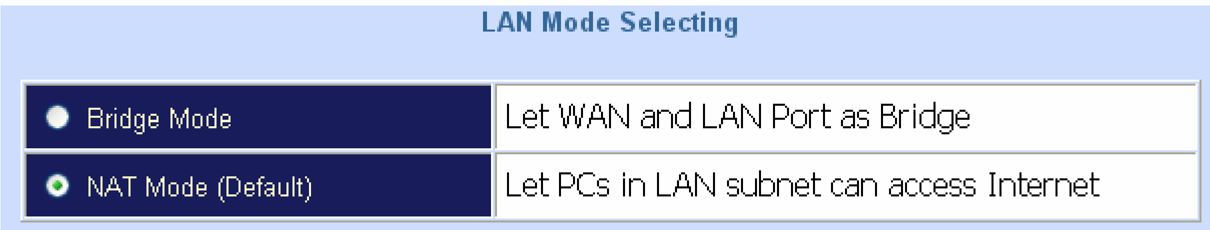
```
WAN port display the IP address, Subnet Mask and default
gateway IP address if DHCP client is successful
```



LAN Setting

There are two kinds of network feature to configure: **Bridge** Mode and **NAT** Mode:

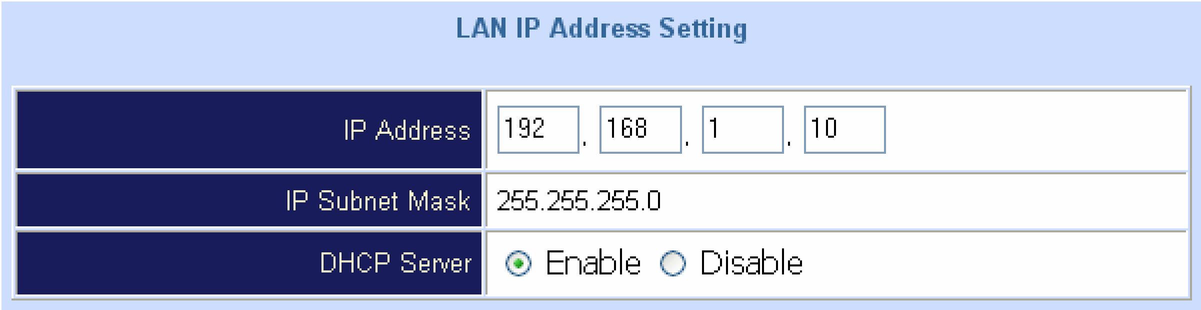
Bridge Mode	Select this VIP-Gw as Bridge. (WAN Port and LAN Port use the same IP address)
NAT Mode	Each of the VIP-GW has two Ethernet interfaces, one is for connecting to local network users, and the other is for connecting to an external broadband device (i.e. DSL modem/router or Cable modem). The LAN port is connected to the local Ethernet network. WAN is connected to the external broadband device. The LAN IP address/netmask is for private users or NAT users, and the WAN IP address/netmask is for public users.



LAN IP Network Configuration

IP Address: Private IP address for connecting to a local private network (**Default: 192.168.0.1**).

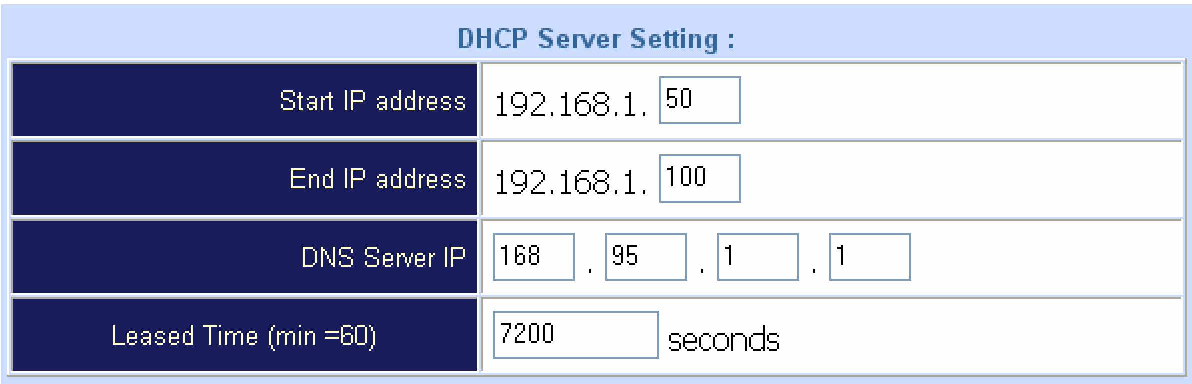
Subnet Mask: Netmask for the local private network (**Default: 255.255.255.0**).



DHCP Server Configuration

DHCP stands for Dynamic Host Configuration Protocol. It can automatically dispatch related IP settings to any local user configured as a DHCP client. The DHCP server supports up to 253 users (PCs) on **Yes:** Enables the DHCP server. **No:** Disables the DHCP server.

Start IP Address	Sets the start IP address of the IP address pool.
End IP Address	Sets the end of IP address in the IP address pool.
DNS Server IP Address	DNS stands for Domain Name System. Every Internet host. must have a unique IP address, also they may have a human friendly, easy to remember name such as www.yahoo.com. The DNS server converts the human friendly name into it's equivalent IP address.
Primary IP Address	Sets the IP address of the primary DNS server.
Secondary IP Address	Sets the IP address of the secondary DNS server.

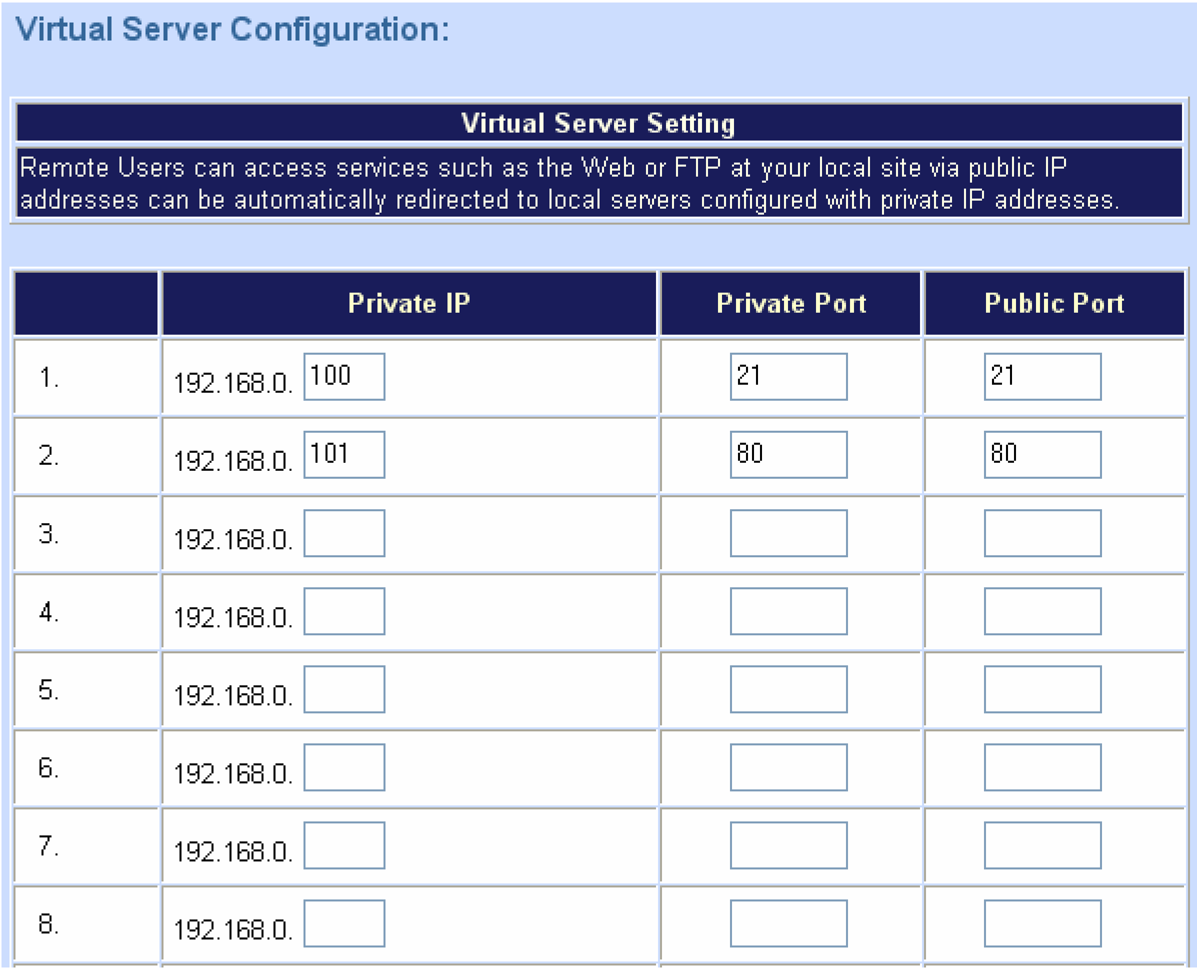


Virtual Server

"Natural firewall" allows requests for Internet access from the local network. However, any request from the Internet to the local network is blocked. By setting the Virtual Server function, computers outside the Intranet are allowed to access specific ports of local client. The Virtual Server Port Table may be used to expose internal servers to the public domain or open a specific port number to internal hosts. Internet hosts can use the WAN IP address to access internal network services, such as FTP, WWW, and Telnet etc.

How to set a Virtual Server

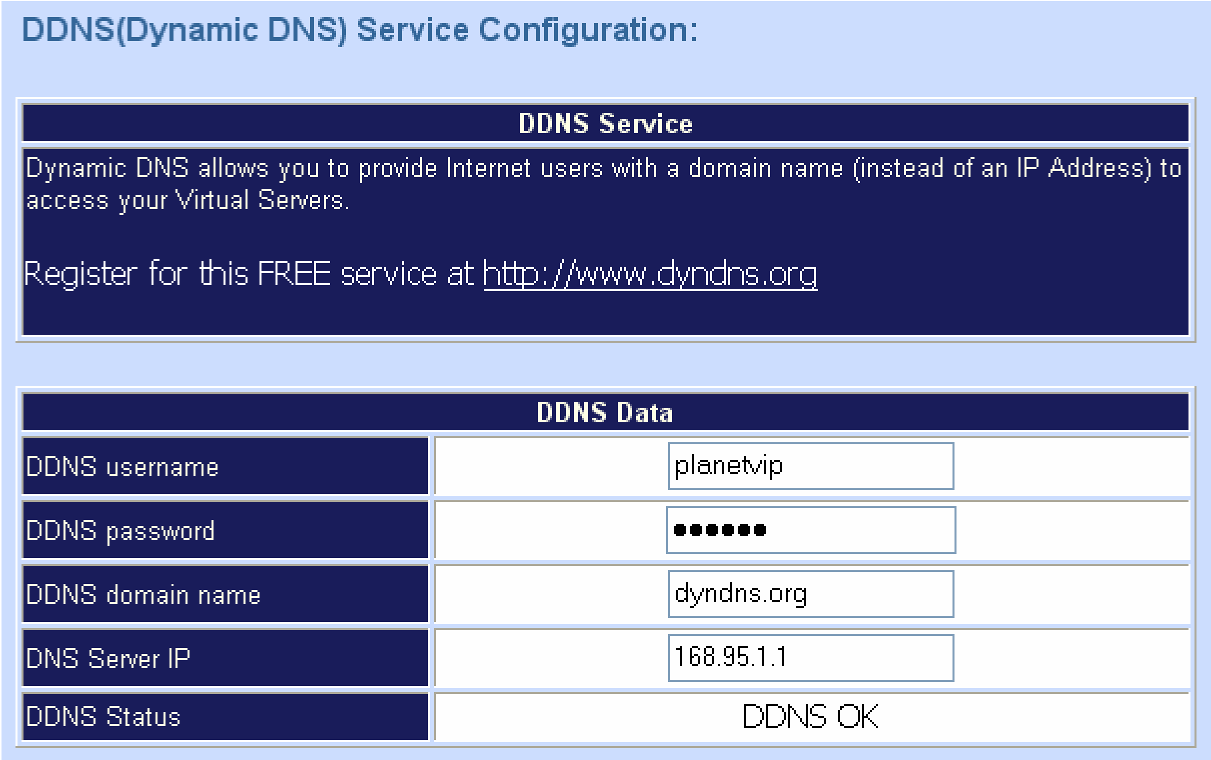
The following example shows how an internal FTP server is exposed to the public domain. The internal FTP server is running on the local host addressed as **192.168.0.100**.



Public Port	Specifies which port should be redirected to the internal host.
Private IP	Specifies the private IP address of the internal host offering the service.
Private Port	Specifies the private port number of the service offered by the internal host.
Apply	Click here to add the port-mapping entry and enable the service.

Dynamic DNS

DDNS is a service that maps Internet domain names to IP addresses. DDNS serves a similar purpose to DNS: DDNS allows anyone hosting a Web or FTP server to advertise a public name to prospective users. Unlike DNS that only works with static IP addresses, DDNS works with dynamic IP addresses, such as those assigned by an ISP or other DHCP server. DDNS is popular with home network, who typically receive dynamic, frequently-changing IP addresses from their service provider. To use DDNS, one simply signs up with a provider and installs network software on their host to monitor its IP address.

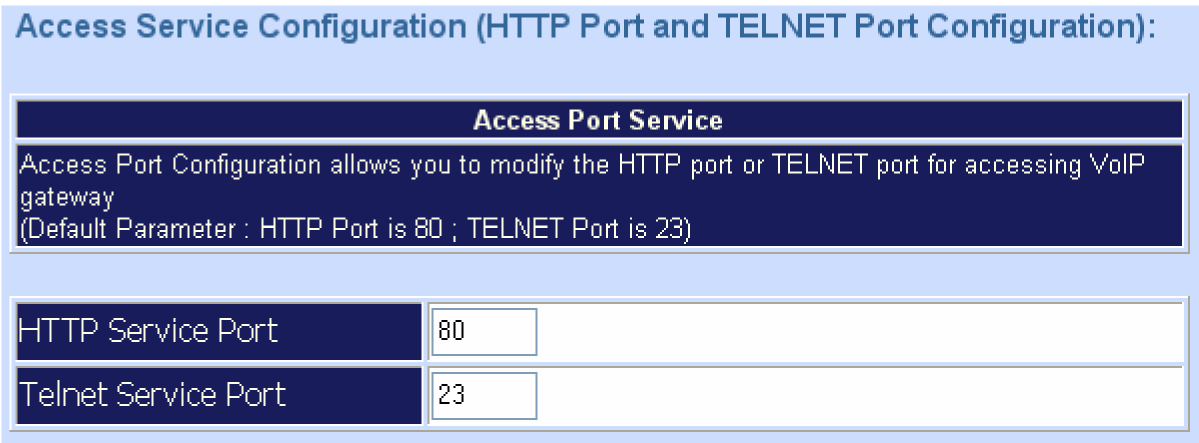


User Name	Input your DDNS User Name
Password	Input your DDNS Password
Domain Name	Input you set from your DDNS
DNS Server IP	Input your DNS Server IP

Netwrok Management

Network Parameter allows you to modify the access port of gateway.

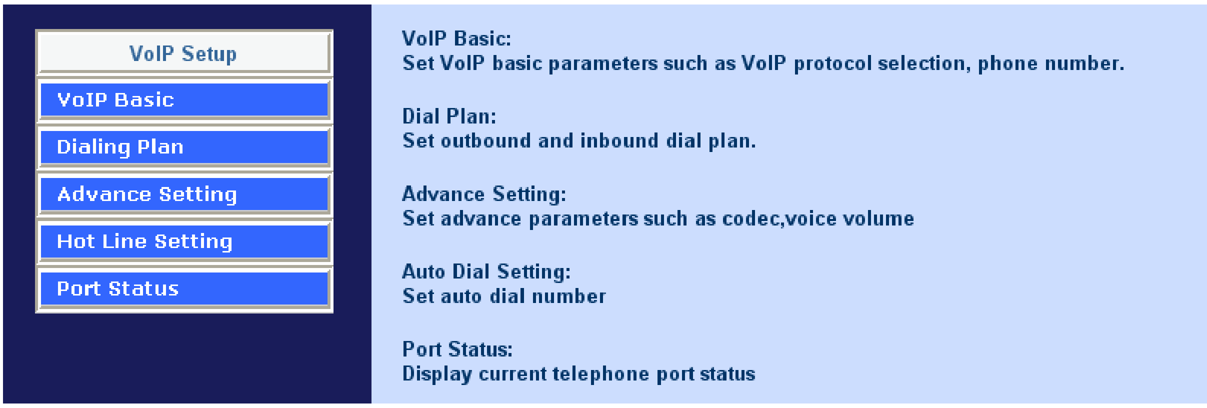
For example: Setting HTTP port: **80** and Setting TELNET port: **23**



Advance Setup of VoIP Setup

In Advanced Setup, VIP-GW provides user two major parts function to configure:
One is “**Network Setup**”, the other one is “**VoIP Call Setup**”

VoIP Setup Label	
VoIP Basic	The PLANET series gateway support 2~24 phone/line for SIP and H.323 VoIP call applications. You can configure these ports from this menu.
Dialing Plan	Users could apply any dial policy by setting Dial Plan including outgoing dial plan and incoming dial plan.
Advanced Setting	VIP-GW support for silence compression, DTMF Relay, Codec Selection, FAX mode Option, H323 Register Type and H.323 Fast-Start/Normal-Start function. FXO AC impedance, Volume Adjustment, RRQ TTL, RFC2833 Payload, IP TOS,..etc
Hot Line Setting	Let user can set up “hotline” to dial the phone number automatically.
Port Status	Display the telephone interface status



VoIP Basic Configuration to H.323 protocol

VoIP Basic Configuration: (Configure the VoIP protocol to **H.323** Protocol)

VoIP Basic Configuration

VoIP Protocol Setting H.323 [Select]

E.164 Number Setting (MAX 20 digit) :

Port 1 E.164 Number	none
Port 2 E.164 Number	none
Port 3 E.164 Number	none
Port 4 E.164 Number	none

Configure the numbering with FXS / FXO ports. (Depending on GW model number: if user uses the model number is VIP-1680, this VIP-1680 has 16 voice channels for setting, and the VIP-2480 had 24 voice channels for setting)

FXS Number	The representation number is the phone number of the telephone that is connected to FXS port.
FXO Number	FXO ports are connected to the extension ports of the **PBX system** or the **PSTN line**. They have a common Line Hunting Group Number. When this number is dialed, the VIP-GW system will find a free FXO line connected to PBX. This hunting will skip all busy lines and absent lines and find only the idle line to the PBX. After the available line is found, you can hear the dial tone from PBX. After that, you can dial the needed phone number out through PBX.

Configure the ANI (Answer Number Indication) / Caller ID of the FXS/FXO ports

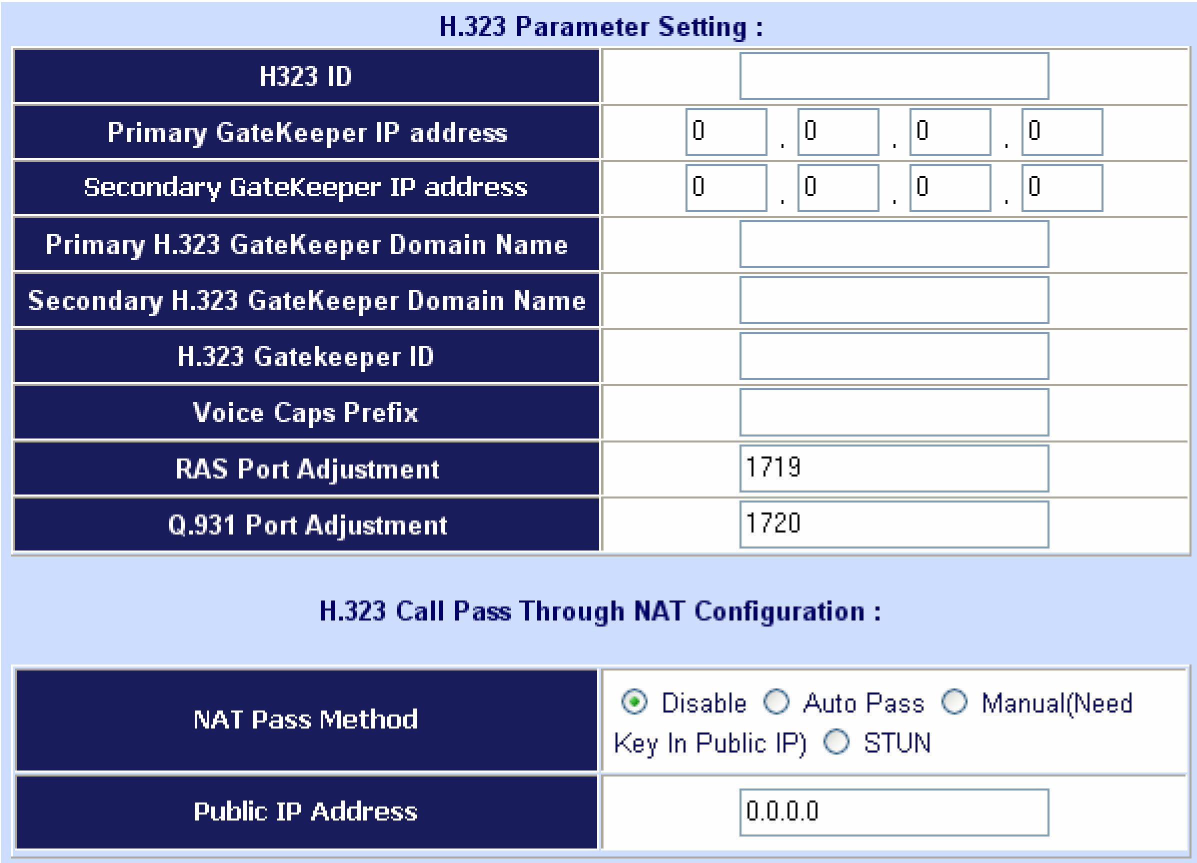
ITSP needs ANI for authorization when gateway calls Off-Net call to PSTN number or mobile phone number.

Caller ID / ANI Setting for Off-Net Call Setting (MAX 20 digit) :

Port 1 Caller ID / ANI	none
Port 2 Caller ID / ANI	none
Port 3 Caller ID / ANI	none
Port 4 Caller ID / ANI	none

Register to H.323 Gatekeeper server

Note: If user does not have Gatekeeper, please go to H.323 Dialing Plan Policy for more understandings.



