PLANET
Networking & Communication


# H.323/SIP VoIP GSM Gateway

## VIP-281GS

## User's manual

**Version 1.1.0**

# Copyright



The products and programs described in this User's Manual are licensed products of PLANET Technology, This User's Manual contains proprietary information protected by copyright, and this User's Manual and all accompanying hardware, software, and documentation are copyrighted.

No part of this User's Manual may be copied, photocopied, reproduced, translated, or reduced to any electronic medium or machine-readable form by any means by electronic or mechanical. Including photocopying, recording, or information storage and retrieval systems, for any purpose other than the purchaser's personal use, and without the prior express written permission of PLANET Technology.

# 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 H.323/SIP VoIP GSM Gateway:

Model: VIP-281GS

Rev: 1.1 (October, 2009)

Part No. EM-VIP281GSV1.1

## TABLE OF CONTENTS

# Chapter 1
# Introduction

## Overview

With years of Internet telephony and router manufacturing experience, PLANET proudly introduces the The PLANET VIP-281GS VoIP GSM Gateway is a signal-GSM channel gateway that supports SIP and H.323 VoIP protocol at the same time. The VIP-281GS provides a total solution for integrating voice-data network and the Global System for Mobile Communications (GSM).

The VIP-281GS is equipped with both FXS and PSTN interfaces, which gives the gateway a wide range of potential applications. The VIP-281GS can be installed on a PBX trunk line to enrich its trunks-GSM and VoIP routes. The PBX is able to have voice communication to either VoIP or GSM environment by the least costs.

Meanwhile, the VIP-281GS is designed for comfort, ease-of-use with a sophisticated and satisfaction to customers. The VIP-281GS not only inherits traditions of quality voice communications but the VIP-281GS also eliminates the human resource of VoIP network deployment. With optimized H.323/SIP architecture, the VIP-281SG is the ideal choices for P2P voice chat and ITSP cost-saving solution, but also provides network-converting feature to translate the packet network into traditional PBX system.

With built-in PPPoE/DHCP/DDNS clients, up to 2 concurrent connections in VIP-281GS, voice communications can be established from anywhere around the world. The VIP-281GS comes with intuitive user-friendly and powerful management interface (web/telnet), that can dramatically reduce IT personnel resource and complete GSM/VoIP deployment in a short time. Plus remote management capability, 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.

## Network Features

- **Point-to-Point Protocol over Ethernet (PPPoE) Client Support:**
  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 computers.
- **Smart QoS**
  The smart QoS provides stable voice quality while users access internet from private LAN to internet at the same time. This device would start suppressing throughput automatically

when VoIP call was proceeded and it 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 Virtual Servers.
- **NAT Traversal**
  The NAT traversal allows gateway to operate behind any NAT/Firewall device. There is no need to change any configuration of NAT/Firewall like setting virtual server.

## VoIP Features

- H.323 / SIP dual mode communication
- SIP 2.0 (RFC3261), H.323v4 compliant
- Peer-to-Peer / H.323 GK / SIP proxy calls
- PSTN lifeline support
- 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.
- Built-in adaptive buffer that helps to smooth out the variations of delay (jitter) in voice traffic.
- Voice channels status display: This function displays each port status such as on-hook, off-hook, calling number, talk duration, codec.

## GSM Features

- SMS Server for SMS sending and receiving
- Worldwide GSM network usable (850/900/1800/1900 MHz)
- Supports GSM PIN code protection

# Package Content

The contents of your product should contain the following items:

- ➢ Voice Gateway VIP-281GS unit
- ➢ Power adapter
- ➢ GSM Antenna
- ➢ Quick Installation Guide
- ➢ User's Manual CD
- ➢ RJ-45 cable x 1

## Physical Details

The following figure illustrates the front/rear panel of VIP-281GS series:

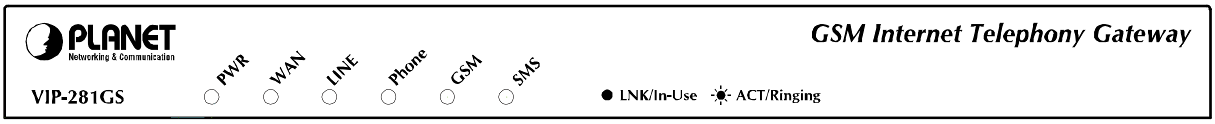



Figure 1-1. Front panel of VIP-281GS

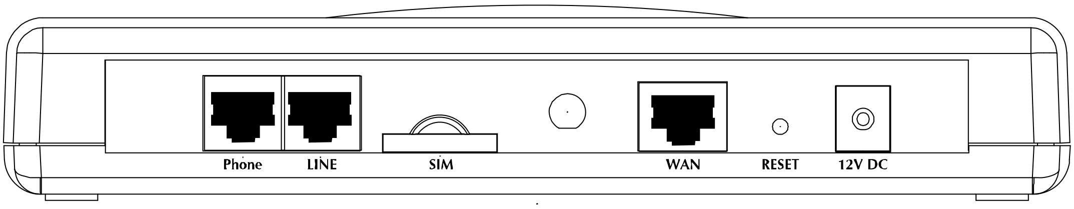



Figure 1-2. Rear panel of VIP-281GS

### Front Panel LED Indicators & Rear Panels

| Front Panel LED | State | Descriptions |
|---|---|---|
| **PWR** | On<br>Off | GSM GW is powered ON<br>GSM GW is powered Off |
| **WAN Port** | ON<br>Flashing<br>Off | Network connection established<br>Data traffic on cable network<br>Waiting for network connection |
| **Line** | ON<br>Flashing<br>Off | Line is busy<br>Ring Indication<br>Line is not enabled |
| **Phone** | On<br>Flashing<br>Off | Telephone Set is Off-Hook<br>Ring Indication<br>Telephone Set is On-Hook |
| **GSM** | On<br>Flashing | GSM Network is found and working properly<br>Searching GSM Network |
| **SMS** | On<br>Flashing | Short message waiting Indicator<br>Sending short message |

Table 1-1. Front panel description of VIP-281GS

**Note**

The Default WAN IP is http://172.16.0.1. Press RESET button on rear panel over 5 seconds will reset the VoIP GSM Gateway to this default LAN/WAN IP address and Username/Password function.

| Rear Panel | Descriptions |
|---|---|
| **Phone** | Phone port was connected to your telephone sets or Trunk Line of PBX. |
| **Line** | Can be Connected to PBX or CO line with RJ-11 analog line. PSTN not FXO port, can’t connect PSTN to VoIP,. When PSTN call comes, it will transfer to FXS port, let FXS can pick up call from VoIP or PSTN. |
| **SIM** | The port which you can Insert SIM Card |
| **Antenna Connector** | Connect the antenna to the gateway. |
| **WAN** | Connect to the network with an Ethernet cable. This port allows your ATA to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks. |
| **Reset** | Push this button until 3 seconds, and ATA will be set to factory default configuration. |
| **12V DC (Power)** | The supplied power adapter connects here. |

Table 1-2. Rear panel description of VIP-281GS

**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

## Physical Installation Requirement

This chapter illustrates basic installation of VIP-281GS 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-281GS provides GUI (Web based, Graphical User Interface) for machine management and administration.

### Web configuration access:

To start VIP-281GS web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0 or higher with Java support

Default WAN interface IP address of VIP-281GS is **172.16.0.1**. You may now open your web browser, and insert **http://172.16.0.1** in the address bar of your web browser to logon VIP-281GS web configuration page.

VIP-281GS will prompt for logon username/password, please enter: **admin / 123** to continue machine administration.


Figure 2-1. Login prompt of VIP-281GS

**Note**

Please locate your PC in the same network segment (172.16.0.x) of VIP-281GS. 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.