H.323 Parameters Label	
H.323 ID	Sets the unique name of this Gateway, that is communicated as part of H.323 messaging.
Primary Gatekeeper IP Address **Secondary Gatekeeper IP Address**	There are two gatekeeper address fields, one is primary, the other secondary. If this gateway does not want to register to any gatekeeper, just set value 0 to the primary gatekeeper address. If the primary gatekeeper address is not 0, the gateway will register to the primary gatekeeper. If the second gatekeeper is not 0, the gateway will try to register to the second gatekeeper when failed to register to primary gatekeeper, i.e. if both the primary gatekeeper and second gatekeeper
Primary Gatekeeper Domain Name **Secondary Gatekeeper Domain Name**	Let user use Domain Name of H.323 Gatekeeper.
H.323 Gatekeeper ID	The Gatekeeper ID; usually do not need to set this field unless the gatekeeper must need this value.

Voice Cap Prefix	Let user set prefix number in RRQ nonstandard voicecap entry.
RAS Port Adjustment	In H.323 standard the RAS default port number is 1719. The VoIP gateway provides user to change RAS port number to meet the network environment.(Some area carrier blocks or forbidden the default port number)
Q.931 Port Adjustment	In H.323 standard the default Q.931 port number is 1720. The VoIP gateway provides user to change Q.931 port to meet the network environment. (Some area carrier blocks or forbidden the default port number)
H.323 Call Pass through NAT	
H.323 ID	Sets the unique name of this Gateway, that is communicated as part of H.323 messaging.
H.323 Pass Through NAT method	**1. Disable :** The Gateway operates in public IP address **2. Auto Detection:** When the Gateway register to GNU Gatekeeper, please select this option. **3. Manual Setting:** When the Gateway registers to H.323 Gatekeeper and operate under NAT (enable DMZ), please select this option and key in IP address.

Dialing Plan to H.323 protocol

The "**Dialing plan**" needs setting when the user uses the method of Peer-to-Peer H.323 VoIP call or registering H.323 Gatekeeper mode. The H.323 Dialing Plan has three kinds of directions: Outgoing (call out) and Incoming (call in) and PSTN route.

Outgoing Dial Plan	Peer-to-Peer call mode: Effective Registering to H.323 Gatekeeper mode: Effective
Incoming Dial Plan	Peer-to-Peer call mode: Effective Registering to H.323 Gatekeeper mode: The leading number would **register** to H.323 Gatekeeper
PSTN Route Dial Plan	Peer-to-Peer call mode: The same as the Incoming dial plan Registering to H.323 Gatekeeper mode: The leading number would **NOT register** to H.323 Gatekeeper

In the "Outgoing Dial Plan Configurations" settings: Maximum Entries : 50
"Outbound number" is the leading digits of the call out dialing number.
"Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.

"Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
"Add Digit Number" is the digits that will be added to the beginning of the dialed number.
"Destination IP Address / Domain Name" is the IP address / Domain Name of the destination gateway that owns this phone number.

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
		~				ADD

DELETE Outbound Dial Plan From To

Scenario description: Normally dial

001x leading call out, call to destination IP address: 172.16.0.100

002x leading call out, call to destination domain name: h323gw.test.com

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	001x	4 ~ 20	0	None	172.16.0.100	
2	002x	4 ~ 20	0	None	h323gw.test.com	
		~				ADD

DELETE Outbound Dial Plan From To

Scenario description: Speed dial

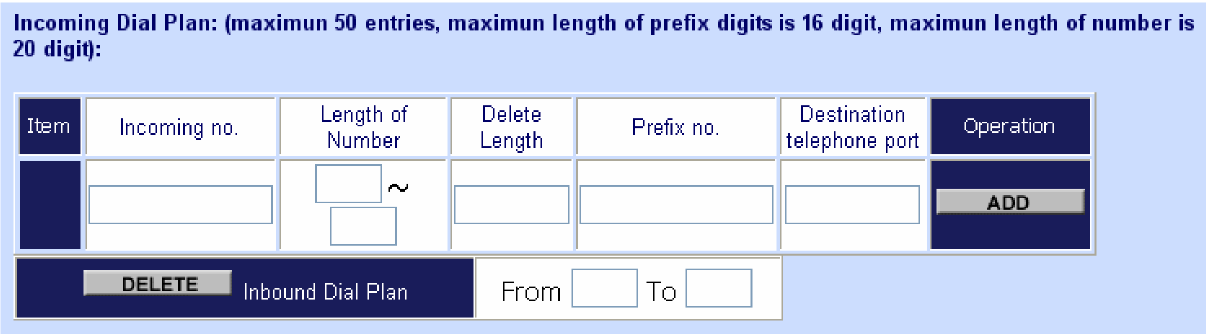
If user dials "101", the gateway automatically dials "1234567890" to destination IP address: 172.16.0.101

If user dials "202", the gateway automatically dials "0987654321" to destination IP address: 172.16.0.202

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	101	3 ~ 3	3	1234567890	172.16.0.101	
2	202	3 ~ 3	3	0987654321	172.16.0.202	
		~				ADD

DELETE Outbound Dial Plan From To

In the "Incoming Dial Plan Configurations" settings: Maximum Entries : 50
"Inbound number" is the leading digits of the dialing number.
"Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.
"Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
"Add Digit Number" is the digits that will be added to the beginning of the dialed number.
"Destination telephone port" is **"FXS/FXO port number"** ; this is for local dial plan setting phone number.



Scenario description: Hunting for FXS port (VIP-480FS)

Port 1: FXS

Port 2: FXS

Port 3: FXS

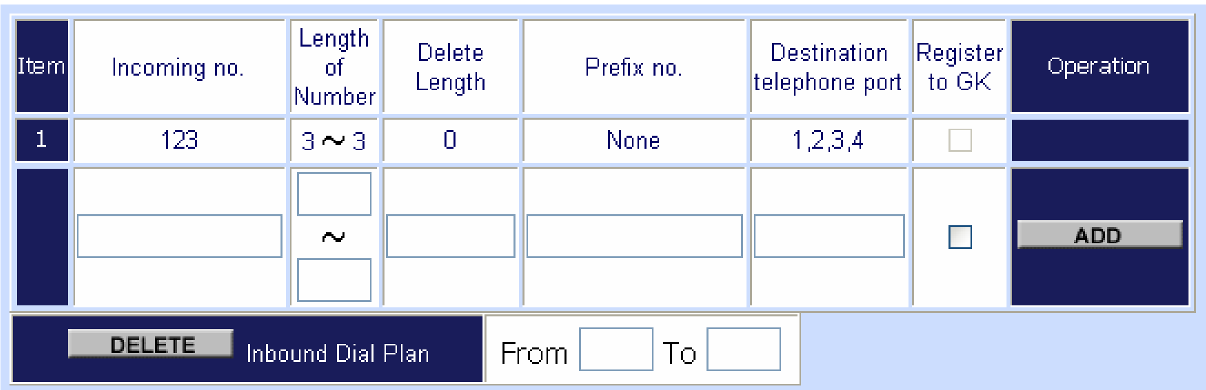
Port 4: FXS

H.323 number "123" call incoming, the port 1 will be ringing.

If port 1 is busy, the port 2 will be ringing.

If port 1 and port 2 are busy, the port 3 will be ringing.

If port 1, port 2 and port 3 are busy, the port 4 will be ringing.



Note: "123" will be register to H.323 Gatekeeper if "Register to GK" was enabled, show as below:

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	123	3 ~ 3	3	None	1,2,3,4	☑	
		~				☐	ADD

DELETE Inbound Dial Plan From To

Scenario description: Hunting for FXO port (VIP-480FO)

Port 1: FXO was connected to PSTN.

Port 2: FXO was connected to PSTN.

Port 3: FXO was connected to PSTN.

Port 4: FXO was connected to PSTN.

H.323 number “123” call incoming, the port 1 will be off-hook and hear the dial tone from PSTN.

If port 1 is busy, the port 2 will be will be off-hook and hear the dial tone from PSTN.

If port 1 and port 2 are busy, the port 3 will be off-hook and hear the dial tone from PSTN.

If port 1, the port 2 and port 3 are busy, the port 4 will be off-hook and hear the dial tone from PSTN.

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	123	3 ~ 3	0	None	1,2,3,4	☐	
		~				☐	ADD

DELETE Inbound Dial Plan From To

Note: “123” will be register to H.323 Gatekeeper if “Register to GK” was enabled, show as below:

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	123	3 ~ 3	3	None	1,2,3,4	☑	
		~				☐	ADD

DELETE Inbound Dial Plan From To

Scenario description: Termination call to FXO for one-shoot call

Port 1: FXO was connected to PSTN (area code is 81xxxxxxxx).

H.323 leading number “081x” incoming, and delete the first one digit “0”, and call to PSTN number.

Note: “081x” will be registered to H.323 Gatekeeper if “Register to GK” was enabled, show as below:

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	081x	4 ~ 20	1	None	1	☑	
		~				☐	ADD
DELETE Inbound Dial Plan	From	To					

Scenario description: Termination call to FXO

Port 1: FXS

Port 1: FXO was connected to PSTN (area code is 92xxxxxxxx).

Port 1 FXS call to “092x” to PSTN, the FXO port will delete the first one digit “0” and call to PSTN number.

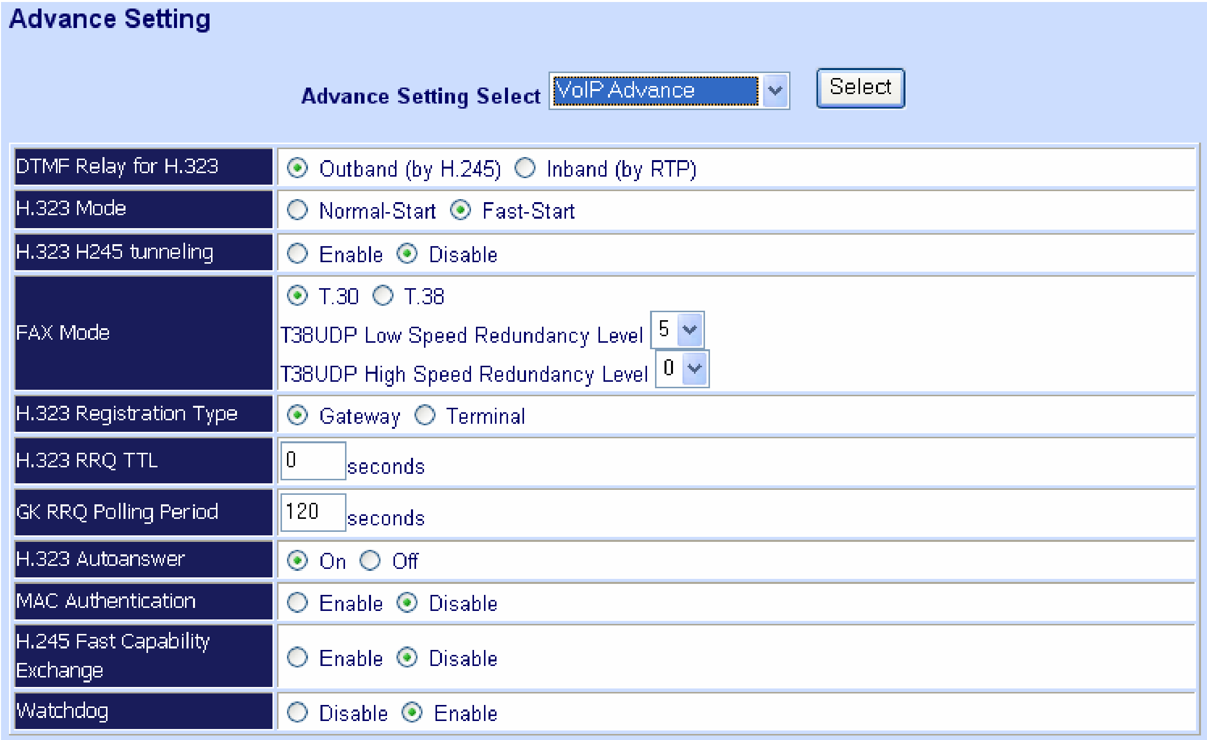
Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	092x	4 ~ 20	0	None	2	☐	
		~				☐	ADD
DELETE Inbound Dial Plan	From	To					

Note: “092x” will be NOT register to H.323 Gatekeeper when gateway when registering H.323 Gatekeeper mode

Advance Setting to H.323 protocol

In Advanced Setting , VIP-GW provides user three major parts function to configure:

One is “**VoIP Advance**”, the other one is “**Telephone Advance**” and “**Network Advance**”



H.323 VoIP Advance Configurtion	
Smart-QoS	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
DTMF Relay for H.323	After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are two methods of sending the DTMF tone. The first is “in band”, that is, sending the DTMF tone in the voice packet. The other is “out band”, that is, sending the DTMF tone as a signal. Sending DTMF tone as a signal could tolerate more packet loss caused by the network. If this selection is enabled, the DTMF tone will be sent as a signal.
H.323 Start Mode	This selection could force the gateway to use normal start mode (default mode) or fast start mode when establishing a VoIP call. Many other gateways only support normal start mode, enable this selection when it is necessary. The default is disabled (using fast start mode).
H.323 H.245 Tunneling	This selection could force the gateway to use normal start mode (default mode) or fast start mode when establishing a VoIP call. Many other gateways only support normal start mode, enable this selection when it is necessary. The default is disabled (using fast start mode).
FAX Mode Option	T.30/T.38 real-time FAX compliant Voice/FAX auto-switch.

H.323 RRQ TTL	The T.38 is a "Real Time Group 3 FAX communication over IP network" format. That's meaning it's a protocol for Fax over IP. You have to enable this function.
	This command configures the number of seconds that the gateway should be considered active by the H.323 Gatekeeper. The gateway transmits this value in the RRQ message to the gatekeeper.The default value is "0".
H.323 Registration type	There are 2 choices for this setting. "Gateway" means it will act as the VIP-GW. "Terminal" means it will act as the IP phone terminal.

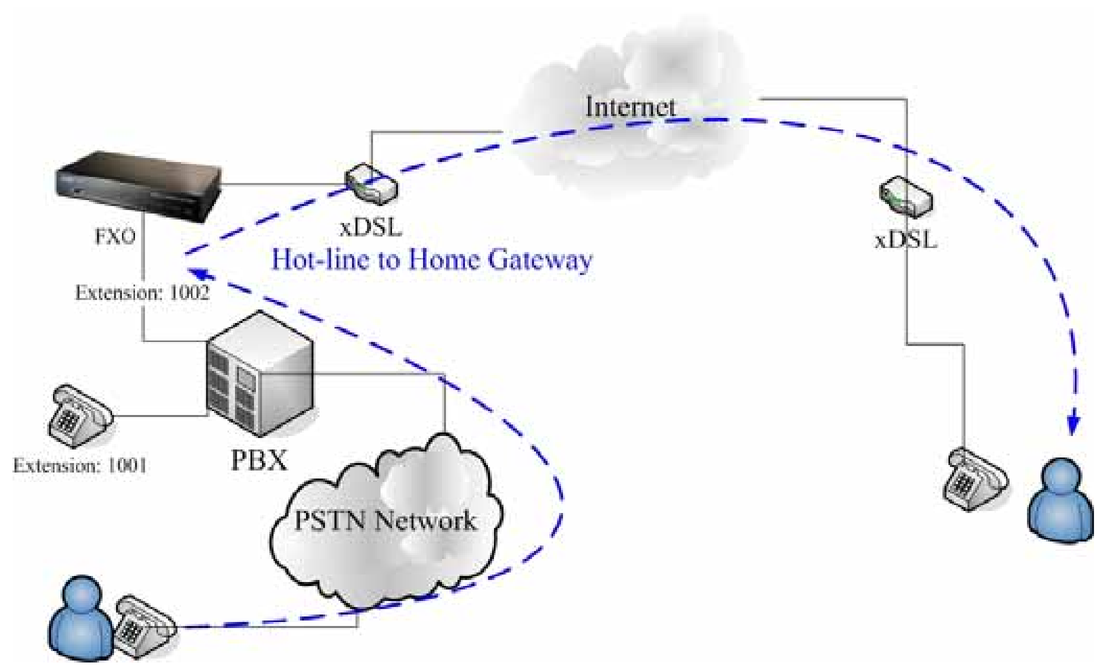
H.323 Telephone Advance Configuration	
Silence Compression	If this function is enabled, when silence is occurred for a period of time, no data will be sent across the network during this period in order to save bandwidth.
Dial Complete Tone	Disable / Enable dialing complete tone.
Voice Codec option	The codec is used to compress the voice signal into data packets. Each codec has different bandwidth requirement. There are four kinds of codec, **G.723, G.729AB, G.711_u** and **G.711_A**. The default value is **G.723**.
FXS Impedance	The FXS provides 600/900 OHM impedances for selection.
FXO AC Impedance	The FXO provides wild and complex ac termination impedances for selection.
Phone (Line) in/out volume	You can adjust the Phone (Line) in/out volume, range from -9db to 9db.
FXO Tx/Rx Gain	You can adjust the FXO Tx/Rx Gain, range from -6db to 6db.
UK PSTN release tone detection	When you use the Gateway to UK, you can Enable this selection to detection release tone.

Scenario description: Flash detection and generation duration

1. PSTN Call from PSTN to Office PBX and dial the extension 102 go to gateway.
2. Call to gateway of oome by Hotline.
3. Home user needs call transfer to extension number 101.
4. Dial flash and gateway FXS detect and generate the flash to PBX in office.

Flash Detection: Let you change flash detection (milliseconds) of gateway when phone generate flash to FXS.

Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.



Ring Frequency	You can configure how long the Ring Frequency do you want to use.
FXO Battery Reverse	Enable battery reverse to detect polarity from PSTN line. The PSTN line can send H.323 case: Sending the Q.931 connect signal to caller when detecting polarity reverse from PSTN line.
FXO Answer Mode	When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 1. **Ringing Answer Mode** (Default Setting): FXO answer the call once the ring coming from PSTN line. 2. **Connecting Answer Mode:** **Case A:** "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring comes from PSTN line. **Case B:** "Hot Line Number" was assigned and the hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. **(Note:** This case can avoid charging for the local PSTN call when the remote VoIP device still rings.) **Case C:** "Hot Line Number" was setting and the hot line number was assigned to another FXS port in same gateway. FXO port will not answer (off-hook) till the phone (connected to the FXS port) was picked up by user.

Note: This case can avoid the local PSTN charge when the FXS port still ring.

3. **Non Answer Mode:** FXO will NOT answer the call in any time.

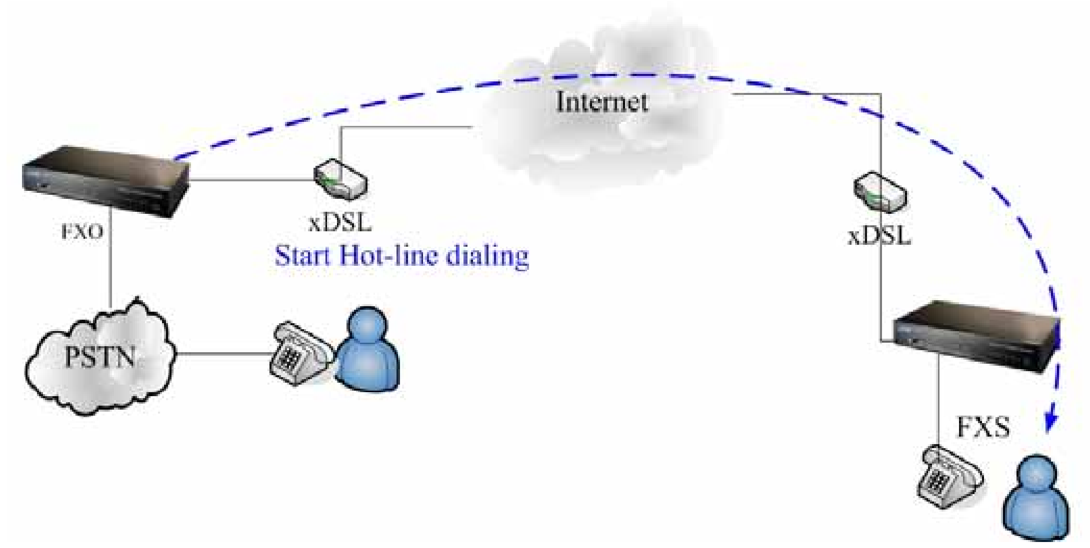
Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination

Scenario description: H.323 call connecting answer mode

Case B: "Hot Line Number" was assigned and the Hot line number belongs to remote H.323 device.

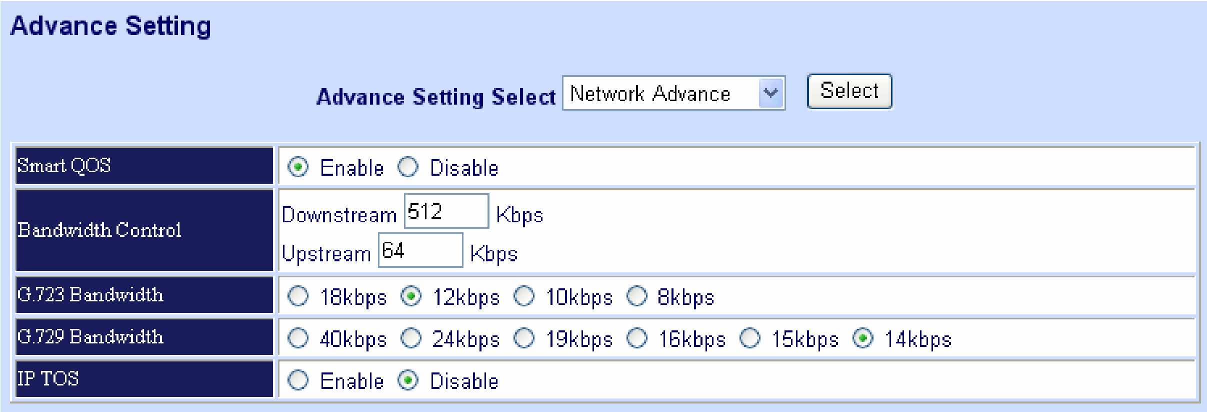
Note: The remote H.323 device need Disable the "Auto Answer"

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to remote H.323 gateway
2. The phone of remote H.323 gateway start ring.
3. When the phone was picked up, the remote H.323 Gateway send "Q.931 connects" signal to FXO port.
4. Once FXO port receives the "Q.931 connects" signal, FXO port would off-hook to answer the PSTN call.



Case C: "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same gateway.