## 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:

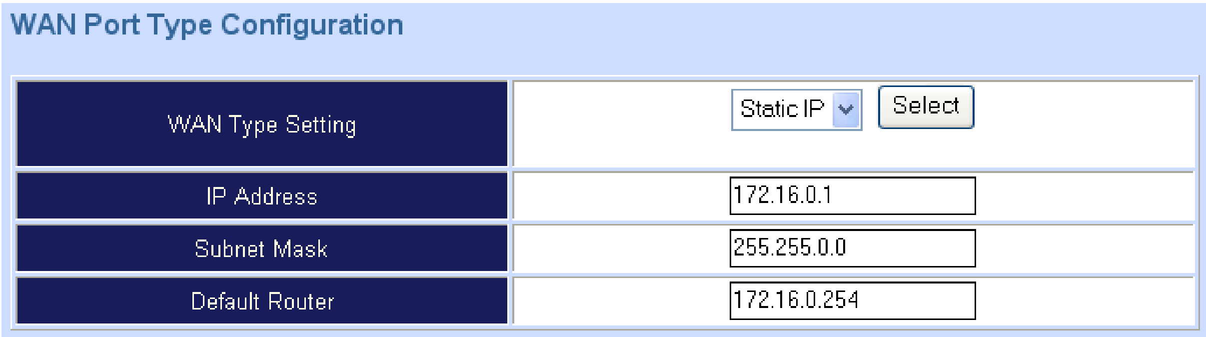



Figure 2-2. WAN port configuration

| **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, |

Table 2-1. WAN port configuration descriptions

**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.

# Chapter 3

# Network Service Configurations

## Configuring and monitoring your VoIP Gateway from web browser

The VIP-281GS 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 GSM 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-281GS web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0 or higher with Java support

## Manipulation of VoIP GSM Gateway via web browser

**Log on VoIP GSM Gateway via web browser**

After TCP/IP configurations on your PC, you may now open your web browser, and input ***http://172.16.0.1*** to logon VoIP GSM gateway web configuration page.

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


Figure 3-1. Login prompt of VIP-281GS


Figure 3-2. System configuration

# VIP-281GS Setup for Quick Start

**System Configuration**

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

## 1. Network Setup (WAN Port Type Setup)

For most users, Internet access is the primary application. The Gateway 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**" from within the **Advance Setup**, the following setup page will be show.

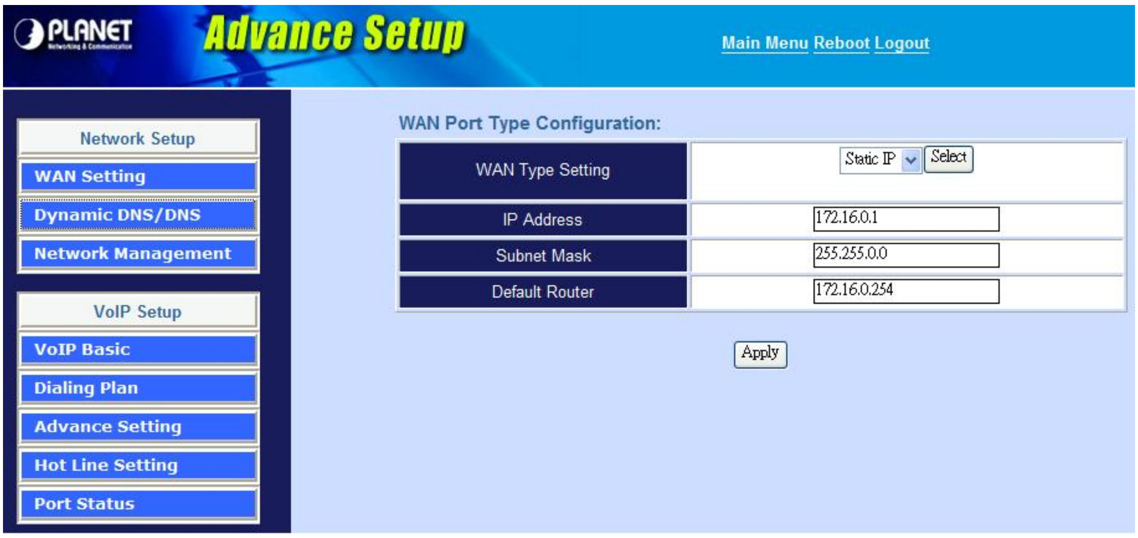



Figure 3-3. WAN setting

| 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 |

Table 3-1. WAN setting descriptions

**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.

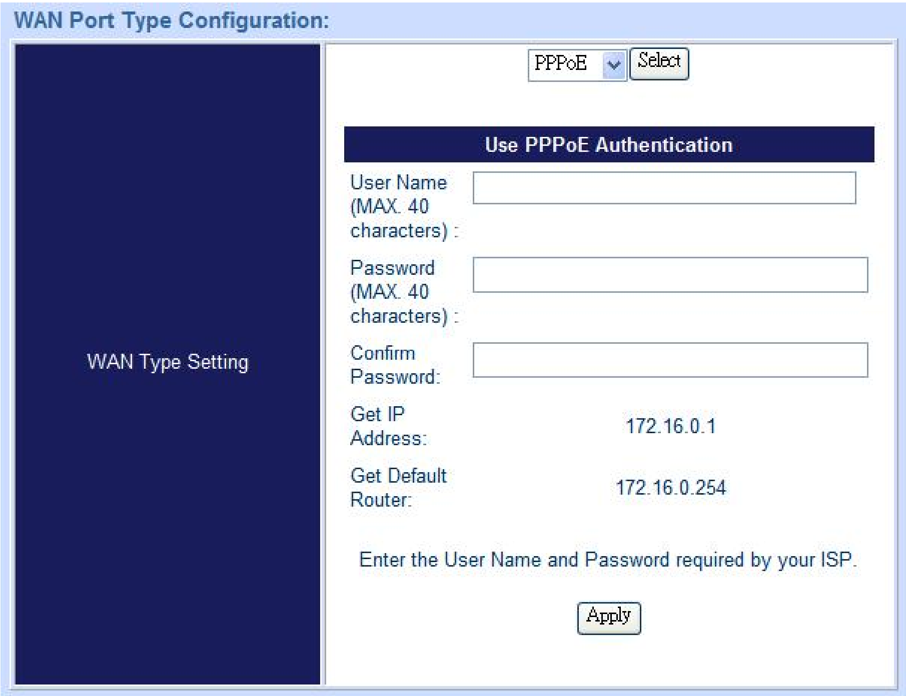



Figure 3-4. PPPoE enable setting

| Three methods are available for Internet Access | |
|---|---|
| **User Name** | Enter User Name provided by your ISP |
| **Password** | Enter Password provided by your ISP |
| **Confirm Password** | Enter Password to confirm again |