1. When the call com from PSTN to FXO, FXO start the hot line dialing to FXS port.
2. The phone start ring.
3. Once the phone was picked up, FXO port would off-hook to answer the PSTN call.



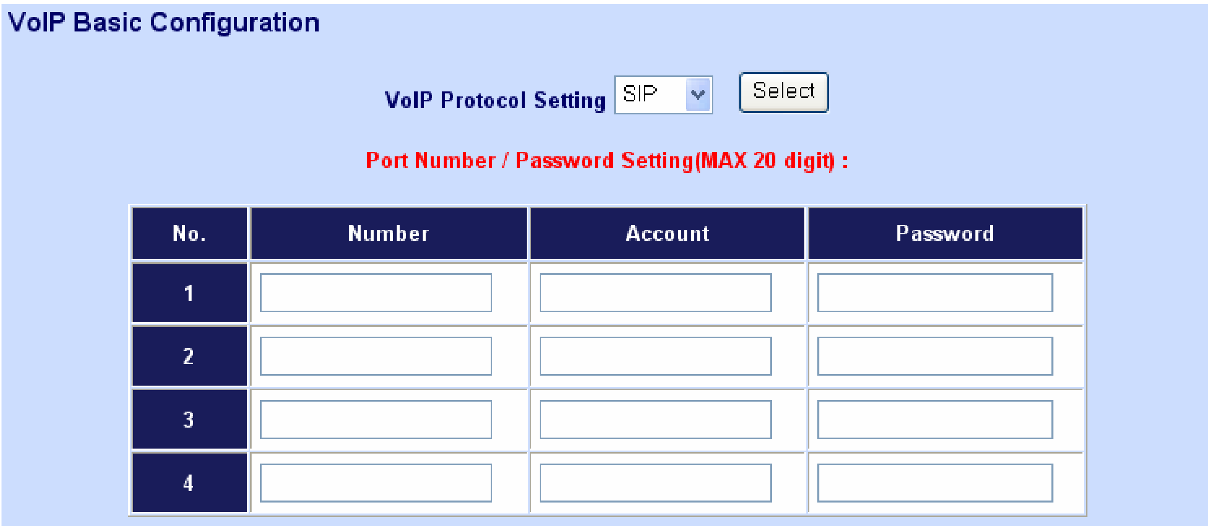
H.323 Netwrok Advance Configuration	
Smart-QoS	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control **G.723/G.729 Bandwidth**	You can configure your bandwidth what the Max byte of download and upload of ADSL modem rate.
IP TOS	Enable / Disable Type of Service in IP packets.

VoIP Basic Configuration to SIP Protocol

Select "SIP Protocol"

SIP number (username) and Password Setting: Please fill out the SIP account including username / password from ITSP.

Note: now only support digits type for SIP number / username



SIP Hunting Table: This allows gateway can answer SIP call from internet by Hunting.

For example: Port 1 and port 2 is hunting for the port 1 SIP account. If the port 1 is incoming call, the other one SIP call from internet will ring port 2.

SIP Hunting Table :

No.	Hunting Member
1	☑ Port 1 ☐ Port 2 ☐ Port 3 ☐ Port 4
2	☐ Port 1 ☑ Port 2 ☐ Port 3 ☐ Port 4
3	☐ Port 1 ☐ Port 2 ☑ Port 3 ☐ Port 4
4	☐ Port 1 ☐ Port 2 ☐ Port 3 ☑ Port 4

SIP Proxy Server Setting	
Domain/Realm	Enter the SIP realm in this field
SIP Proxy Server	Enter the SIP service IP address or domain name in this field (the domain name that comes after the @ symbol i n a full SIP URI). Use Net2Phone Service Provider
Register Interval (seconds)	This field sets how long an entry remains registered with the SIP register server. The register server can use a different time period. The gateway sends another registration request after half of this configured time period has expired.
SIP Authentication	Enable or disable MD5 authentication with SIP proxy server.
Outbound Proxy Server	The outbound proxy method is just very like the proxy server built-in NAT pass-through solution, except that the packets need to pass through the outbound proxy server.
SIP NAT Traversal Method	STUN client / Symmetric RTP

SIP Proxy Setting :

Domain/Realm	
SIP Proxy Server	0.0.0.0/0 ☐ use net2phone
Register Interval(seconds)	900
SIP Authentication	○ Enable ◉ Disable
Outbound Proxy Server	0.0.0.0/0

NAT Pass Setting:

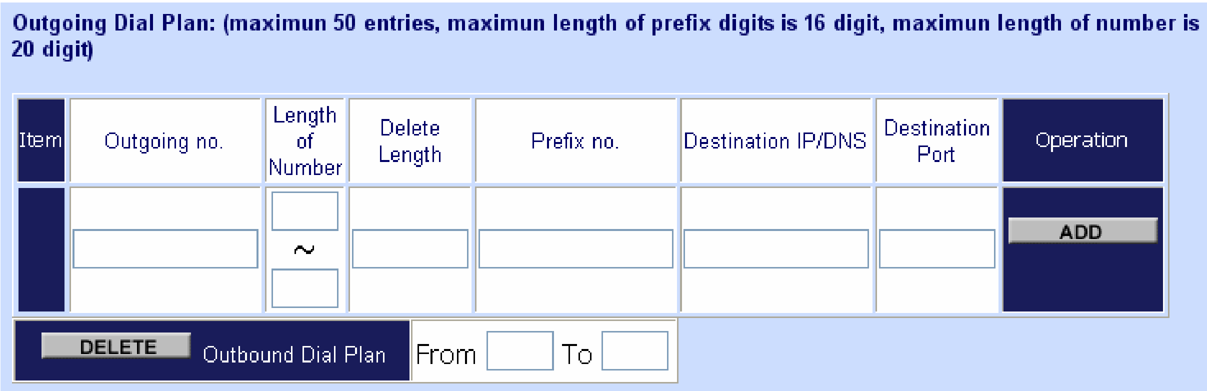
NAT Pass Method	○ STUN ◉ Symmetric RTP
STUN Server address	64.69.76.21
STUN Server port	3478

Dialing Plan to SIP protocol

The "**Dialing plan**" needs setting when the user uses the method of Peer-to-Peer or registering SIP proxy server mode. The SIP dialing plan has two kinds of directions: Outgoing (call out) and incoming (call in).

Outgoing Dial Plan	Peer-to-Peer call mode: Effective Registering to SIP Proxy Server Mode: Effective
Incoming Dial Plan	Peer-to-Peer call mode: Effective Registering to SIP proxy server mode: The leading number would **register** to SIP proxy server
PSTN Route Dial Plan	Peer-to-Peer call mode: The same as the incoming dial plan Registering to SIP proxy server mode: The leading number would **NOT register** to SIP proxy server

In the "Outgoing Dial Plan Configurations" settings: Maximum Entries : 50
"Outbound number" is the leading digits of the call out dialing number.
"Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.
"Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
"Add Digit Number" is the digits that will be added to the beginning of the dialed number.
"Destination IP Address / Domain Name" is the IP address / Domain Name of the destination gateway that owns this phone number.
"Destination Port" is the UDP port of the remote SIP proxy, which usually refer to the SIP server on the ITSP side.



Scenario description: Normally dial

2290x leading call out, call to destination domain name: sipgw.test.com

221 leading call out, call to destination IP address: 172.16.0.100

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	2209x	5 ~ 20	0	None	sipgw.test.com	5060	
2	221	3 ~ 3	0	None	172.16.0.100	5060	
		~					ADD

Outbound Dial Plan From To

Scenario description: Speed dial

If user dials "101", the gateway automatically dials "1234567890" to destination IP address: 172.16.0.101

If user dials "202", the gateway automatically dials "0987654321" to destination IP address: 172.16.0.202

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	101	3 ~ 3	3	1234567890	172.16.0.101	5060	
2	202	3 ~ 3	3	9876543210	172.16.0.202	5060	
		~					ADD

DELETE Outbound Dial Plan From To

In the "Incoming Dial Plan Configurations" settings: Maximum Entries : 50
"Inbound number" is the leading digits of the dialing number.
"Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.
"Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
"Add Digit Number" is the digits that will be added to the beginning of the dialed number.
"Destination Tele port" is **"FXS/FXO port number"**; this is for local dial plan setting phone number.

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
		~				ADD

DELETE Inbound Dial Plan	From To

Scenario description: Hunting for FXS port (VIP-400FS)

Port 1: FXS

Port 2: FXS

Port 3: FXS

Port 4: FXS

H.323 number “123”call incoming, the port 1 will be ringing.

If port 1 is busy, the port will be ringing.

If port 1 and port 2 are busy, the port 3 will be ringing.

If port 1, port 2 and port 3 are busy, the port 4 will be ringing.

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	123	3 ~ 3	3	None	1,2,3,4	
		~				ADD

DELETE Inbound Dial Plan	From To

Note: “123” will be **NOT** register to SIP Proxy Server when Gateway is Registering SIP proxy server mode

Scenario description: Hunting for FXO port (VIP-400FO)

Port 1: FXO was connected to PSTN.

Port 2: FXO was connected to PSTN.

Port 3: FXO was connected to PSTN.

Port 4: FXO was connected to PSTN.

H.323 number “123”call incoming, the port 1 will be off-hook and hear the dial tone from PSTN.

If port 1 is busy, the port will be will be off-hook and hear the dial tone from PSTN.

If port 1 and port 2 are busy, the port 3 will be off-hook and hear the dial tone from PSTN.

If port 1, port 2 and port 3 are busy, the port 4 will be off-hook and hear the dial tone from PSTN.

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	123	3 ~ 3	3	None	1,2,3,4	
		~				ADD
DELETE Inbound Dial Plan	From To					

Note: "123" will be **NOT** register to SIP proxy server when gateway is registering SIP proxy server mode

Scenario description: Termination call to FXO for one-shoot call

Port 1: FXO was connected to PSTN (area code is 81xxxxxxxx).

SIP leading number "081x" incoming, and delete the first one digit "0", and call to PSTN number.

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	081x	4 ~ 20	1	None	1	
		~				ADD
DELETE Inbound Dial Plan	From To					

Note: "081x" will be **NOT** register to SIP proxy server when gateway is registering SIP proxy server mode.

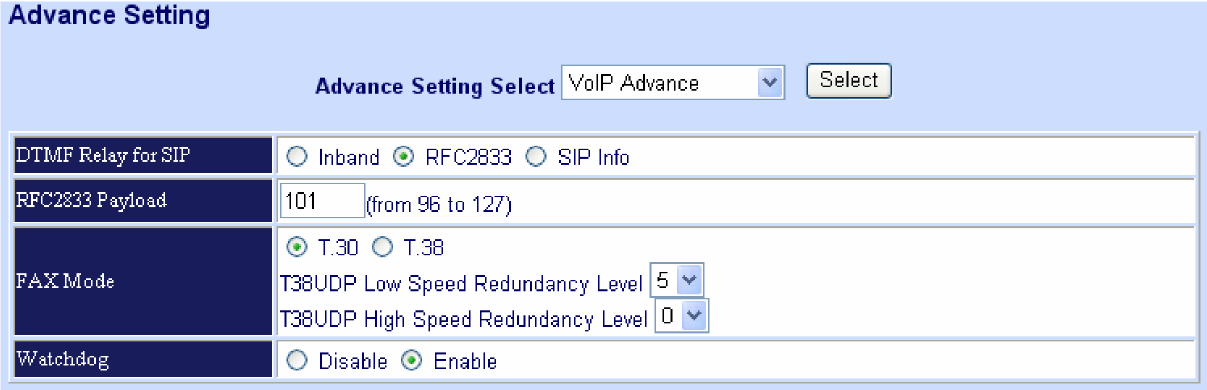
Advance Setting to SIP protocol

In Advanced Setting, VIP-GW provides user three major parts function to configure:

One is "**VoIP Advance**", the other one is "**Telephone Advance**" and "**Network Advance**"

SIP VoIP Advance Configurtion	
DTMF Method for SIP	After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are three methods of sending the DTMF tone. The first one is "in band", that is, sending the DTMF tone in the voice packet. The second one is "RFC2833", that is, sending the DTMF tone as a RTP payload signal. The third one is "SIP Info", that is, sending the DTMF tone as a SIP signal. Sending DTMF tone as a signal could tolerate more packet loss caused by the network. If this selection is enabled, the DTMF tone will be sent as a signal.

FAX Mode Option	T.30/T.38 real-time FAX compliant Voice/FAX auto-switch. The T.38 is a "Real Time Group 3 FAX communication over IP network" format. That's meaning it's a protocol for FAX over IP. You have to enable this function.



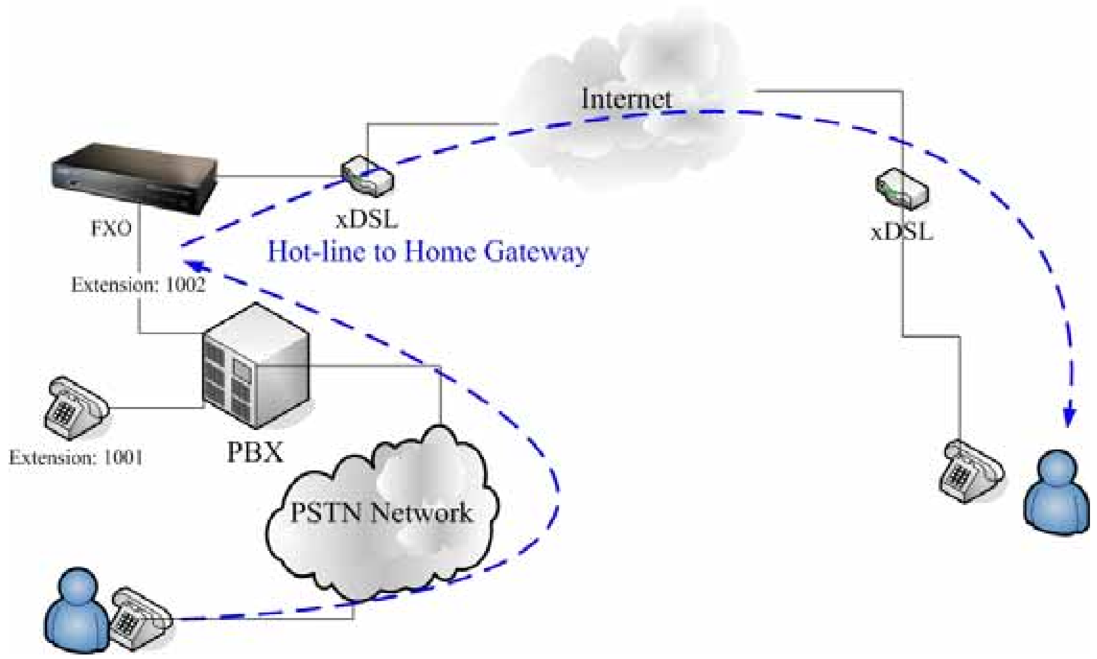
SIP Telephone Advance Configuration	
Silence Compression	If this function is enabled, when silence is occurred for a period of time, no data will be sent across the network during this period in order to save bandwidth.
Dial Complete Tone	Disable / Enable dialing complete tone.
Voice Codec option	The codec is used to compress the voice signal into data packets. Each codec has different bandwidth requirement. There are four kinds of codec, **G.723, G.729AB, G.711_u** and **G.711_A**. The default value is **G.723**.
FXS Impedance	The FXS provides 600/900 OHM impedances for selection.
FXO AC Impedance	The FXO provides wild and complex ac termination impedances for selection.
Phone (Line) in/out volume	You can adjust the phone (Line) in/out volume, range from -9db to 9db.
FXO Tx/Rx Gain	You can adjust the FXO Tx/Rx Gain , range from -6db to 6db.
UK PSTN release tone detection	When you use the gateway to UK, you can enable this selection to detection release tone.

Scenario description: Flash detection and generation duration

5. PSTN call from PSTN to office PBX and dial the extension 102 go to gateway.
6. Call to gateway of home by hotline.
7. Home user needs call transfer to extension number 101.
8. Dial flash and gateway FXS detect and generate the flash to PBX in office.

Flash Fetection: Let you change flash detection (milliseconds) of gateway when phone generate flash to FXS.

Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.



Ring Frequency	You can configure how long the Ring Frequency do you want to use.
FXO Battery Reverse	Enable battery reverse to detect polarity from PSTN line. The PSTN line can send SIP case: Sending the 200 OK connect signal to caller when detecting polarity reverse from PSTN Line.
FXO Answer Mode	When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 4. **Ringing Answer Mode** (Default Setting): FXO answer the call once the ring coming from PSTN line. **5. Connecting Answer Mode:** **Case A:** “Hot Line Number” was NOT assigned in the FXO port. FXO answer the call once the ring comes from PSTN line. **Case B:** “Hot Line Number” was assigned and the Hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. **(Note:** This case can avoid charging for the Local PSTN call when the remote VoIP device still rings.) **Case C:** “Hot Line Number” was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till

the Phone (connected to the FXS port) was picked up by user.

(**Note:** This case can avoid the Local PSTN charge when the FXS port still ring.)

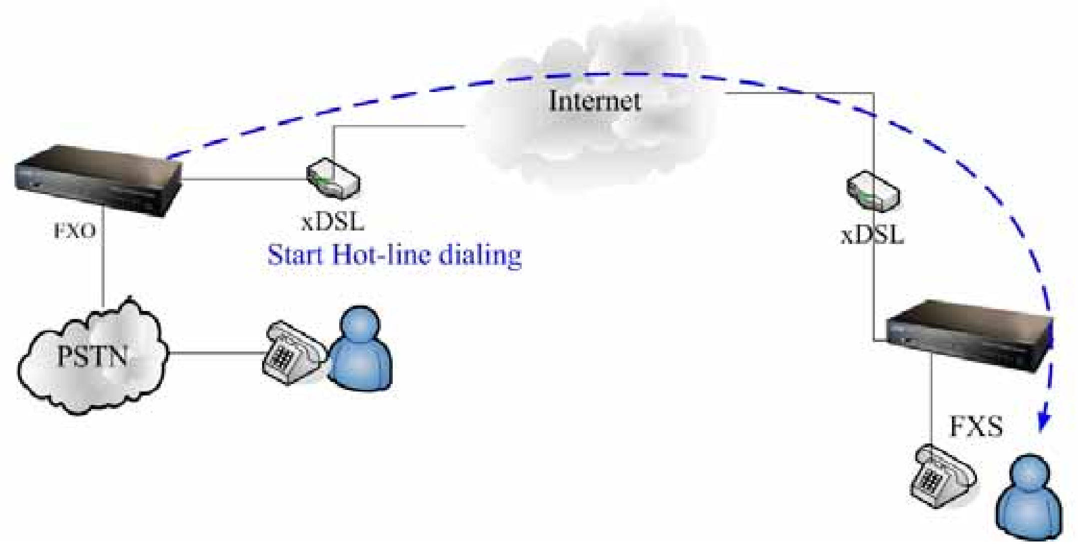
6. **Non Answer Mode:** FXO will NOT answer the call in any time.

   (**Note:** Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

Scenario description: SIP call connecting answer mode

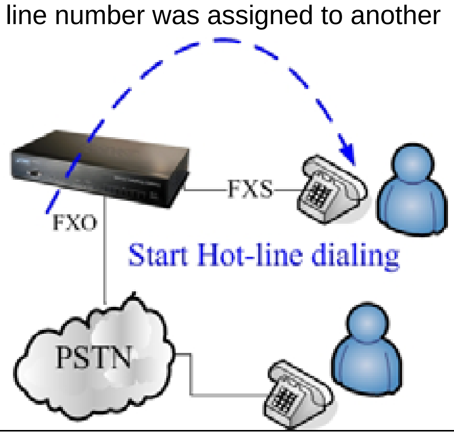
Case B: “Hot Line Number” was assigned and the hot line number belongs to SIP device.

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to remote SIP gateway
2. The phone of remote SIP gateway start ring.
3. When the phone was picked up, the remote SIP Gateway sends “SIP 200 OK” signal to FXO port.
4. Once FXO port receives the “SIP 200 OK” signal, FXO port would off-hook to answer the PSTN call.



Case C: “Hot Line Number” was setting and the Hot line number was assigned to another FXS port in same Gateway.

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to FXS port.
2. The phone start ring.
3. Once the phone was picked up, FXO port would off-hook to answer the PSTN call.



SIP Netwrok Advance Configuration	
Smart-QoS	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control **G.723/G.729 Bandwidth**	You can configure your bandwidth what the Max byte of download and upload of ADSL modem rate.
IP TOS	Enable / Disable Type of Service in IP packets.

Advance Setting

Advance Setting Select Network Advance Select

Smart QOS	◉ Enable ○ Disable
Bandwidth Control	Downstream 512 Kbps Upstream 64 Kbps
G.723 Bandwidth	○ 18kbps ◉ 12kbps ○ 10kbps ○ 8kbps
G.729 Bandwidth	○ 40kbps ○ 24kbps ○ 19kbps ○ 16kbps ○ 15kbps ◉ 14kbps
IP TOS	○ Enable ◉ Disable

Port Status

Port Status Display: This selection will display concurrent call status of this gateway. The status information of each voice channel includes codec, dialing number and destination IP address. The status is refreshed every 3 seconds.