Table 3-2. PPPoE enable descriptions

**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.

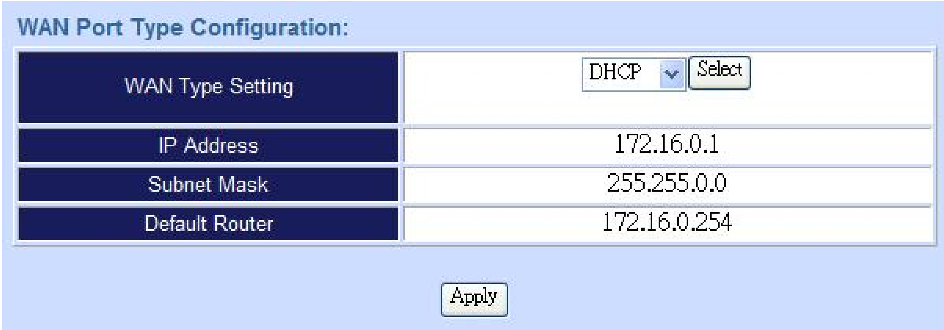



Figure 3-5. DHCP setting

## 2. VoIP Basic Setup:

**STEP1 : Configure VoIP Call Signal Protocols :**

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

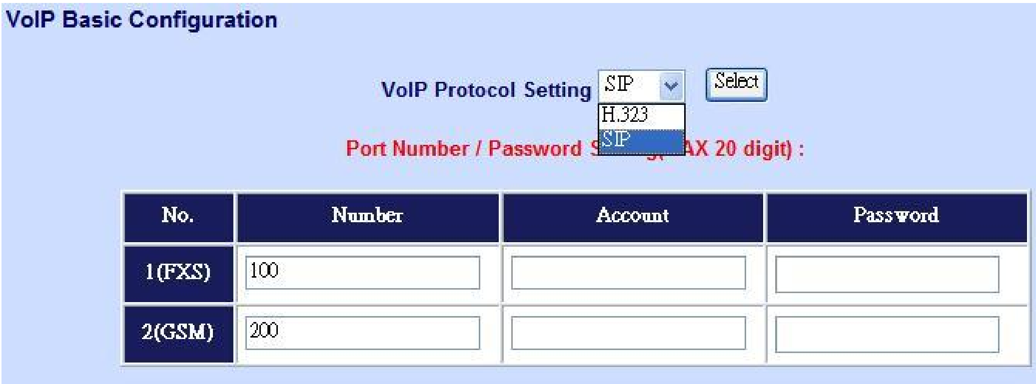



Figure 3-6. FXS/GSM number setting

**STEP2 : Configure the numbering with Phone(FXS)/GSM ports.**

| | |
|---|---|
| **FXS Number** | The representation number is the phone number of the telephone that is connected to Phone port |
| **GSM Number** | The representation number is the phone number of SIM CARD |

Table 3-3. FXS/GSM number descriptions

**STEP3: Let GW Register to Gatekeeper/SIP Proxy Server**

(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 gateway does not want to register to any SIP Proxy Server, just set value 0 .0.0.0 to the sip proxy server address. |

Table 3-4. Gatekeeper/SIP proxy descriptions

**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).

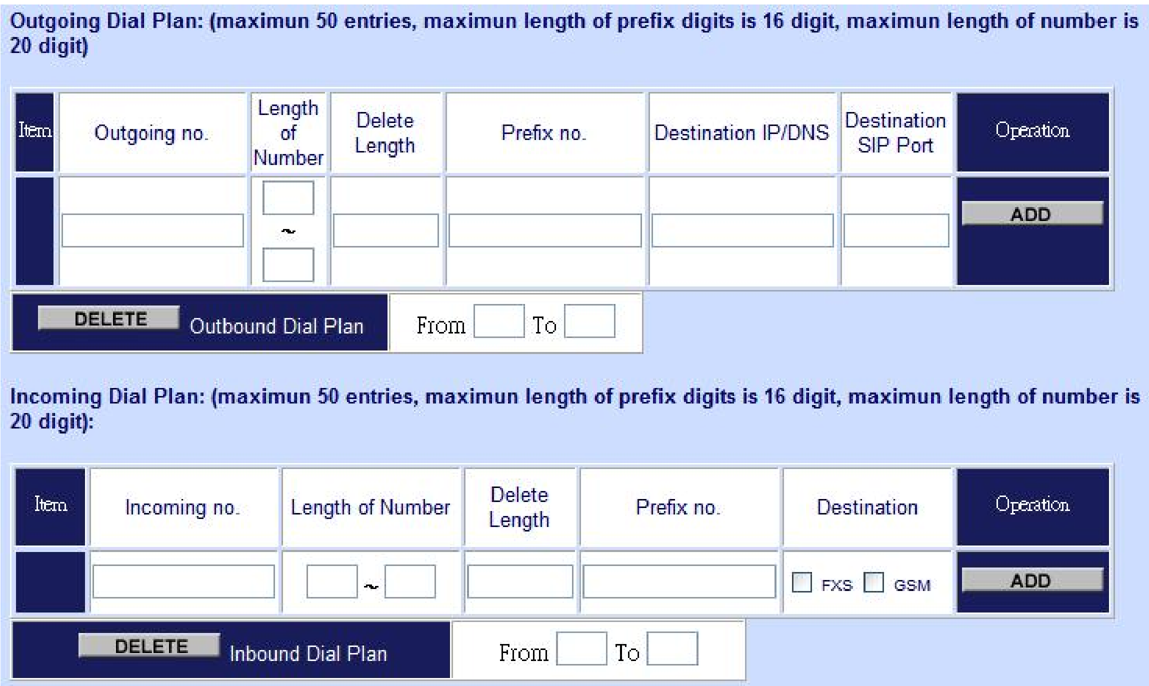



Figure 3-7. Dial plan setting

**In the “Outgoing Dial Plan” settings:**

“**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 configuration setup, please press “Save Configuration” and “Reboot” hyperlinks to save the configuration and rebooting Gateway. After 20 Seconds, you could re-login the Gateway.