Port Status:

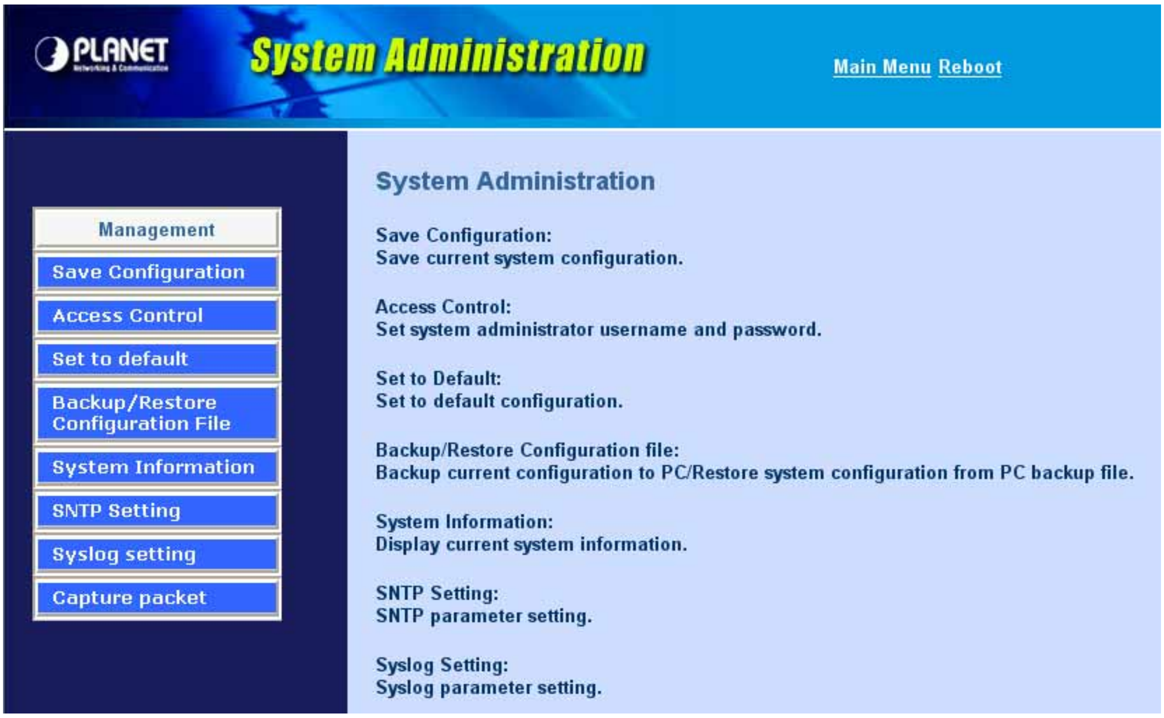
Port No.	Type	Status	Codec	Direction	Dial No.	Caller No.	Dest/Source	IN	OUT	Duration
1	FXS	onhook	none	none	none	none	none	0	0	0
2	FXS	onhook	none	none	none	none	none	0	0	0
3	FXS	onhook	none	none	none	none	none	0	0	0
4	FXO	onhook	none	none	none	none	none	0	0	0

5

Chapter 5 System Administrations

Management

Management Label	
Save Configuration	You can save configuration and restart the gateway with the default configuration or with the current running configuration.
Access Control	Users can sets/changes the administrator password...
Set to Default	You can restart the VIP-GW with the default configuration.
Backup/Restore Configuration	User can backup the configuration file of VPI-GW to PC or restore the configuration file from PC.
System Information	Display software version, WAN Type, VoIP status, VoIP codec, and phone interface and system information.
SNTP Setting	SNTP (Simple Network Time Protocol) configuration for synchronizing gateway clocks in the global Internet.
Syslog Setting	VIP-GW can send log information to Syslog Server by UDP ports 514.
Capture Packets	The VIP-GW supports packets capture and save the packets to your PC.



Save Configuration

This page allows you to click **“Save Configuration and Reboot”** to save configuration and begin to restart.

Access Control

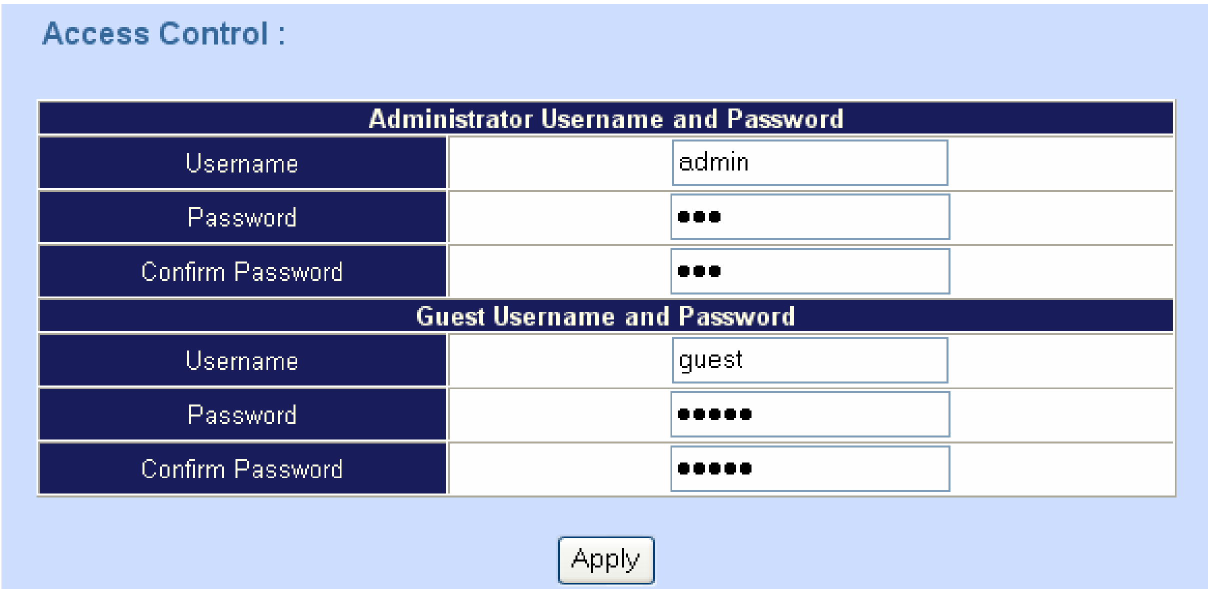
Changing the Administrator/Guest Password

For security reasons, we strongly recommend that you set an administrator/password for the router. On first setup the router requires no password. If you don’t set a password the router is open and can be logged into and settings changed by any user from the local network or the Internet.

Click **Access Control Setup**, the following screen will open.

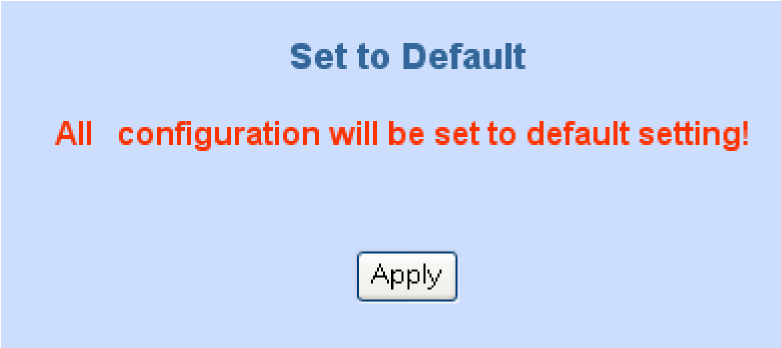
Administrator username/password: **admin/123**

Guest username/password: **guest/guest**



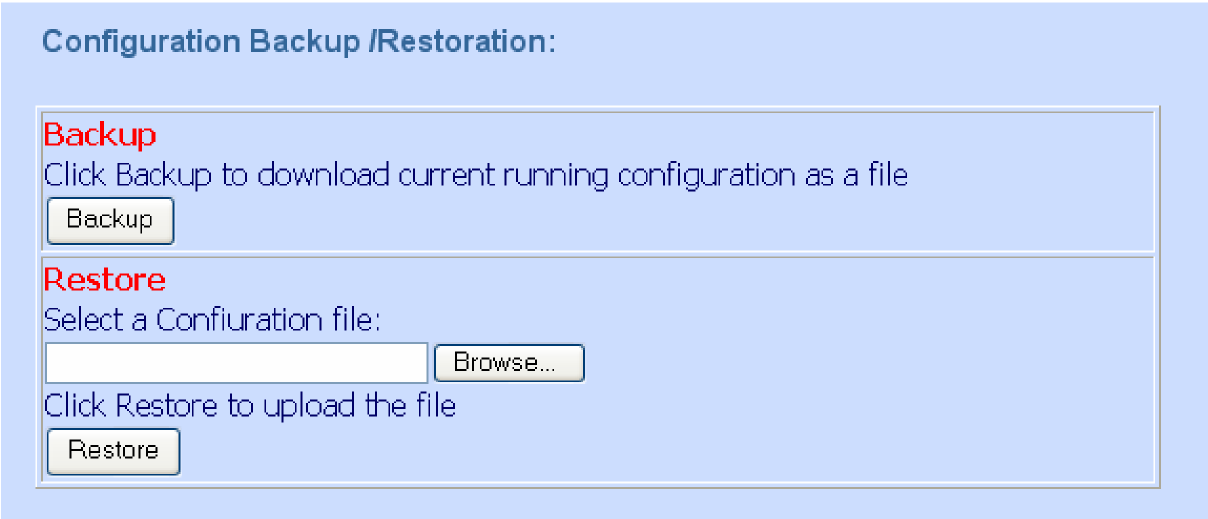
Set To Default Configuration

If you want to reboot the router using **factory default configuration**, click “**Apply**” then reset the router’ s settings to default values.



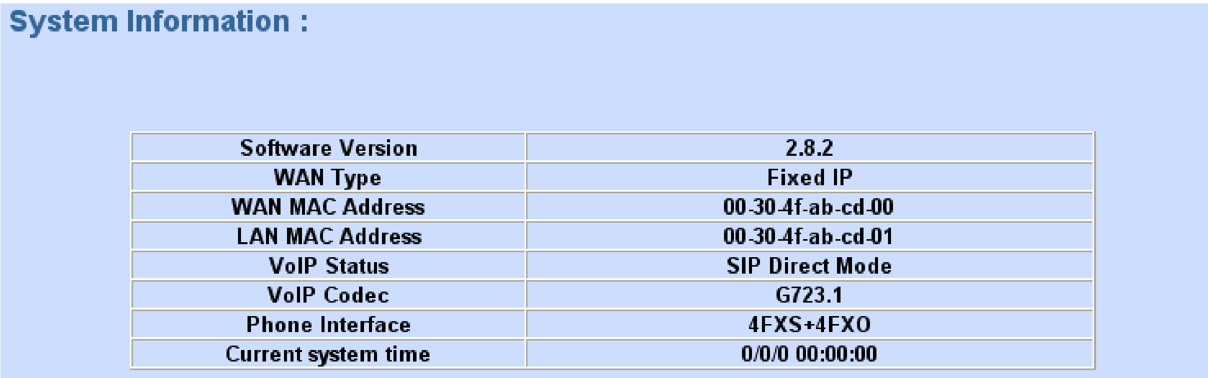
Backup/Restore Configuration to a File

User can backup the configuration to a File at Microsoft Operation System. And also restore the configuration file to the VIP-GW from PC.



System Information Display Function

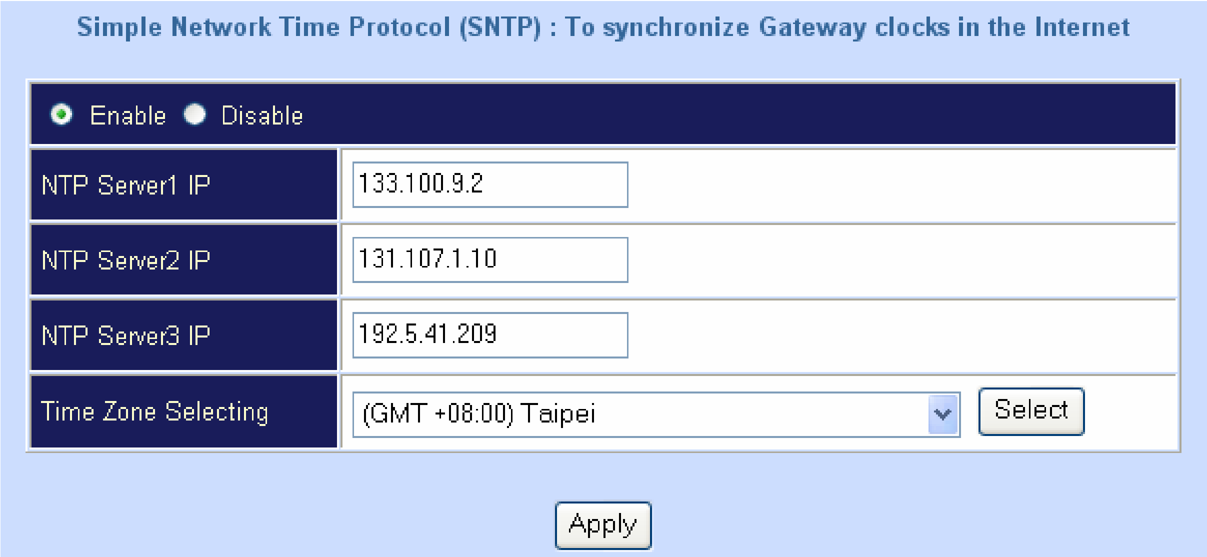
Click **System Information Display to** open the Online Status page. In the example, on the following page, both PPPoE connections is up on the WAN interface, H323/SIP Status, MAC address, Register Status.., etc.



Software Version	2.8.2
WAN Type	Fixed IP
WAN MAC Address	00-30-4f-ab-cd-00
LAN MAC Address	00-30-4f-ab-cd-01
VoIP Status	SIP Direct Mode
VoIP Codec	G723.1
Phone Interface	4FXS+4FXO
Current system time	0/0/0 00:00:00

SNTP Setting Function

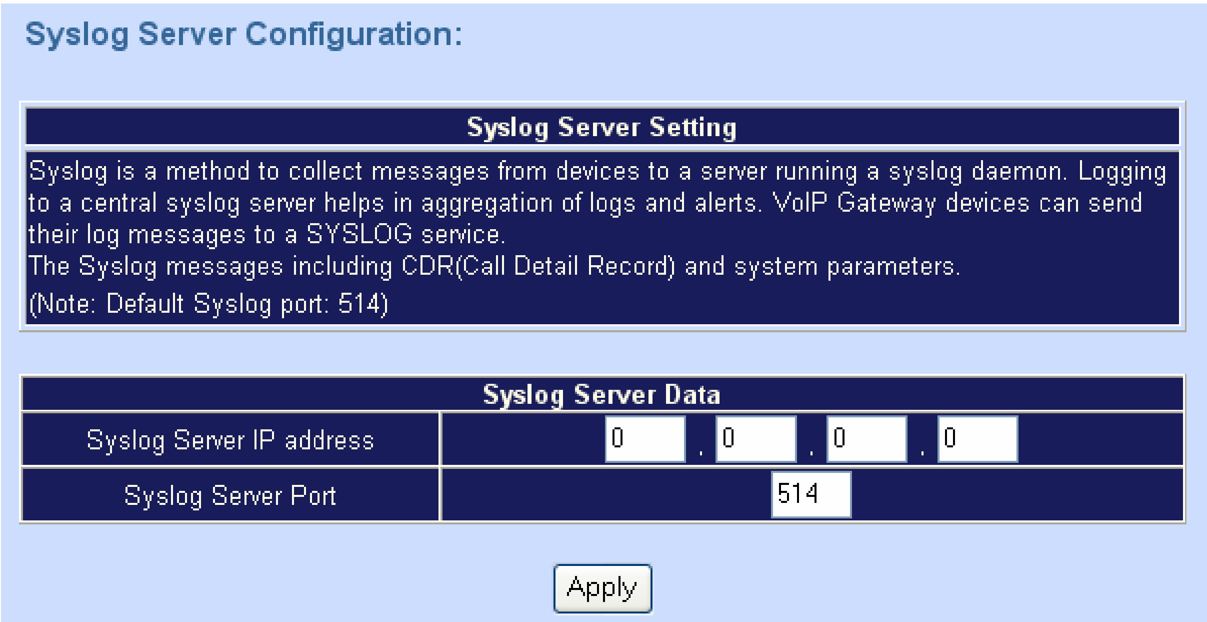
Click **SNTP setting to** open the Online Status page. In the example, on the following page:

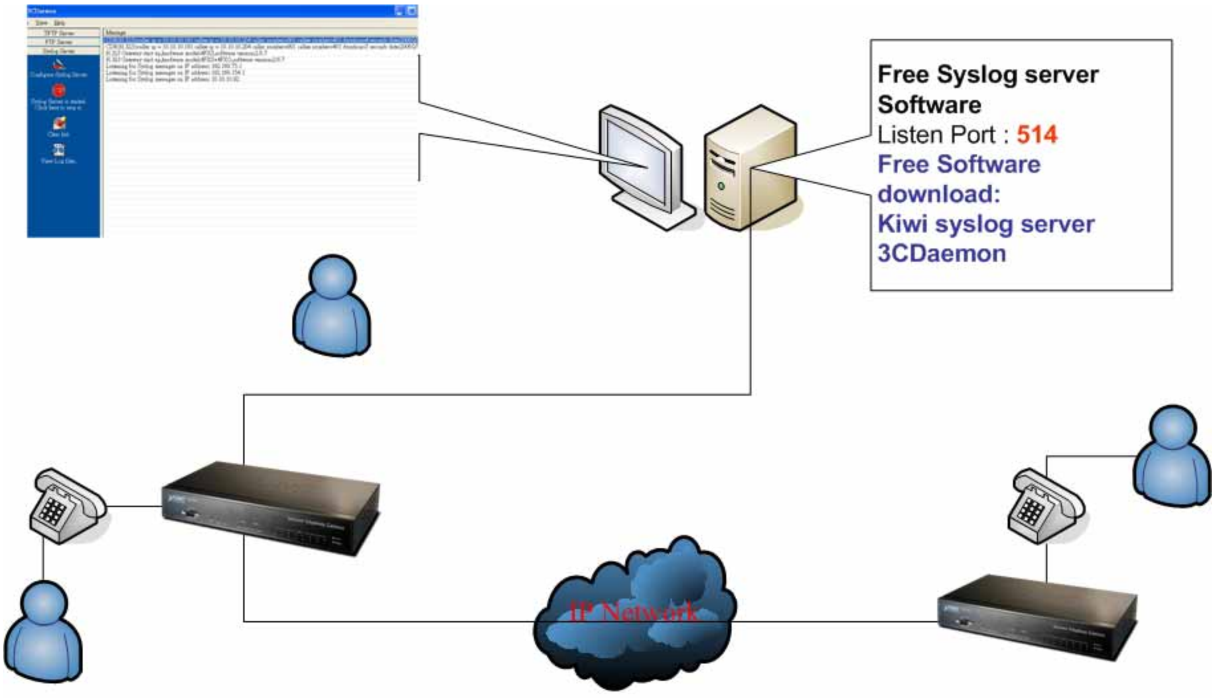


Use SNTP Setting— when checked, gateway uses a Simple Network Time Protocol (SNTP) to set the date and time. The gateway synchronizes the gateway's time after you select the time zone. Use SNTP Setting; select the time zone which gateway was at.

Syslog setting

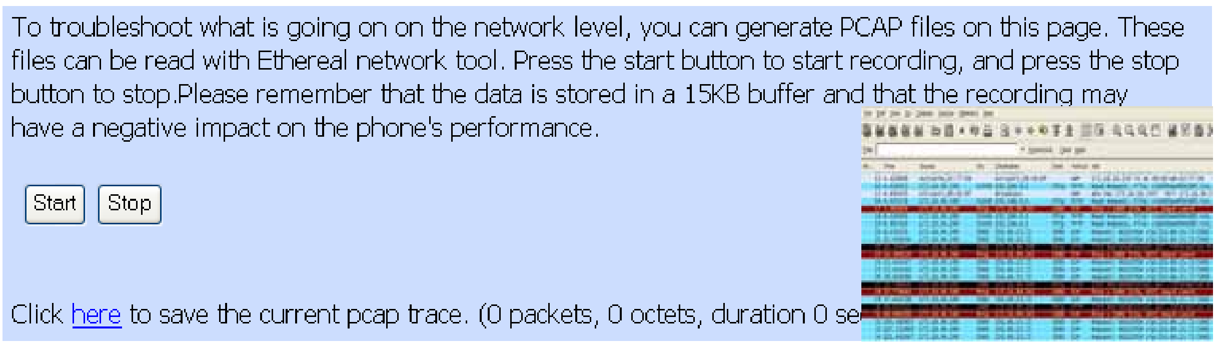
Use Syslog server to record your VIP-GW log file. To set the Syslog server IP address for this function. Kindly please download for this FREE service at http://www.kiwisyslog.com/index.php for more understandings.





Capture packetackets Function

Use “Capturer Packets” to record VIP-GW packets. Users can start and stop the capture then save the file to PC. Use the Ethereal Tool (www.ethereal.com) to analyze the packets.



Appendix A

Voice communications

The chapter shows you the concept and command to help you configure your PLANET VIP-GW through sample configuration. And provide several ways to make calls to desired destination in VIP-GW. In this section, we'll lead you step by step to establish your first voice communication via web browsers operations.

Concepts: Voice port

There are two type of the voice port, **FXO** (Foreign exchange Office) and **FXS**. (Foreign exchange Station) On the printing of the RJ-11 port, you should find that.

FXO (Foreign exchange Office) port

The FXO port allows the connection with a device that already has a fixed number; say 222, or 412-1111. So the only connections for FXO port will be to your local PSTN or one of your extension-line from your PBX system.

With your FXO connect to PSTN; the Internet Voice can then have a local call through this line/number (412-1111). Or, locally, you can have an Internet Call through the line 412-1111

The same to PBX system, you are required to know with which extension number to the FXO port. Your PBX users will need to know this number in the future.

☝ **Hint**	FXO port cannot connect to an end-node like telephone or fax machine (since they do not provide a number!). If you connect those to FXO port, you will hear nothing once you pick up the handset.

FXS (Foreign exchange Station) port

The FXS port allows the connection to an end node, like **telephone**, **fax machine**, or **out-line of PBX system**.

FXS port is as like your local phone service provider who provides a number to you. It is easy to tell that after you have connected an end-device to FXS port and you will hear the dial-tone from FXS port once the hand set off-hook.

Caution	The FXS port is with voltage and current. **DO NOT** connects the port to any PBX extension line or PSTN line. This may make the FXS port or your PBX extension port malfunction.

H.323 VoIP Call: Peer-To-Peer Mode

Scenario 1: Gateway 1 to Gateway 2 PLAR connection

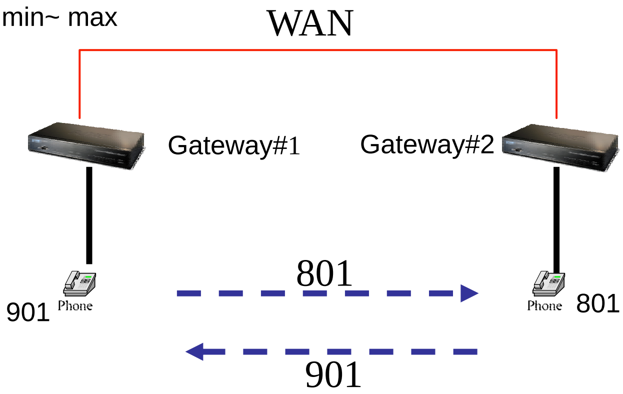
H.323 Call (Peer-To-Peer Mode)

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3 |Des: GW2 IP address	No: 9x | Digit: 3~3 | Des: GW1 IP address

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 2: Gateway 1 (with PBX) to Gateway 2 PLAR connection

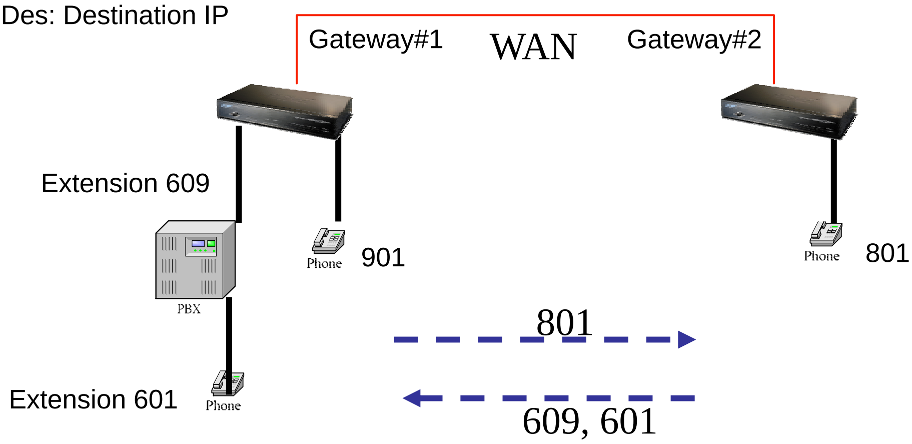
H.323 Call (Peer-To-Peer Mode) with PBX: Call PBX Extension

Method 1: Two-Stage-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3 | Des: GW2 IP address	No: 9x | Digit: 3~3 | Des: GW1 IP address No: 6x | Digit: 3~3 | Des: GW1 IP address

x: wild card

Des: Destination IP



Scenario 3: Gateway 1 (with PBX/PSTN) to Gateway 2 PLAR connection

Call Method: Two-Stages-Dialing

H.323 Call (Peer-To-Peer Mode) with PBX: Remote Call PSTN number

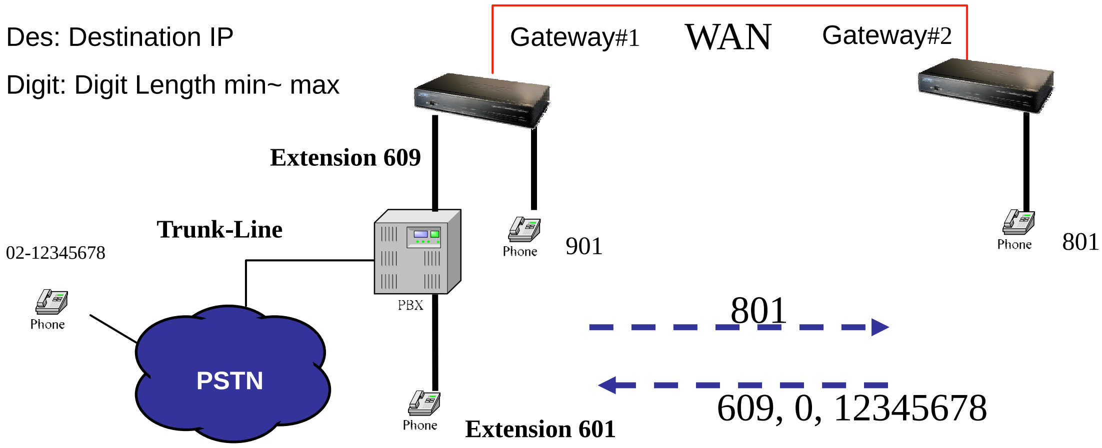
Method 1: Two-Stages-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x I Digit:3~3 I Des: GW2 IP address	No: 9x | Digit: 3~3 | Des: GW1 IP address No: 6x I Digit:3~3 I Des :GW1 IP address

X: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 4: Gateway 1 (with PBX/PSTN) to Gateway 2 PLAR connection

Call Method: One-Shot-Dialing

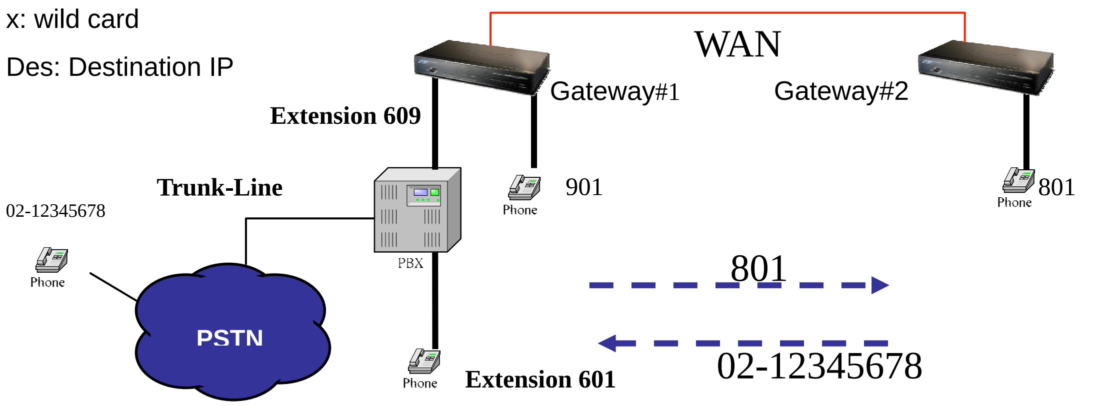
H.323 Call (Peer-To-Peer Mode) with PBX: Remote Call PSTN number

Method 2: One-Shot-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3 |Des:GW2 IP address	No: 9x | Digit: 3~3 | Des: GW1 IP address
Incoming Dial Plan	No: 6x | Digit: 3~3 | Des: GW1 IP address
No: 02x| Digit:3~10 |Strip:2 |Prefix: 0,,,|FXO port	No: 02xl Digit: 3~10 I Des: GW 1 IP address

x: wild card

Des: Destination IP



Scenario 5: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

H.323 Call (Peer-To-Peer Mode) : Remote Call PSTN number

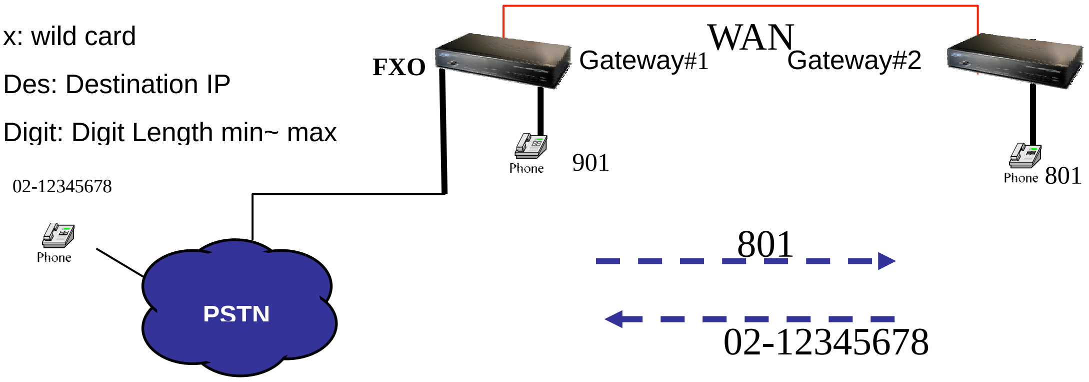
Method: One-Shot-Dialing

Outgoing Dial plan No: 8x | Digit: 3~3, Des | GW2 IP address Incoming Dial Plan No: 02x | Digit: 3~10 |Strip:2 | FXO port	Outgoing Dial plan No: 9x | Digit: 3~3 | Des: GW1 IP address No: 6x | Digit: 3~3 | Des: GW1 IP address No: 02x| Digit: 3~10 | Des: GW 1 IP address

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 6: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

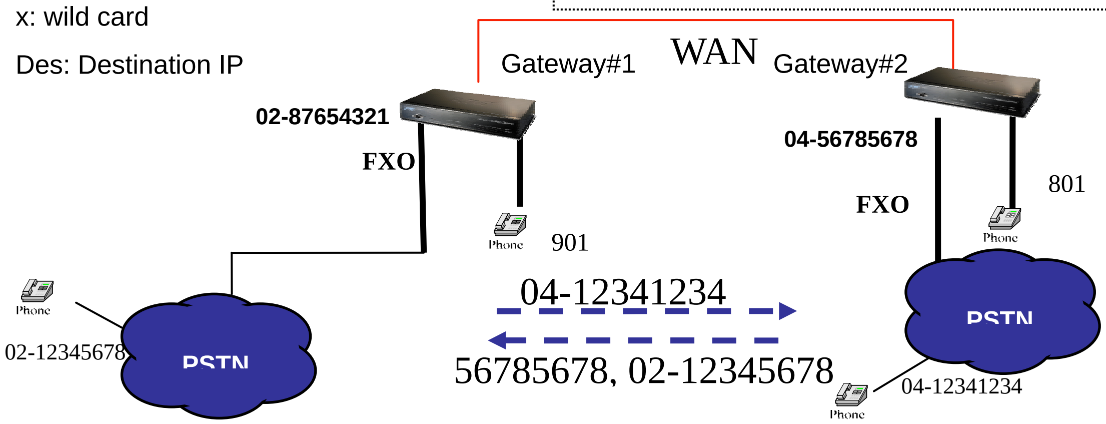
H.323 Call (Peer-To-Peer Mode) : PSTN Call PSTN number

Method: One-Shot-Dialing

Outgoing Dial plan No: 8x | Digit: 3~3 | Des: GW2 IP address No: 04x | Digit: 3~10 | Des: GW2 IP address Incoming Dial Plan No: 02x | Digit: 3~10 |Strip:2 | FXO port	Outgoing Dial plan No: 9x | Digit: 3~3 | Des: GW1 IP address No: 6x | Digit: 3~3 | Des: GW1 IP address No: 02x| Digit: 3~10 | Des: GW 1 IP address Incoming Dial Plan No: 04x | Digit: 3~10 | Strip:2 | FXO port

x: wild card

Des: Destination IP



H.323 VoIP Call: Gatekeeper Mode

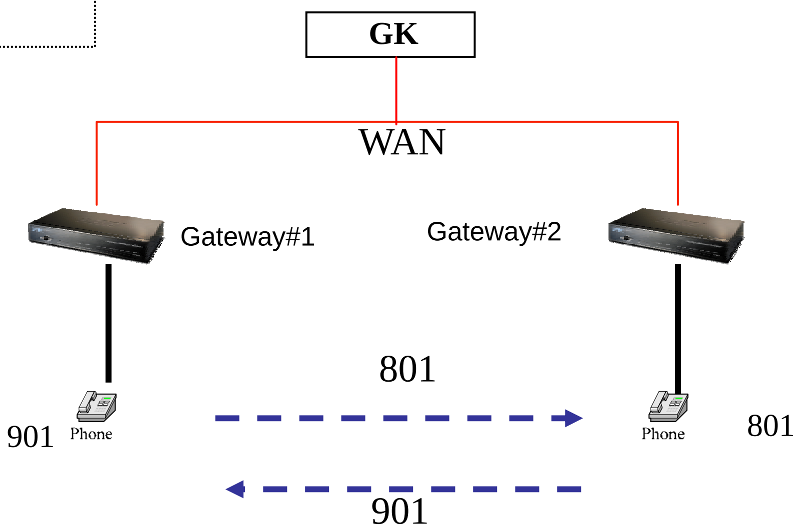
Scenario 7: Gateway 1 to Gateway 2 PLAR connection

H.323 Call (GK Mode)

Register Number List

GW1: 801

GW2: 901



Scenario 8: Gateway 2 to Gateway 1 (Call PBX extension number) PLAR connection

Call Method: Two-Stages-Dialing

H.323 Call (GK Mode) with PBX: Call PBX Extension

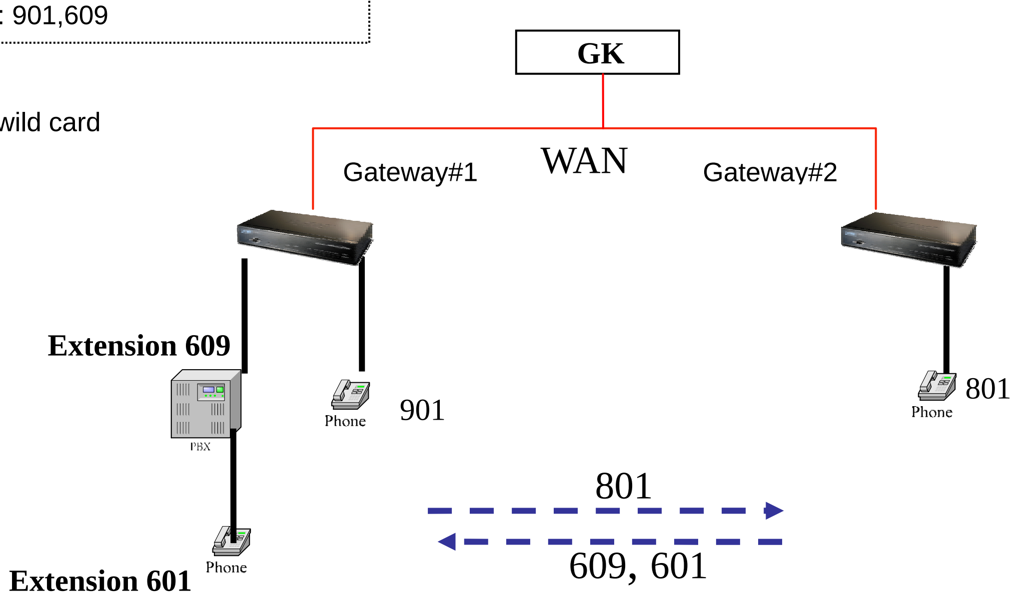
Method 1: Two-Stage-Dialing

Register Number List

GW1: 801

GW2: 901,609

x: wild card



Scenario 9: Gateway 2 to Gateway 1 (Remote Call PSTN number with PBX) PLAR connection

Call Method: Two-Stages-Dialing

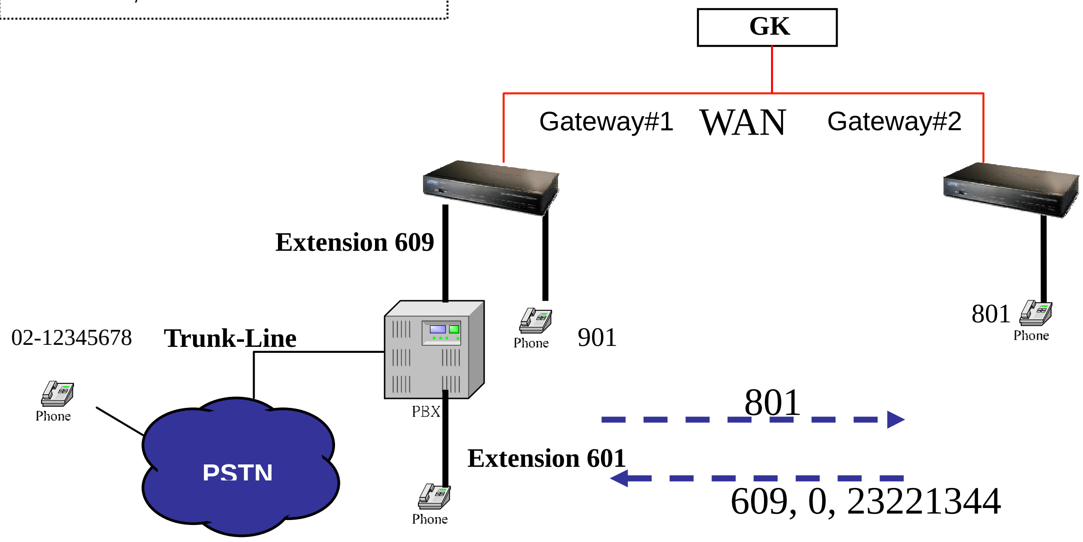
H.323 Call (GK Mode) with PBX: Remote Call PSTN number

Method 1: Two-Stages-Dialing

Register Number List

GW1: 801

GW2: 901,609



Scenario 10: Gateway 2 to Gateway 1 (Remote Call PSTN number with PBX) PLAR connection

Call Method: One-Shot-Dialing

H.323 Call (GK Mode) with PBX: Remote Call PSTN number

Method 2: One-Shot-Dialing

Incoming Dial Plan

No: 02x | Digit: 3~10 |Strip:2 |Prefix: 0, | FXO port

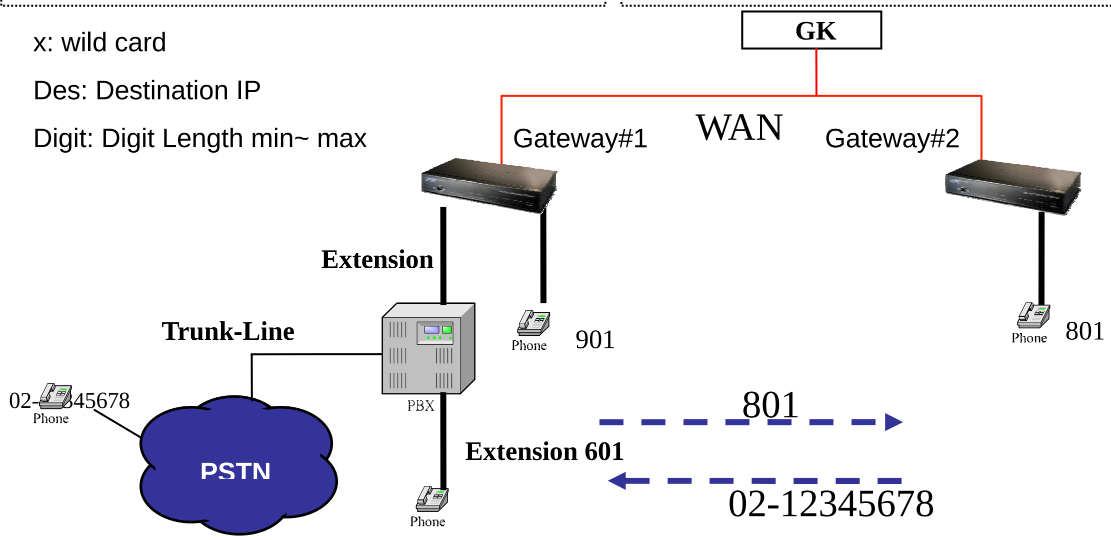
Register Number List

GW1: 801, GW2: 901,609,02x

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 11: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

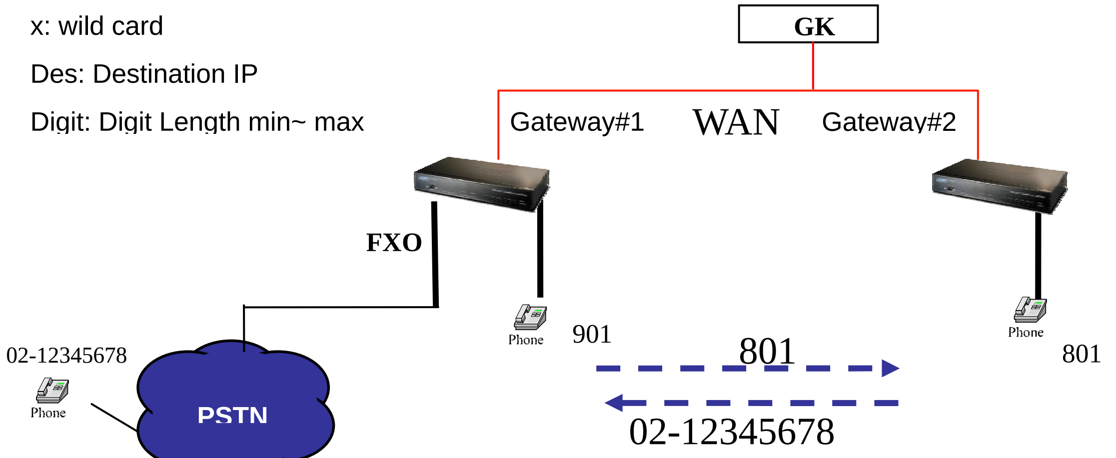
H.323 Call (GK Mode) : Remote Call PSTN number
Method: One-Shot-Dialing

Incoming Dial Plan	Register Number List
No: 02x | Digit: 3~10 | Strip:2 | FXO port	GW1: 801, GW2: 901,609,02x

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 12: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

H.323 Call (GK Mode) : PSTN Call PSTN number
Method: One-Shot-Dialing

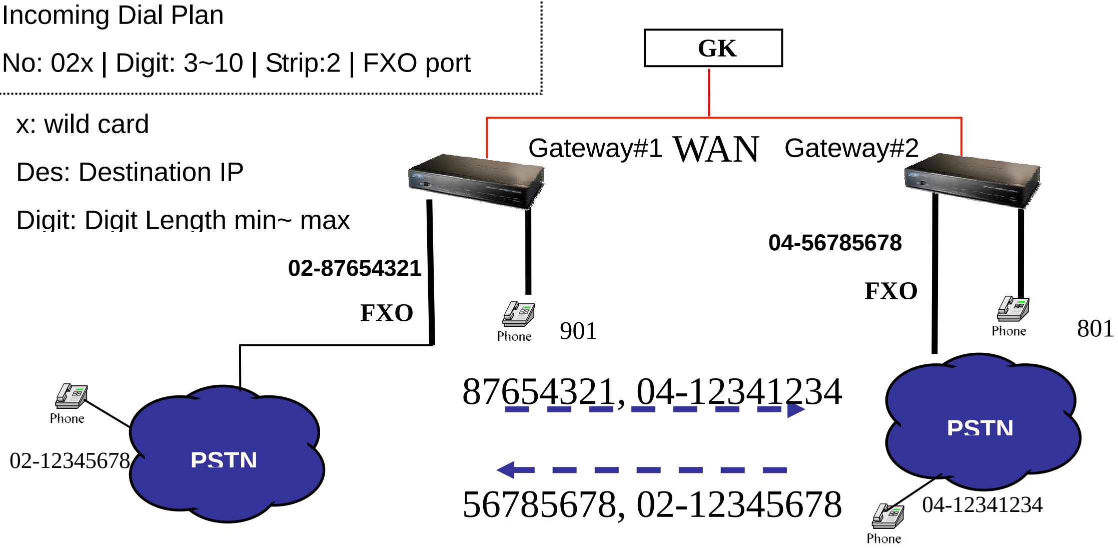
Register Number List	Incoming Dial Plan
GW1: 801 , GW2: 901,609,02x, 04 xs	No: 04x | Digit: 3~10 | Strip:2 | FXO port