# Chapter 4

# GSM Setup

## GSM Setup

In GSM Setup, VIP-281GS provides user the major parts GSM function to configure:

| GSM Setup Label | |
|---|---|
| **GSM Parameter** | GSM Parameter allows you to modify the option of GSM network. |
| **PSTN Dialplan** | Users could apply any dial policy by setting Dial Plan to route the Calls to PSTN |
| **GSM Dialplan** | Users could apply any dial policy by setting Dial Plan to route the Calls to GSM Network. |
| **Send SMS** | The Option is used to send short message to mobile phones |
| **Receive SMS** | This function is used to save the short messages on SIM card to a external file |
| **Terminate Black List** | The numbers in the list can not call from VoIP to GSM Network |
| **Originate Black List** | The numbers in the list can not call from GSM Network to VoIP |

Table 4-1. GSM setup descriptions


Figure 4-1. GSM setup setting

## GSM Parameter

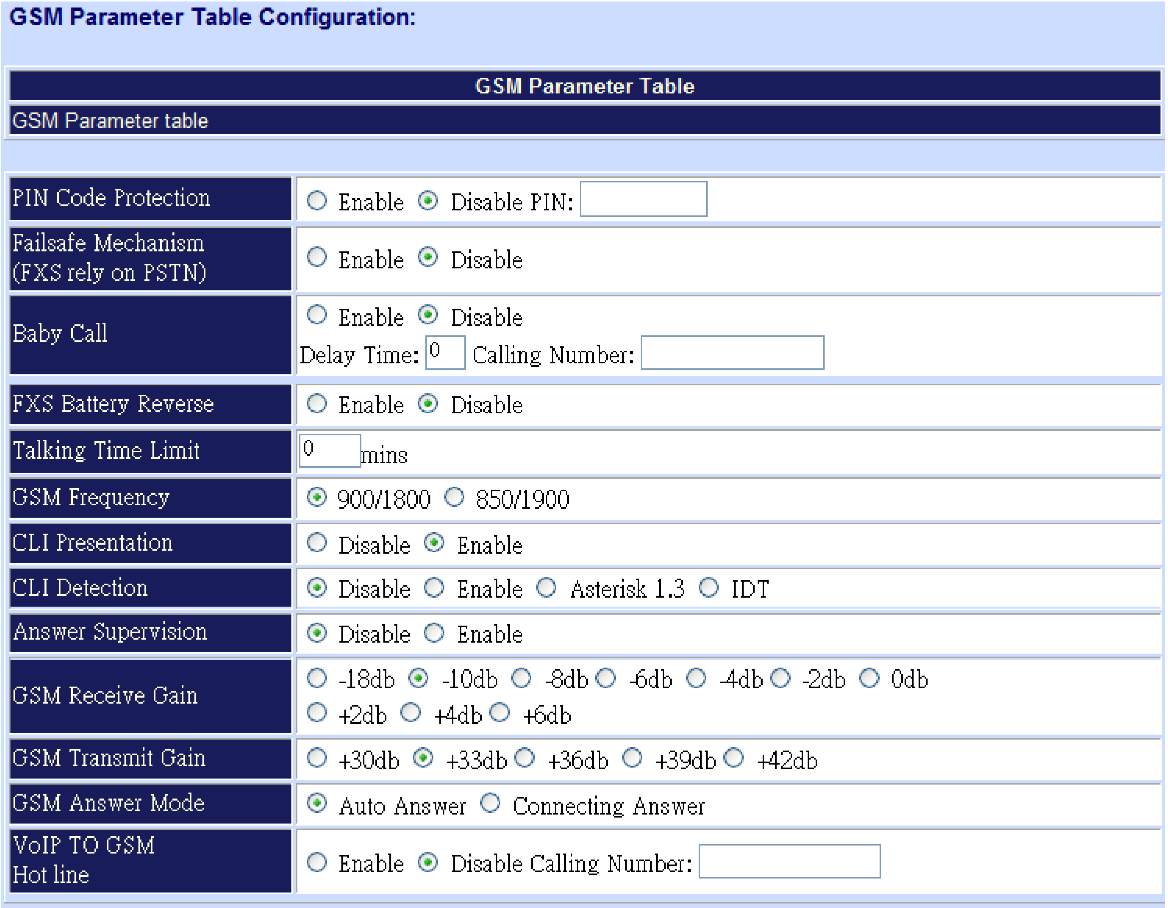



Figure 4-2. GSM parameter setting

| GSM parameter configuration | |
|---|---|
| **PIN Code Protection** | Enable PIN Code protection |
| **Failsafe Mechanism** | If enable, when GSM Network is failed or GSM Gateway is out of the GSM service range. **ALL** the calls from FXS will route to PSTN port. |
| **Baby Call** | When the calls come to FXS port, it will call hot line number to GSM automatically. |
| **FXS Battery Reverse** | Enable battery reverse generator. |
| **Talking Time limit** | The period of talking time, when the time ends, a beep sound will come out as a warning sound. |