Incoming Dial Plan
No: 02x | Digit: 3~10 | Strip:2 | FXO port

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



SIP VoIP Call: Peer-To-Peer Mode

Scenario 13: Gateway 1 to Gateway 2 PLAR connection

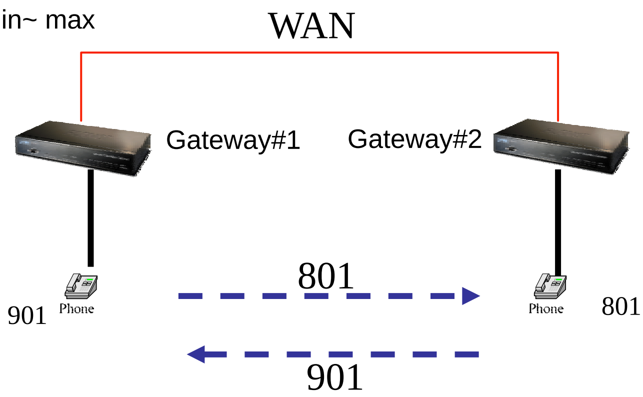
SIP Call (Peer-To-Peer Mode)

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3, Des | GW1 IP address	No: 9x | Digit: 3~3, Des | GW1 IP address

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 14: Gateway 2 to Gateway 1 (Call PBX extension number) PLAR connection

Call Method: Two-Stages-Dialing

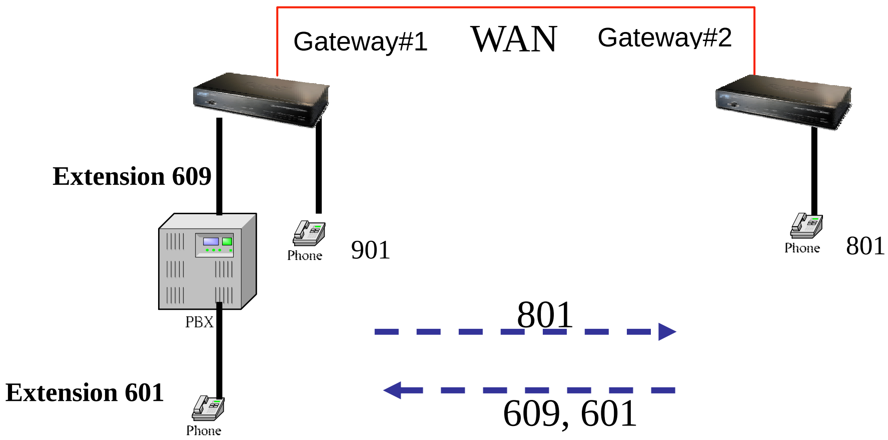
SIP Call (Peer-To-Peer Mode) with PBX: Call PBX Extension

Method 1: Two-Stage-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3 |Des GW2 IP address	No: 9x | Digit: 3~3 |Des: GW1 IP address No: 6x | Digit: 3~3 |Des: GW1 IP address

x: wild card

Des: Destination IP



Scenario 15: Gateway 2 to Gateway 1 (Remote Call PSTN number with PBX) PLAR connection

Call Method: Two-Stages-Dialing

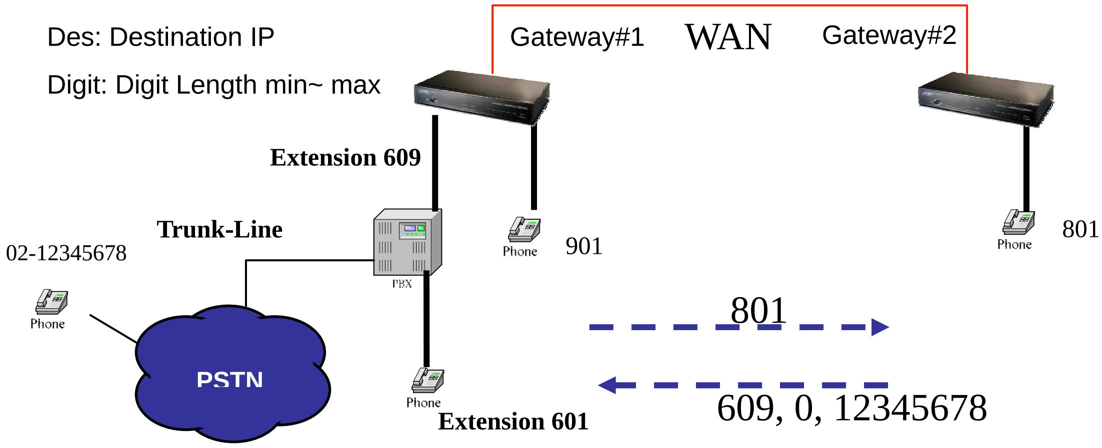
SIP Call (Peer-To-Peer Mode) with PBX: Remote Call PSTN number
Method 1: Two-Stages-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3, Des | GW2 IP address	No: 9x | Digit: 3~3, Des | GW1 IP address
	No: 6x | Digit: 3~3, Des | GW1 IP address

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 16: Gateway 2 to Gateway 1 (Remote Call PSTN number with PBX) PLAR connection

Call Method: One-Shot-Dialing

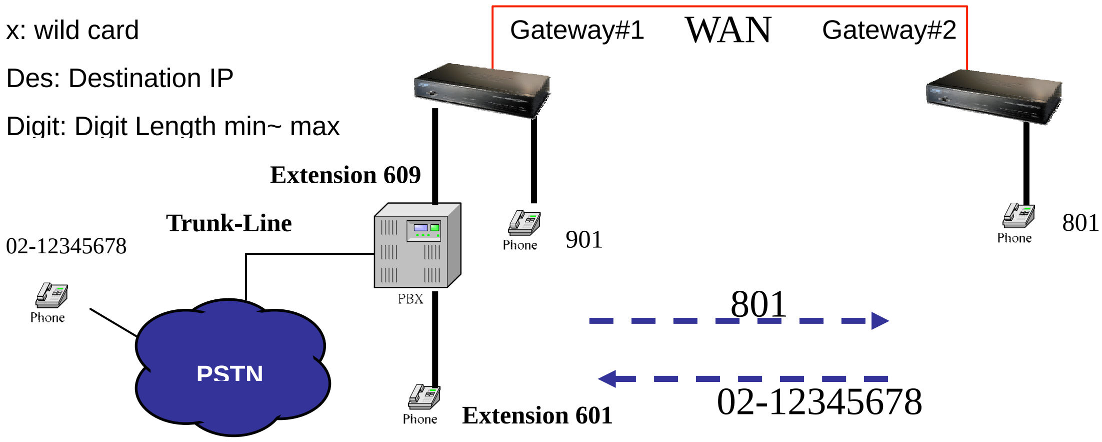
SIP Call (Peer-To-Peer Mode) with PBX: Remote Call PSTN number
Method 2: One-Shot-Dialing

Outgoing Dial plan	Outgoing Dial plan
No: 8x | Digit: 3~3, Des | GW2 IP address	No: 9x | Digit: 3~3 | Des: GW1 IP address
Incoming Dial Plan	No: 6x | Digit: 3~3 | Des: GW1 IP address
No:02x | Digit: 3~10 |Strip:2|Prefix:0,,,| FXO port	No: 02x| Digit: 3~10 | Des: GW 1 IP address

x: wild card

Des: Destination IP

Digit: Digit Length min~ max



Scenario 17: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

SIP Call (Peer-To-Peer Mode) : Remote Call PSTN number

Method: One-Shot-Dialing

Outgoing Dial plan

No: 8x | Digit: 3~3, Des | GW2 IP address

Incoming Dial Plan

No: 02x | Digit: 3~10 | Strip:2 | FXO port

Outgoing Dial plan

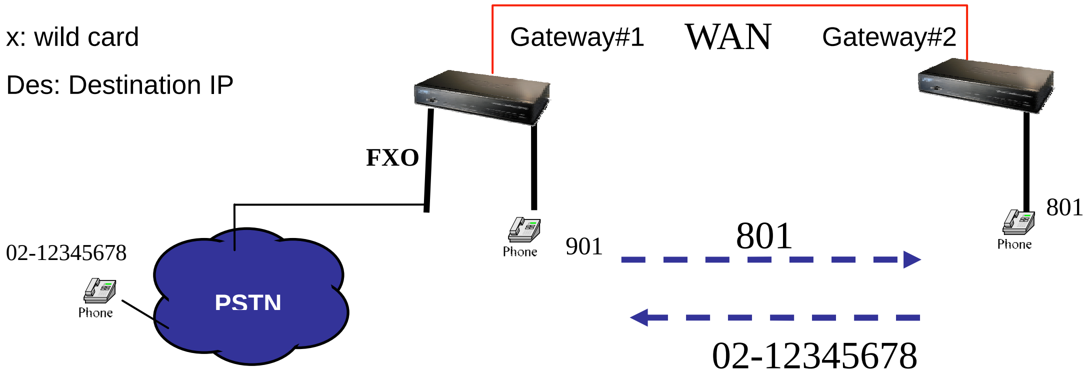
No: 9x | Digit: 3~3 | Des: GW1 IP address

No: 6x | Digit: 3~3 | Des: GW1 IP address

No: 02x| Digit: 3~10 | Des: GW 1 IP address

x: wild card

Des: Destination IP



Scenario 18: Gateway 2 to Gateway 1 (PSTN Call PSTN number) PLAR connection

Call Method: One-Shot-Dialing

SIP Call (Peer-To-Peer Mode) : PSTN Call PSTN number

Method: One-Shot-Dialing

Outgoing Dial plan

No: 8x | Digit: 3~3, Des | GW2 IP address

No: 04x| Digit: 3~10 | Des: GW2 IP address

Incoming Dial Plan

No: 02x | Digit:3~10 | Strip :2 | FXO port

Outgoing Dial plan

No: 9x | Digit: 3~3 | Des: GW1 IP address

No: 6x | Digit: 3~3 | Des: GW1 IP address

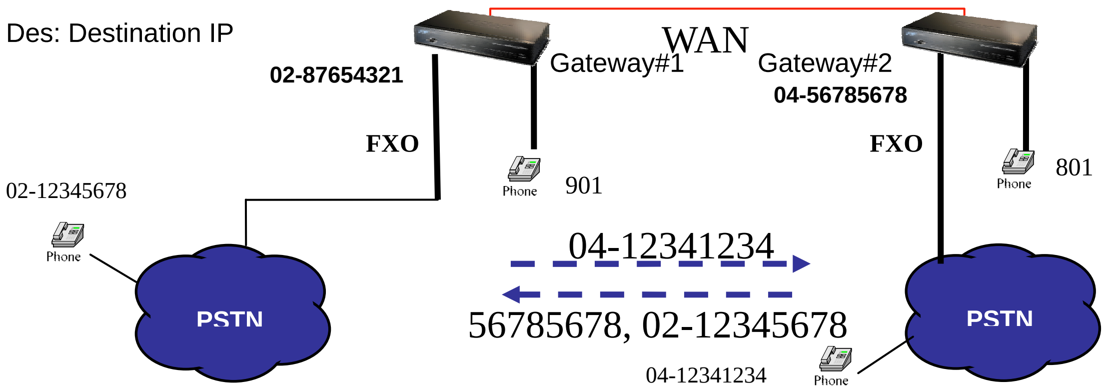
No: 02x| Digit: 3~10 | Des: GW 1 IP address

Incoming Dial Plan

No: 04x | Digit: 3~10 | Strip:2 | FXO port

x: wild card

Des: Destination IP

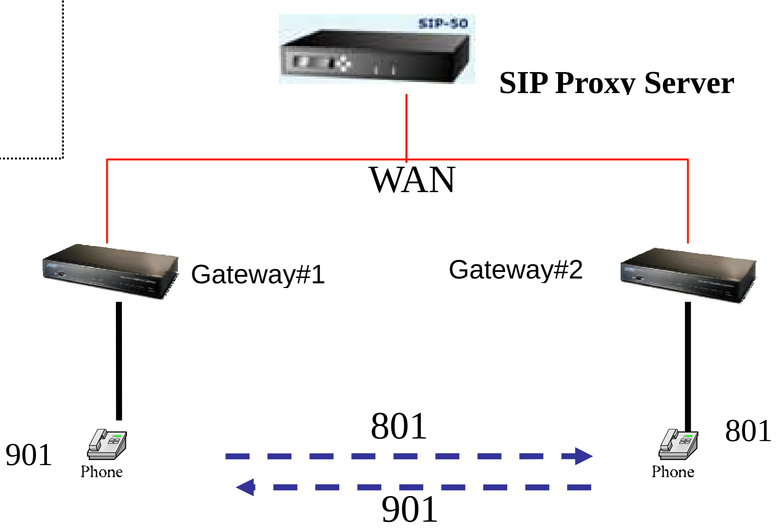


SIP VoIP Call: SIP Proxy Server

Scenario 19: Gateway 1 to Gateway 2 PLAR connection

SIP Call (Register to SIP Proxy Server Mode)

Register Number List
GW1: 801
GW2: 901

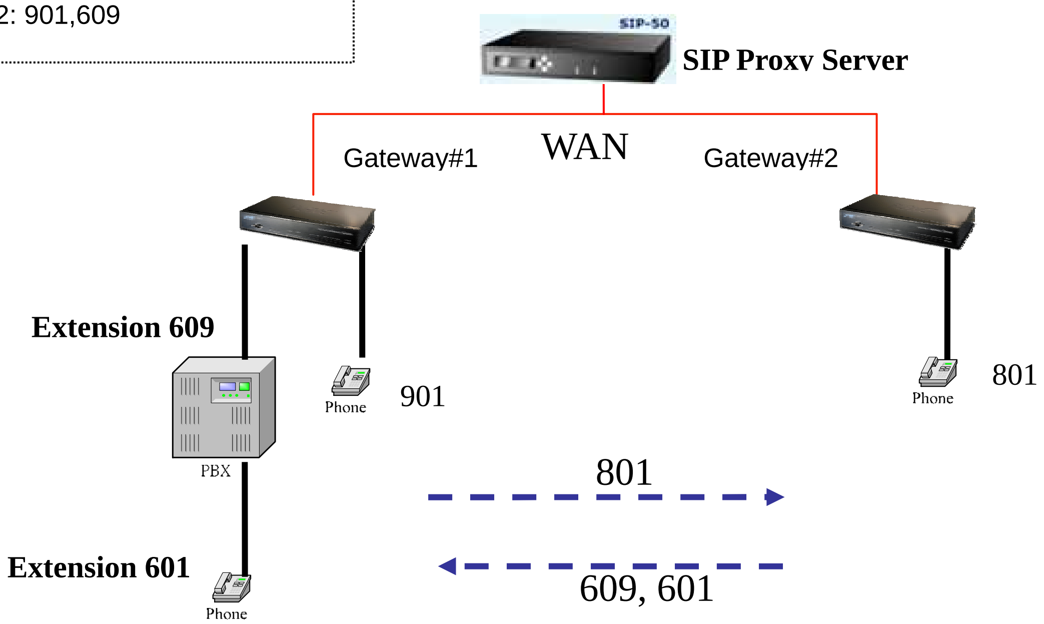


Scenario 20: Gateway 2 to Gateway 1 (Call PBX extension number) PLAR connection

Call Method: Two-Stages-Dialing

SIP Call (SIP Proxy Server Mode) with PBX: Call PBX Extension
Method 1: Two-Stage-Dialing

Register Number List
GW1: 801
GW2: 901,609



Scenario 21: Gateway 2 to Gateway 1 (Remote Call PSTN number with PBX) PLAR connection

Call Method: Two-Stages-Dialing

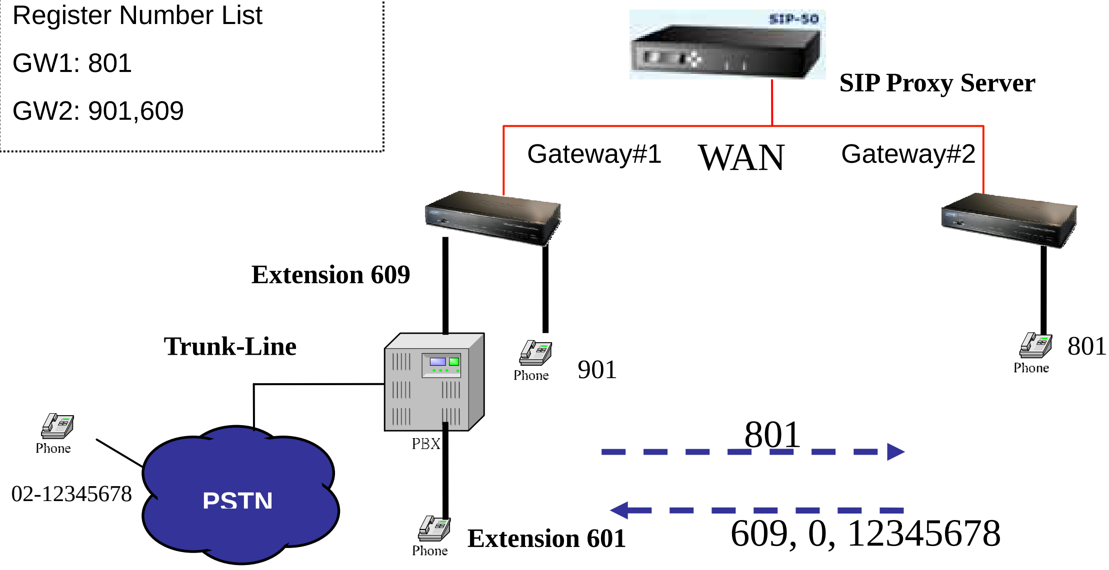
SIP Call (SIP Proxy Server Mode) with PBX: Remote Call PSTN number

Method: Two-Stages-Dialing

Register Number List

GW1: 801

GW2: 901,609



Scenario 22: Gateway 2 to Gateway 1 (Remote Call PSTN number) PLAR connection

Call Method: Two-Stages-Dialing

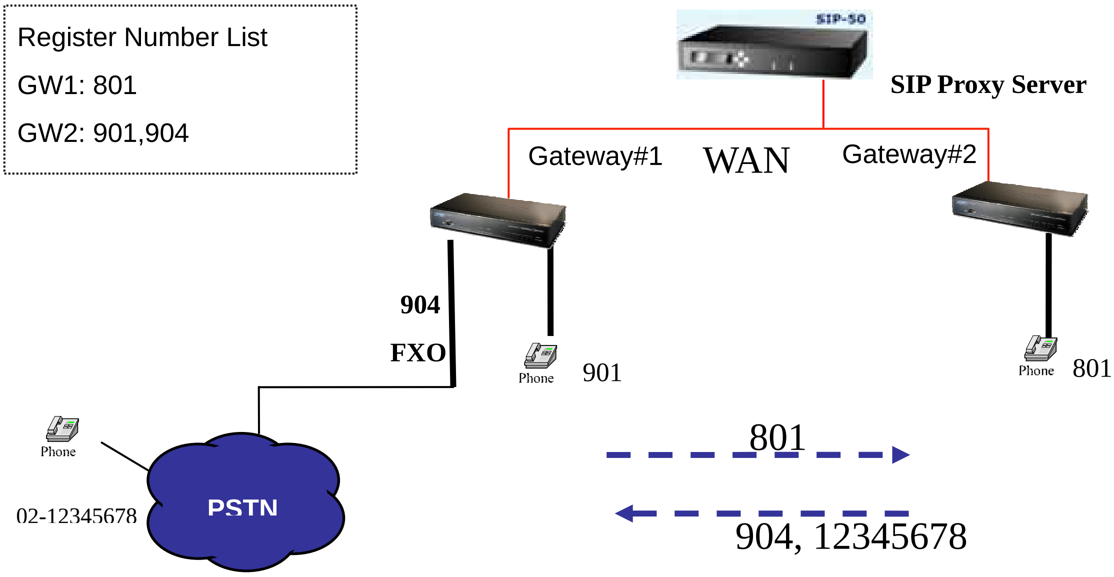
SIP Call (SIP Proxy Server Mode) : Remote Call PSTN number

Method: Two-Stages-Dialing

Register Number List

GW1: 801

GW2: 901,904



Scenario 23: Gateway 2 to Gateway 1 (PSTN Call PSTN number) PLAR connection

Call Method: Two-Stages-Dialing

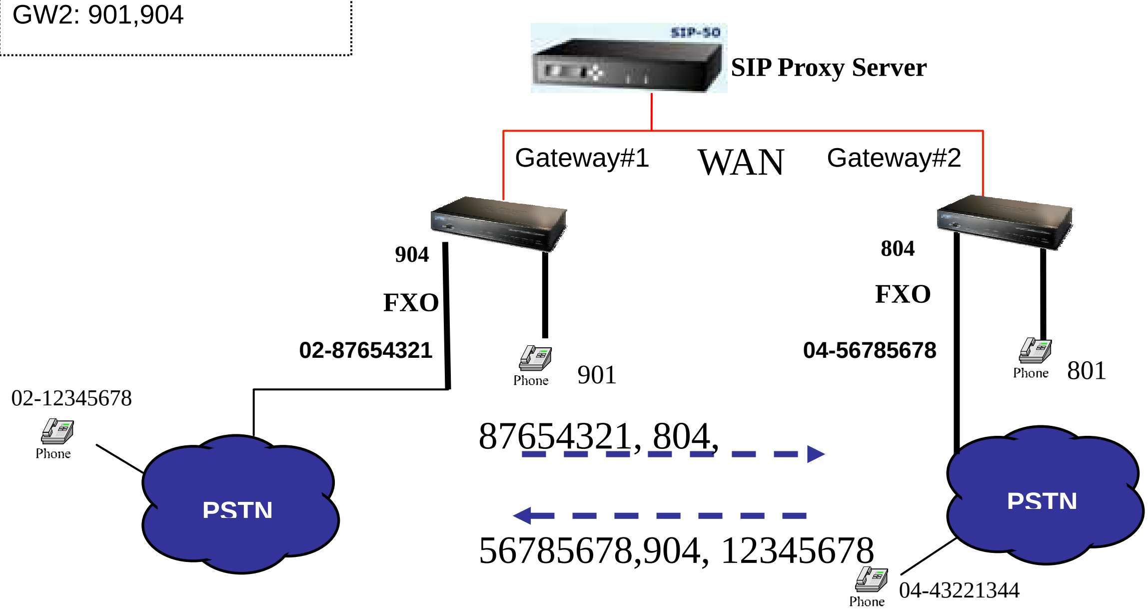
SIP Call (SIP Proxy Server Mode)　: PSTN Call PSTN number

Method: Two-Stages-Dialing

Register Number List

GW1: 801,804

GW2: 901,904



Appendix B

FAQ

Q: What is the default administrator password to login to the gateway?
A: By default, your default username is “**admin**”; default password is “**123**” to login to the router. For security, you should modify the password to protect your gateway against hacker attacks. **Note:** Default guest login username/password: **guest/guest**
Q: I forgot the administrator password. What should I do?
A: Press the **Reset** button on the rear panel for over **5** seconds to reset all settings to default values.
Q: What is the default IP address of the router?
A: The default WAN IP address is 172.16.0.1 with subnet mask 255.255.0.0. The default LAN IP address is 192.168.0.1 with subnet mask 255.255.255.0.
Q: Why is it that I can ping to outside hosts, but not access Internet Web sites?
A: Check the DNS server settings on your PC. You should get the DNS servers settings from your ISP. If your PC is running a DHCP client, remove any DNS IP address setting. As the router will assign the DNS settings to the DHCP-client-enabled PC.
Q: 5. What is the maximum number of IP addresses that the DHCP server of the gateway can assign to local PCs?
A: The built-in DHCP server can support 253 IP addresses for local network usage.