| **GSM Frequency** | Select the GSM band |
| **CLI Presentation** | If disable this option, the phone number of SIM card won't be shown in the callee side. |
| **CLI Detection** | If enable, PSTN and GSM number will be carried over Internet in p2p call and asterisk server. if the version of asterisk is old then 1.4,please enable asterisk 1.3. |

| | |
|---|---|
| **Answer Supervision** | Support Battery Reverse Detection. |
| **GSM Receive Gain** | It's able to adjust the GSM Receive Gain, range from -10db to 6db. |
| **GSM Transmit Gain** | It's able to adjust the GSM Transmit Gain, range from 30db to 42db. |
| **GSM Answer Mode** | 1. Auto Answer Mode (Default Setting): GSM Port answers the call once it starts to ring.<br>2. Connecting Answer Mode:<br>Case A: "Hot Line Number" was NOT assigned in the GSM port. GSM answer the call once it starts to ring.<br>Case B: "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device.<br>In this case, GSM port will not answer (off-hook) the call till the user picks up the call.<br>(Note: This case can avoid charging for the call when the remote VoIP device still ring.) |
| **VOIP TO GSM Hot Line** | When VoIP call comes to GSM port, the GSM gateway can dial out to GSM network automatically with specific phone number. |

Table 4-2. GSM parameter descriptions

## PSTN Dialplan

**PSTN Route Numbers:** The numbers which are filled in the form will go through the PSTN line unconditionally. You can use x as wild card.