FAQ 1: Firmware upgrade Requirement and Process

1. Environment Requirement

a) A PC with FTP Server (Server-U software)

b) A PC or Notebook witch connected to LAN port of gateway.

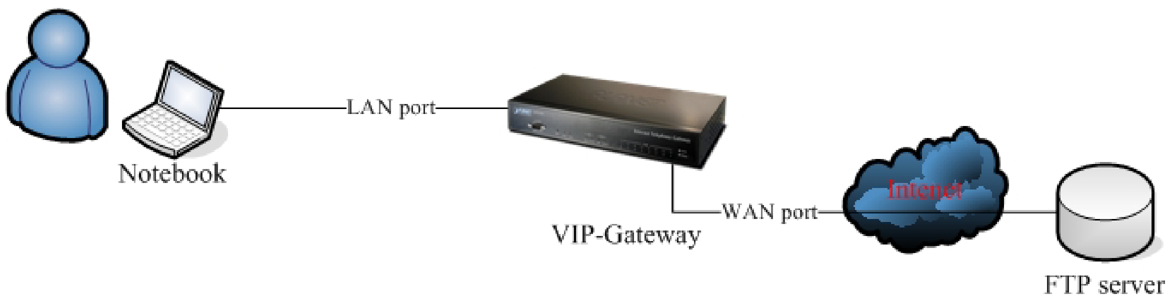
c) Put the image (firmware) named "**FW-VIP880_vxxx.bin** " at the assigned folder in FTP Server.

For example: "**FW-VIP880_v282.bin**" is version **2.8.2L**

Note: Free FTP server: 172.16.0.101

username: xxxx, password: xxxx

Environment Architecture (Gateway and FTP server are in Internet):



2. Upgrading Process

a) Notebook Telnet VoIP GW -> open DOS mode ->C:> telnet 192.168.0.1 (Default LAN port IP)

b) Please insert login password: 123, and select [4] Upgrade Software

```
 Login :

Welcome to 8 Port 4FXS+4FXO VoIP Gateway (version 2.8.2)
==========================================================
                       Main Menu
==========================================================
WAN Status:Fixed IP (NAT Mode)
VoIP Status:SIP Direct Mode
==========================================================
 [1] Advanced Setup.
 [2] System Administration.
 [3] Save Current Configurations.
 [4] Upgrade Software.
 [5] Ping.
 [6] Logout.
 [7] Restart.
 Please Select 1 - 7:
```

c) Please input IP address of FTP server like as: 172.16.0.101, username: xxxx, passswd: xxxx, and image name: **FW-VIP880_v282.bin**

d) Upgrade (y/n): **y**, then will write the firmware to flash.

e) After writing flash, Please reboot the Gateway.

f) If the new firmware (image) was most different with the previous version, please push the hardware reset bottom to set to default.

g) If the VoIP Gateway is in remote site, please use WEB configuration to set to default.

```
Starting the file transfer
############################################
1311648 bytes received in 2304 ms, (569.29Kbytes/sec), transfer succeeded
[5] Socket closed.
226 File sent ok.
[3] Socket closed.
Upgrade(y/n):y

Writing...

 Image size = 1311648, Written size = 1311648
 Write successfully.

 Don't forget to restart the system !
```

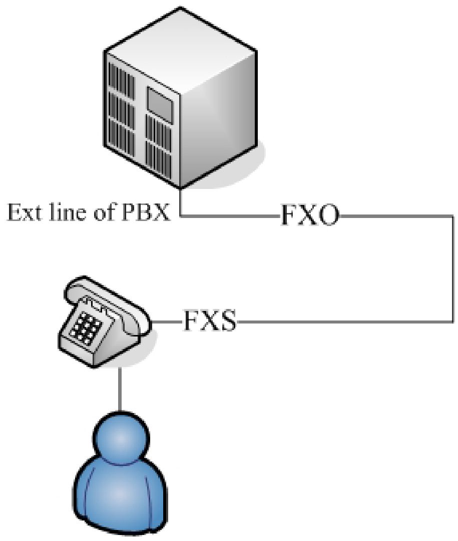
FAQ 2: Busy Tone Learning

STEP 1: Let the FXO port connect to PBX ext.

STEP 2: To dial to the FXO port from PBX another ext.

STEP 3: Hear dial tone, please dial FXS port number. When FXS port ring, please hang up the phone.

STEP 4: At this moment, hear busy tone and also press “y” to learning.




```
1.WAN Setting
2.LAN Setting
3.Virtual Server
4.Dynamic DNS
5.Network Management
6.VoIP Basic
7.Dialing Plan
8.VoIP Advance Setting
9.Hot Line Setting
a.Port Status
b.Busy Tone Learning
c.Show DNS mapping
Select 1-c:b

Learned busy tone on pattren  = (257(min), 292(max))
Learned busy tone off pattern = (222(min), 249(max))
Step 1:Please Dial from Line Port into gateway and then
       - dial the Phone Port number to ring the phoneset.
Step 2:Please OnHook the PBX extension.
Step 3:Press 'y' to start the busy tone auto-learning :
```

FAQ 3: FXO Ringer Voltage Threshold / Ringer Voltage Filter Setting

VIP-Gateway provides ring detector in FXO device avoiding can not answer and always OFF-HOOK status. This ring detector provides two functions to meet the various PBX's extension port:

1. FXO ringer voltage threshold

2. FXO ringer voltage filter

FXO Ringer Voltage Threshold

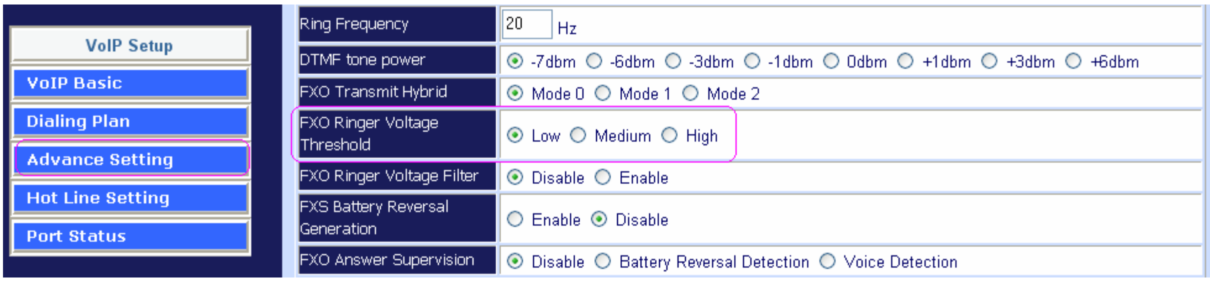
These three settings enable satisfaction of global ringer threshold requirements:

Low : 15V ± 10%

Medium: 21V ± 10%

High : 45V ± 10%

Thresholds are set so that a signal is guaranteed to not be detected below the minimum, and a ringer signal is guaranteed to be detected above the maximum.

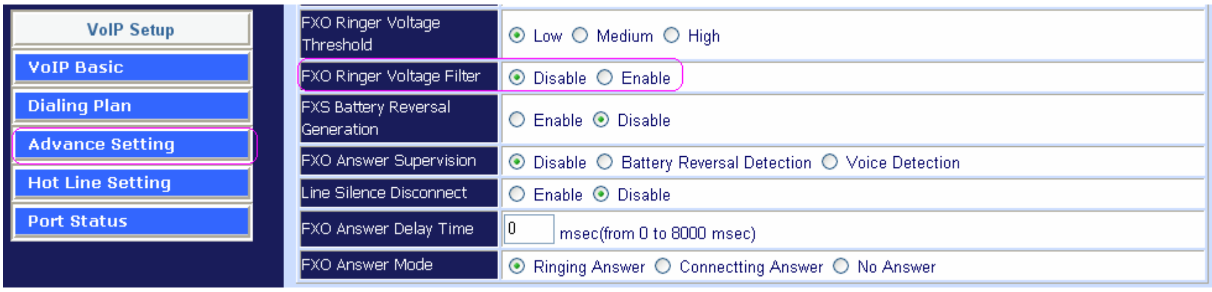


FXO Ringer Voltage Filter

Some vendor's PBX generates the leakage voltage from extension port.

That will mislead the FXO become Off-hook status.

This function was set to avoiding a leakage voltage signal is detected as ring coming.



FAQ 4: Answer Supervision

This chapter document is designed to help explain and resolve issues of answer supervision from a switch or PSTN provider that could result in billing for termination calls.

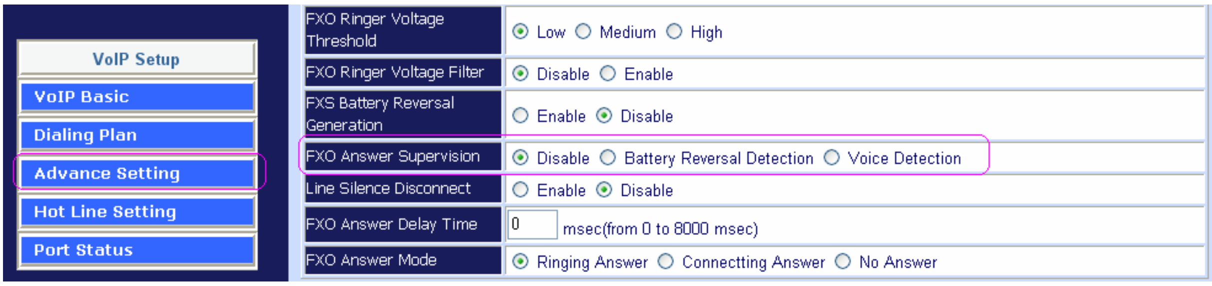
VIP-Gateway provides 2 Types of Answer Supervision:

1. Loop-Start Reverse Battery:

Reverse battery (also called Polarity Reverse) is when the PSTN provider reverses the polarity of the battery voltage, for both answer supervision and disconnects supervision.

2. Voice Detection:

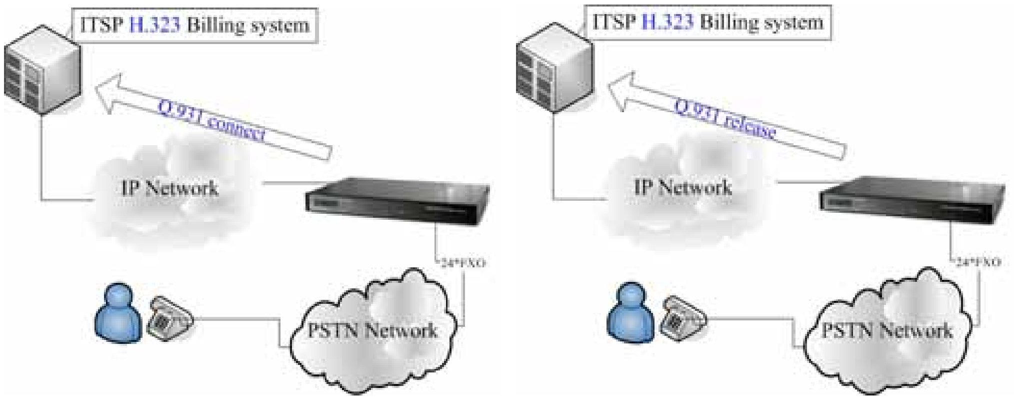
Voice Detection-based answer supervision is a feature where the Gateway can be configured to “listen” on the line for different tones and voice. The Gateway sends a “connect” signals out or “disconnect” signaling using internet.



H.323 scenario description: Loop Start Reverse Battery → PSTN line was set polarity reverse

a) The gateway can send the “Q.931 connect” H.323 signals to Billing System of ITSP, after the user pick up the Phone and detect the PSTN line answer voltage.

b) The gateway can send the “Q.931 Release” H.323 signals to Billing System of ITSP, after the user hang up the Phone and detect PSTN line disconnect voltage.



Scenario description: Voice Detection based on answer supervision

PSTN Line was not support Polarity Reverse:

a) The gateway can send the Q.931 connect H.323 signals to Billing System of ITSP, after the user pick up the Phone and detect the voice.

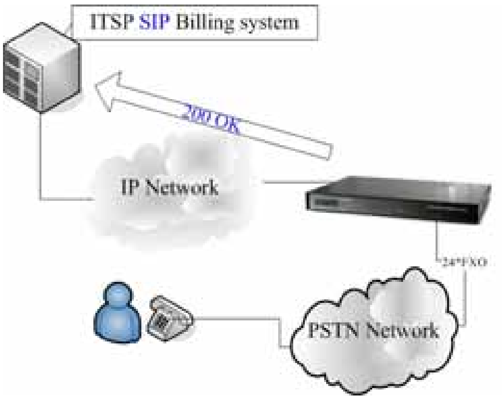
b) The gateway can send the Q.931 Release H.323 signals to Billing System of ITSP, after the user hang up the phone and detect the hang up voice.

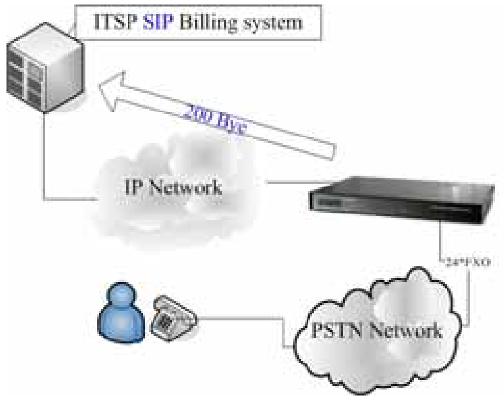
c) This type of answer supervision is not 100% accurate. Any voice frequency is detected as connect, including any intercept or recorded messages.

SIP scenario description: Loop Start Reverse Battery → PSTN line was set polarity reverse

a) The gateway can send the “200 OK” SIP signals to Billing System of ITSP, after the user pick up the Phone and detect the PSTN line answer voltage.

b) The gateway can send the “200 BYE” SIP signals to Billing System of ITSP, after the user hang up the Phone and detect PSTN line disconnect voltage.





Scenario description: Voice Detection based on answer supervision

PSTN Line was not support Polarity Reverse:

a) The gateway can send the 200 OK SIP signals to Billing system of ITSP, after the user pick up the Phone and detect the voice.

b) The gateway can send the 200 BYE SIP signals to Billing system of ITSP, after the user hang up the phone and detect the hang up voice.

c) This type of answer supervision is not 100% accurate. Any voice frequency is detected as connect, including any intercept or recorded messages.

FAQ 5: FXO Answer Mode Setting

FXO Answer Mode Concept: When user calls the PSTN line which was connected with the FXO port, there are three answer modes for user to configure.

Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line.

Connecting Answer Mode:

Scenario A:

“Hot Line Number” was NOT assigned in the FXO port and the FXO answer the call once the rings come from PSTN line.

Scencario B:

“Hot Line Number” was assigned and the hot line number belongs to remote VoIP device. In this scenario, the FXO port will not answer (off-hook) the PSTN till the user picks up the call.

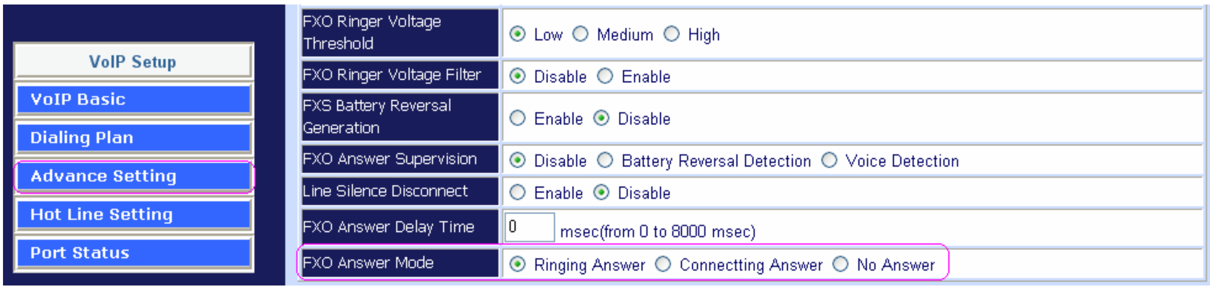
Note: This case can avoid charging for the Local PSTN call when the remote VoIP devices still ring.)

Scenario C:

“Hot Line Number” was setting and the hot line number was assigned to another FXS port in same gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user. **Note:** This case can avoid the Local PSTN charge when the FXS port still ring.)

Non Answer Mode: FXO will NOT answer the call in any time.

Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination



SIP Call Connecting Answer Mode

Scenario B description:

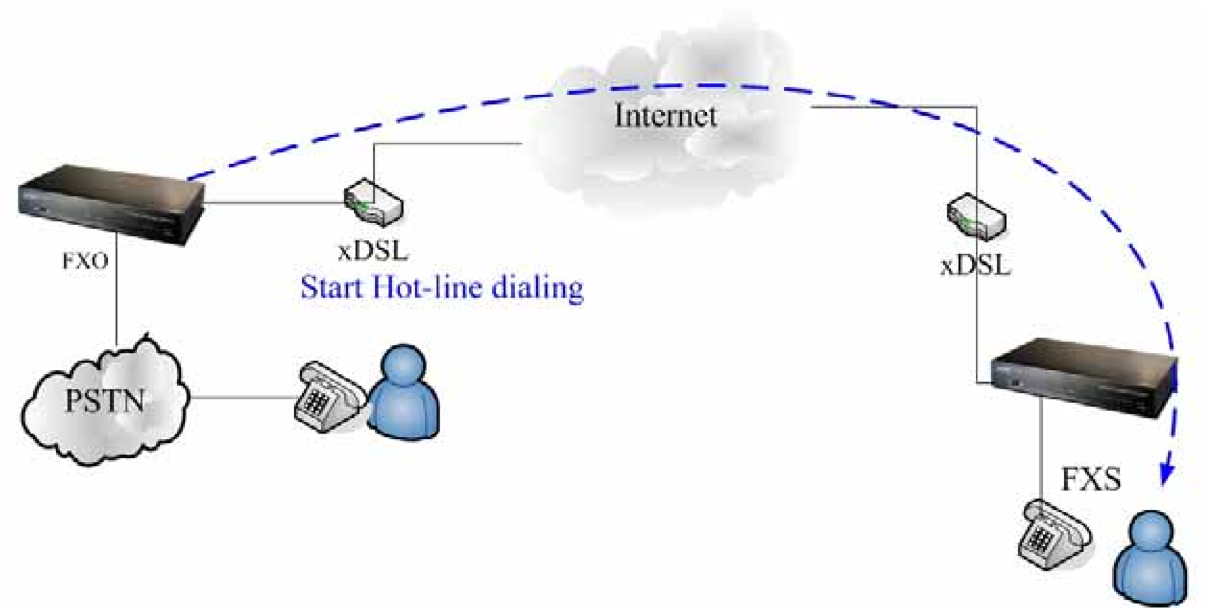
Hot Line Number" was assigned and the hot line number belongs to SIP device.

a) When the call com from PSTN to FXO, FXO start the Hot line dialing to remote SIP gateway

b) The phone of remote SIP gateway start ring.

c) When the phone was picked up, the remote SIP Gateway sends "SIP 200 OK" signal to FXO port.

d) Once FXO port receives the "SIP 200 OK" signal, FXO port would off-hook to answer the PSTN call.



Scenario C description:

"Hot Line Number" was setting and the hot line number was assigned to another FXS port in same gateway.

a) When the call com from PSTN to FXO, FXO start the Hot line dialing to FXS port.

b) The phone start ring.

c) Once the phone was picked up, FXO port would off-hook to answer the PSTN call.

H.323 Call Connecting Answer Mode

Scenario B description:

Hot Line Number" was assigned and the hot line number belongs to remote H.323 device.

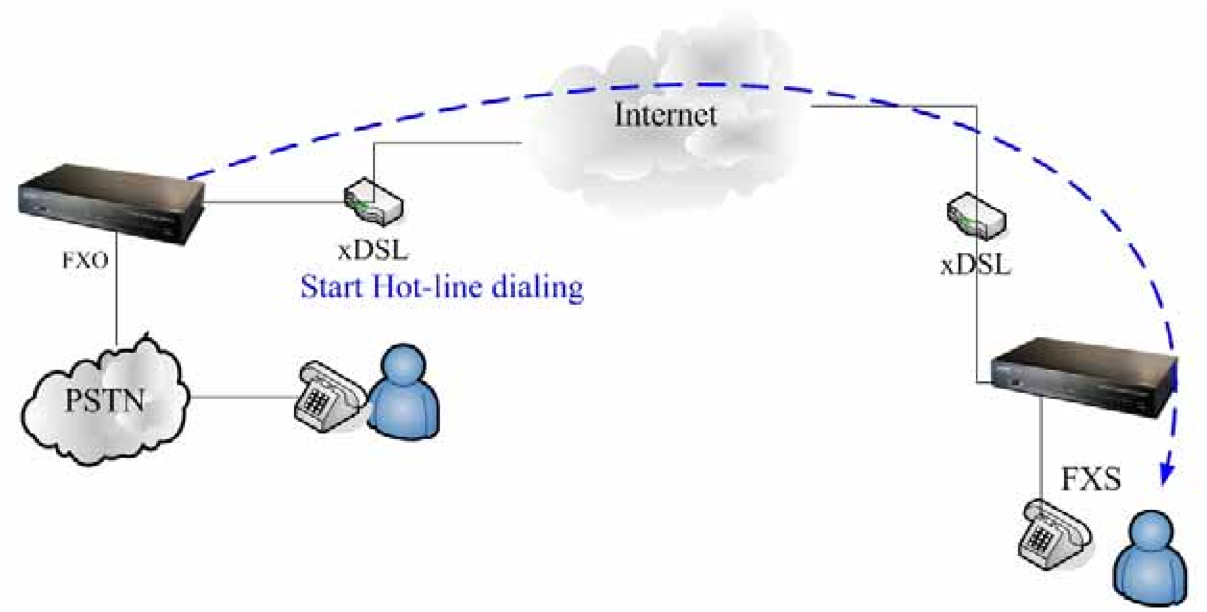
Note: The remote H.323 device need disable the "Auto Answer"

a) When the call com from PSTN to FXO, FXO start the Hot line dialing to remote H.323 gateway

b) The phone of remote H.323 gateway start ring.

c) When the phone was picked up, the remote H.323 Gateway send "Q.931 connects" signal to FXO port.

Once FXO port receives the "Q.931 connects" signal, FXO port would off-hook to answer the PSTN call.



Scenario C description: "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway.

a) When the call com from PSTN to FXO, FXO start the Hot line dialing to FXS port.

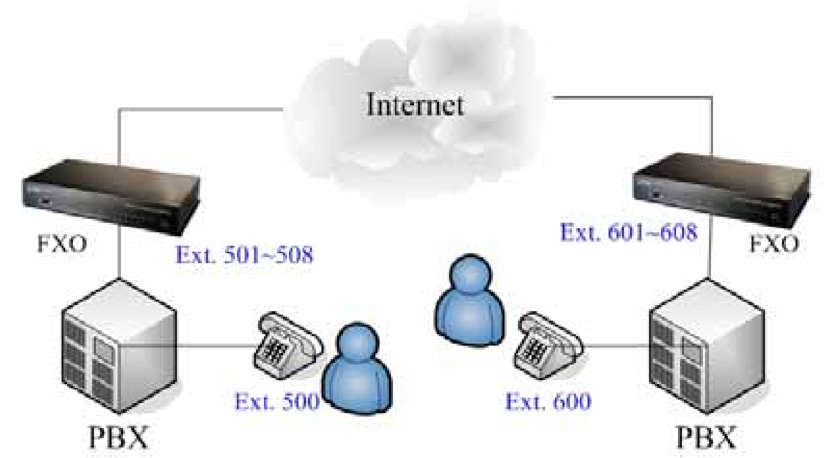
b) The phone start ring.

c) Once the phone was picked up, FXO port would off-hook to answer the PSTN call.

FAQ 6: Peer to Peer call: FXO to FXO

Scenario description:

User (500) on site A (VIP-880FO) wishes to have telephone calls to extension (600) on Site B (VIP-880FO B). User (500) on site B (VIP-880FO B) can connect to ser A in the same way.



IP address of VIP-880FO_A is: **172.16.0.1**

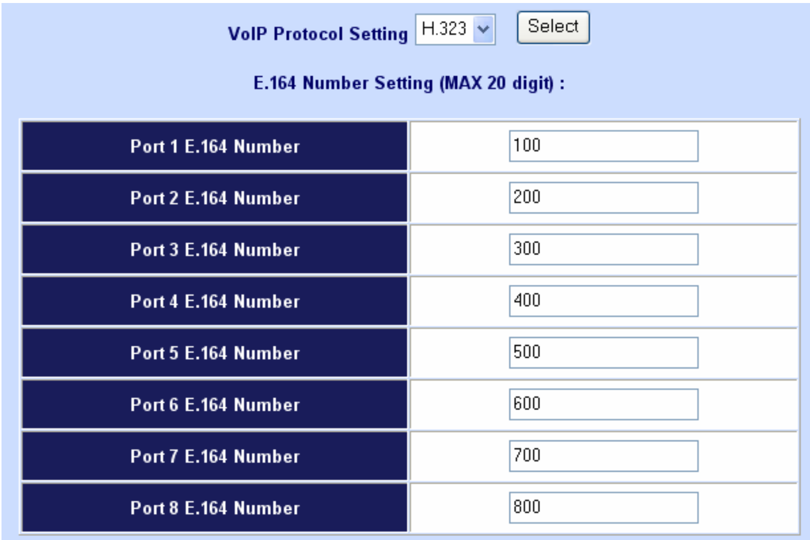
VIP-880FO_A number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**

IP address of VIP-880FO_B is: **172.16.0.2**

VIP-880FO_B number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**



VoIP Protocol Setting H.323 Select

E.164 Number Setting (MAX 20 digit) :

Port 1 E.164 Number	100
Port 2 E.164 Number	200
Port 3 E.164 Number	300
Port 4 E.164 Number	400
Port 5 E.164 Number	500
Port 6 E.164 Number	600
Port 7 E.164 Number	700
Port 8 E.164 Number	800

The dial plan of VIP-880FO_A dial plan setting: that means call **0xxx** leading number go to IP address 172.16.0.2 gateway (VIP-880FO_B).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.2	
		~				ADD

DELETE Outbound Dial Plan From To

The dial plan of VIP-880FO_B dial plan setting: that means call **0xxx** leading number go to IP address 172.16.0.1 gateway (VIP-880FO_A).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.1	
		~				ADD

DELETE Outbound Dial Plan From To

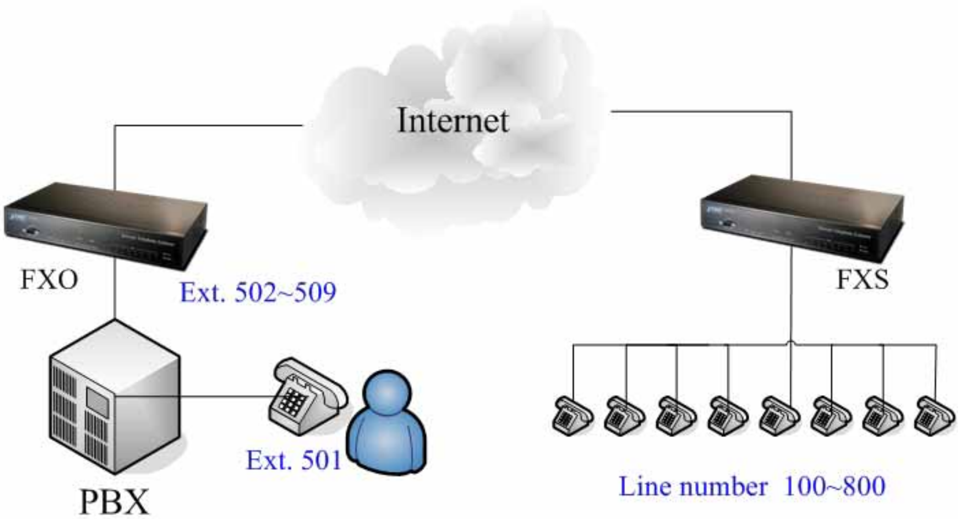
Usage:

The ext.509 dial to ext 501 (connect to FXO port 1) will hear the dial tone, and then dial **0100** go to IP address gateway 172.16.0.2 (VIP-880FO_B), and hear the dial tone again, then dial 609 ext, the ext.609 will ring.

FAQ 7: Peer to Peer call: FXO to FXS

Scenario description:

User (500) on site A (VIP-880FO) wishes to have telephone calls to phone number (100) on Site B (VIP-880FS).



IP address of VIP-880FO is: **172.16.0.1**

VIP-880FO number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**

IP address of VIP-880FS is: **172.16.0.2**

VIP-880FS number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**

VoIP Protocol Setting H.323 Select

E.164 Number Setting (MAX 20 digit) :

Port 1 E.164 Number	100
Port 2 E.164 Number	200
Port 3 E.164 Number	300
Port 4 E.164 Number	400
Port 5 E.164 Number	500
Port 6 E.164 Number	600
Port 7 E.164 Number	700
Port 8 E.164 Number	800

The dial plan of VIP-880FO dial plan setting: that means call **0xxx** leading number go to IP address 172.16.0.2 gateway (VIP-880FS).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.2	
		~				ADD

DELETE Outbound Dial Plan From To

The dial plan of VIP-880FS dial plan setting: that means call **0xxx** leading number go to IP address 172.16.0.1 gateway (VIP-880FO).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.1	
		~				ADD

DELETE Outbound Dial Plan From To

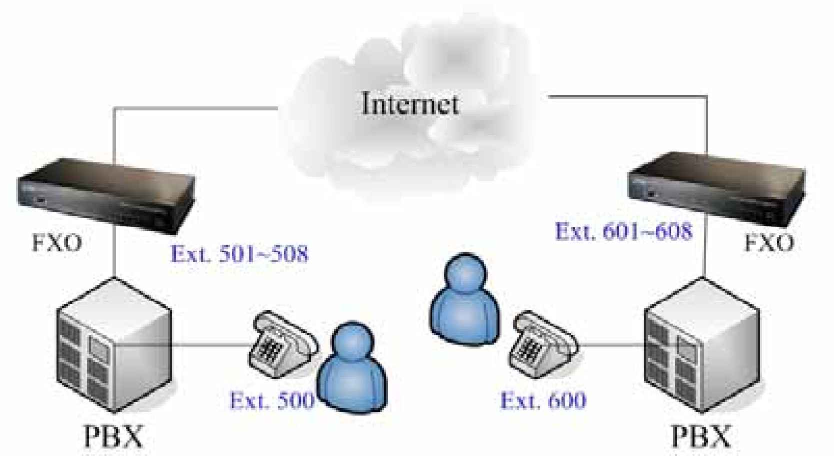
Usage:

The ext.509 dial to ext 501 (connect to FXO) will hear the dial tone, and then dial **0100** go to IP address gateway 172.16.0.2 (VIP-880FS), and phone of port 1(100) will ring.

FAQ 8: Peer to Peer call for one shoot dialing: FXO to FXO

Scenario description:

User (500) on site A (VIP-880FO) wishes to have telephone calls to extension (600) on Site B (VIP-880FO B). User (500) on site B (VIP-880FO B) can connect to ser A in the sam way.



IP address of VIP-880FO_A is: **172.16.0.1**

VIP-880FO_A number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**

IP address of VIP-880FO_B is: **172.16.0.2**

VIP-880FO_B number and dial plan setting:

Each port number is **100,200,300,400,500,600,700,800**

VIP-880FO_A Dial Plan setting:

The Incoming Call Dial Plan of VIP-880FO_A: that means incoming call **50x** leading number will **hunt port 1 to port 8**

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	50x	3 ~ 3	0	None	1,2,3,4,5,6,7,8		
		~					ADD

DELETE Inbound Dial Plan From To

The dial plan of VIP-880FO_A: that means call **0xxx** leading number will go to IP address 172.16.0.2 gateway (VIP-880FO_B).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.2	
		~				ADD
DELETE Outbound Dial Plan	From	To				

VIP-880FO_B number and dial plan setting:

The dial plan of VIP-880FO_B dial plan setting: that means incoming call **60x** leading number will **hunt port 1 to port 8**

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Register to GK	Operation
1	60x	3 ~ 3	0	None	1,2,3,4,5,6,7,8	☐	
		~				☐	ADD
DELETE Inbound Dial Plan	From	To					

The dial plan of VIP-880FO_B: That means call **0xxx** leading number will go to IP address 172.16.0.1 gateway (VIP-880FO_A).

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	0	4 ~ 4	1	None	172.16.0.1	
		~				ADD
DELETE Outbound Dial Plan	From	To				

Usage:

The ext.500 dial to ext 501 (connect to FXO) will hear the dial tone, and then dial **0100** go to IP address gateway 172.16.0.2, the ext.600 will ring.

FAQ 9: Peer to Peer call: Hotline setting

Hot line Basic Concept:

Any number set in Hot line field will be dialed by VoIP call automatically.

For FXS port case: When user picks up the phone, the gateway will dial the hot line number to internet by VoIP call.

For FXO port case: When the FXO off-hook (PSTN call coming or PBX extension ring in), the gateway will dial the hot line number to internet by VoIP call.

Scenario description: Peer to Peer direct call via SIP or H.323 mode

STEP 1: To set the outgoing call dial plan in gateway, for example the number "911" call to gateway which's the IP address is 172.16.0.119.

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	911	3 ~ 3	0	None	172.16.0.119	
		~				ADD

DELETE	Outbound Dial Plan	From To

STEP 2: To set hot line number in Hot Line Setting

Hot Line Number Setting (Hotline Setting)

Port 1 number	911
Port 2 number	None
Port 3 number	None
Port 4 number	None
Port 5 number	None
Port 6 number	None
Port 7 number	None
Port 8 number	None

STEP 3: When users pick up the phone (port1), the gateway will dial the "911" to the gateway (IP address: 172.16.0.119)

Scenario description: Register to SIP proxy server/H.323 Gatekeeper direct call

STEP 1: Let your VIP-GW register to SIP proxy or H.323 Gatekeeper server

STEP 2: To set hot line number in Hot Line Option

Hot Line Number Setting (Hotline Setting)

Port 1 number	911
Port 2 number	None
Port 3 number	None
Port 4 number	None
Port 5 number	None
Port 6 number	None
Port 7 number	None
Port 8 number	None

STEP 3: When users pick up the phone (port1), the gateway will dial the "911" to SIP proxy or H.323 Gatekeeper server (ITSP)

Scenario description: Register to SIP Proxy / H.323 Gatekeeper server and Peer to Peer direct call first

STEP 1: Let your VIP-GW register to SIP proxy or H.323 Gatekeeper server

STEP 2: To set the outgoing call dial plan in gateway, for example the "911" will call to gateway which's the IP address is 172.16.0.119.

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Operation
1	911	3 ~ 3	0	None	172.16.0.119	
		~				ADD

DELETE	Outbound Dial Plan	From	To

STEP 3: To set hot line number in Hot Line Setting

Hot Line Number Setting (Hotline Setting)

Port 1 number	911
Port 2 number	None
Port 3 number	None
Port 4 number	None
Port 5 number	None
Port 6 number	None
Port 7 number	None
Port 8 number	None

STEP 4: When users pick up the phone (port 1), the gateway will dial the “911” to the gateway which’s IP address 172.16.0.119.

Note: This call will not call to SIP Proxy or H.323 Gatekeeper server because of direct call first.

FAQ 10: SIP speed call setting

Speed calls Concept:

Cut your phone number down to fewer digit dialing!

Life is moving fast - you've got to dial fast. Now you can with Speed Dial. Dial the people you call most with just dialing fewer digits instead of dialing the full phone number.

What's even better is that you can customize and manage your speed dial phone numbers in Dial Plan Setting on your gateway!   Dial Plan allows you to set up to speed dial numbers that can be called with the fewer numbers.

Scenario description A: User wants to dial any number instead of 810-any number

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	x	2 ~ 15	0	810	sip.test.com	5060	
		~					ADD

DELETE Outbound Dial Plan From To

Note: The destination IP address is the domain name of SIP proxy server

Scenario description B: User wants to dial 86-1234567890 instead of 810-86-1234567890

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	86x	3 ~ 15	0	810	sip.test.com	5060	
		~					ADD

DELETE Outbound Dial Plan From To

Note: The destination IP address is the domain name of SIP proxy server

Scenario description C: User wants to dial 888 instead of 810-861234567890

Outgoing Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit)

Item	Outgoing no.	Length of Number	Delete Length	Prefix no.	Destination IP/DNS	Destination Port	Operation
1	999	3 ~ 3	3	8100861234567890	sip.test.com	5060	
		~					ADD

DELETE Outbound Dial Plan From To

Note: The destination IP address is the domain name of sip proxy server

Appendix C

VIP-281 series Specifications

Product	2-Port H.323/SIP VoIP Gateway	
Model	VIP-281	VIP-281FS
Hardware		
WAN	1 x 10/100Mbps RJ-45 port	
LAN	4 x 10/100Mbps RJ-45 port	
Voice	2 x RJ-11 connection (1 x FXS, 1 x FXO)	2 x RJ-11 connection (2 x FXS)
Protocols and Standard		
Standard	H.323 v2/v3/v4 and SIP (RFC 3261) , SDP (RFC 2327), Symmetric RTP, STUN (RFC3489), ENUM (RFC 2916), RTP Payload for DTMF Digits (RFC2833), Outbound Proxy Support.	
Voice codec	G.711(A-law /u-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)	
Fax support	T.30, T.38	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.165/G.168 Echo cancellation Dynamic Jitter Buffer	
Protocols	SIP 2.0 (RFC-3261), H.323, TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS	
Advanced Function	Virtual Server, Smart QoS, IP TOS (IP Precedence) / DiffServ, Build-in NAT router function.	
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Management	Web, Telnet	
LED Indications	System: 1, PWR WAN: 1, LNK/ACT LAN: 4, LNK/ACT Voice 2, In-Use/Ringing	
Dimension (W x D x H)	260 x 135 x 35 mm	
Operating Environment	0~40 degree C, 0~95% humidity	
Power Requirement	12V DC	
EMC/EMI	CE, FCC Class B	

VIP-480 series Specifications

Product	4-Port H.323/SIP VoIP Gateway		
Model	VIP-480	VIP-480FS	VIP-480FO
Hardware			
WAN	1 x 10/100Mbps RJ-45 port		
LAN	4 x 10/100Mbps RJ-45 port		
Voice	4 x RJ-11 connection (2 x FXS, 2 x FXO)	4 x RJ-11 connection (4 x FXS)	4 x RJ-11 connection (4 x FXO)
Protocols and Standard			
Standard	H.323 v2/v3/v4 and SIP (RFC 3261) , SDP (RFC 2327), Symmetric RTP, STUN (RFC3489), ENUM (RFC 2916), RTP Payload for DTMF Digits (RFC2833), Outbound Proxy Support.		
Voice codec	G.711(A-law /u-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)		
Fax support	T.30, T.38		
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.165/G.168 Echo cancellation Dynamic Jitter Buffer		
Protocols	SIP 2.0 (RFC-3261), H.323, TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS		
Advanced Function	Virtual Server, Smart QoS, IP TOS (IP Precedence) / DiffServ, Build-in NAT router function.		
Network and Configuration			
Access Mode	Static IP, PPPoE, DHCP		
Management	Web, Telnet		
LED Indications	System: 1, PWR WAN: 1, LNK/ACT LAN: 4, LNK/ACT Voice 4, In-Use/Ringing		
Dimension (W x D x H)	260 x 135 x 35 mm		
Operating Environment	0~40 degree C, 0~95% humidity		
Power Requirement	12V DC		
EMC/EMI	CE, FCC Class B		

VIP-880 series Specifications

Product	8-Port H.323/SIP VoIP Gateway		
Model	VIP-880	VIP-882	VIP-880FO
Hardware			
WAN	1 x 10/100Mbps RJ-45 port		
LAN	1 x 10/100Mbps RJ-45 port		
Voice	8 x RJ-11 connection (4 x FXS, 4 x FXO)	8 x RJ-11 connection (6 x FXS, 2 x FXO)	8 x RJ-11 connection (8 x FXO)
Protocols and Standard			
Standard	H.323 v2/v3/v4 and SIP (RFC 3261) , SDP (RFC 2327), Symmetric RTP, STUN (RFC3489), ENUM (RFC 2916), RTP Payload for DTMF Digits (RFC2833), Outbound Proxy Support.		
Voice codec	G.711(A-law /μ-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)		
Fax support	T.30, T.38		
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.165/G.168 Echo cancellation Dynamic Jitter Buffer		
Protocols	SIP 2.0 (RFC-3261), H.323, TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS		
Advanced Function	Virtual Server, Smart QoS, IP TOS (IP Precedence) / DiffServ, Build-in NAT router function.		
Network and Configuration			
Access Mode	Static IP, PPPoE, DHCP		
Management	Web, RS-232 Console, Telnet		
LED Indications	System: 2, PWR, CPU WAN: 2, LNK/ACT LAN: 2, LNK/ACT Voice 8, In-Use/Ringing		
Dimension (W x D x H)	300 x 160 x 40 mm		
Operating Environment	0~40 degree C, 0~95% humidity		
Power Requirement	12V DC		
EMC/EMI	CE, FCC Class B		

VIP-1680 series Specifications

Product	16-Port H.323/SIP VoIP Gateway	
Model	VIP-1680	VIP-1680FO
Hardware		
WAN	1 x 10/100Mbps RJ-45 port	
LAN	1 x 10/100Mbps RJ-45 port	
Voice	1 x RJ-21 connector for connecting to telephone patch panel (8 x FXS, 8 x FXO)	1 x RJ-21 connector for connecting to telephone patch panel (16 x FXO)
Protocols and Standard		
Standard	H.323 v2/v3/v4 and SIP (RFC 3261) , SDP (RFC 2327), Symmetric RTP, STUN (RFC3489), ENUM (RFC 2916), RTP Payload for DTMF Digits (RFC2833), Outbound Proxy Support.	
Voice codec	G.711(A-law /u-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)	
Fax support	T.30, T.38	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.165/G.168 Echo cancellation Dynamic Jitter Buffer	
Protocols	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS	
Advanced Function	Virtual Server, Smart QoS, IP TOS (IP Precedence) / DiffServ, Build-in NAT router function.	
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Management	Web, Telnet, Console	
LED Indications	System: 2, PWR/CPU WAN: 2, LNK/ACT LAN: 2, LNK/ACT Voice 16, In-Use/Ringing	
Dimension (W x D x H)	440 x 250 x 44 mm	
Operating Environment	0~40 degree C, 0~95% humidity	
Power Requirement	100~240V AC 50/60Hz	
EMC/EMI	CE, FCC Class B	

VIP-2480 series Specifications

Product	24-Port H.323/SIP VoIP Gateway	
Model	VIP-2480	VIP-2480FO
Hardware		
WAN	1 x 10/100Mbps RJ-45 port	
LAN	1 x 10/100Mbps RJ-45 port	
Voice	1 x RJ-21 connector for connecting to telephone patch panel (12 x FXS, 12 x FXO)	1 x RJ-21 connector for connecting to telephone patch panel (24 x FXO)
Protocols and Standard		
Standard	H.323 v2/v3/v4 and SIP (RFC 3261) , SDP (RFC 2327), Symmetric RTP, STUN (RFC3489), ENUM (RFC 2916), RTP Payload for DTMF Digits (RFC2833), Outbound Proxy Support.	
Voice codec	G.711(A-law /u-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)	
Fax support	T.30, T.38	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.165/G.168 Echo cancellation Dynamic Jitter Buffer	
Protocols	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS	
Advanced Function	Virtual Server, Smart QoS, IP TOS (IP Precedence) / DiffServ, Build-in NAT router function.	
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Management	Web, Telnet, Console	
LED Indications	System: 2, PWR/CPU WAN: 2, LNK/ACT LAN: 2, LNK/ACT Voice 24, In-Use/Ringing	
Dimension (W x D x H)	440 x 250 x 44 mm	
Operating Environment	0~40 degree C, 0~95% humidity	
Power Requirement	100~240V AC 50/60Hz	
EMC/EMI	CE, FCC Class B	