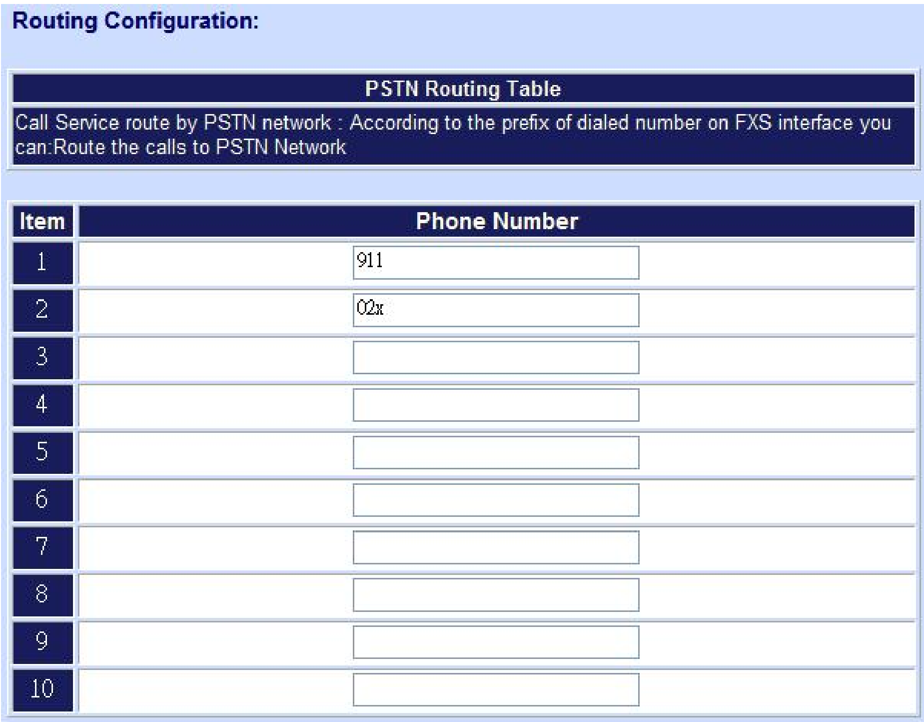



Figure 4-3. PSTN dialplan setting

For examples:

Emergent calls, like 911

Zone Numbers, like 02x (the phone numbers start with 02)

## GSM Dialplan

**GSM Numbers:** The numbers which are filled in the form will go through GSM Network unconditionally. You can use x as wild card.

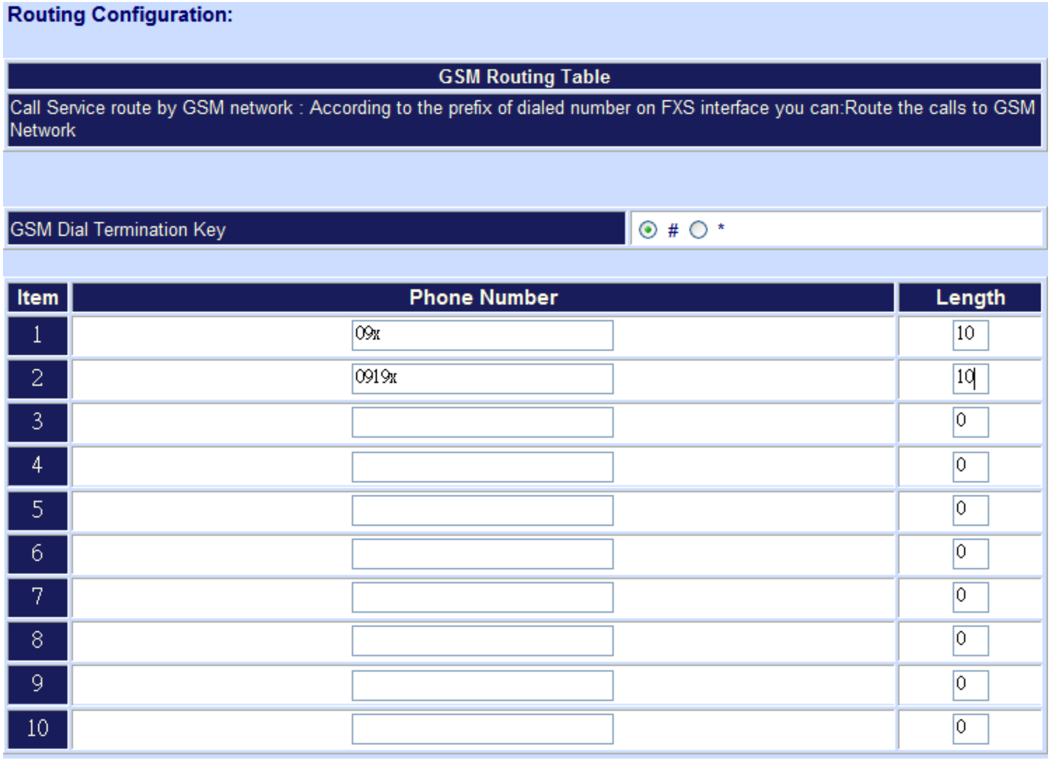



Figure 4-4. GSM dialplan setting

For examples:

**09x** All telephone numbers start with 09

**0919x** All telephone numbers start with 0919

## Send SMS

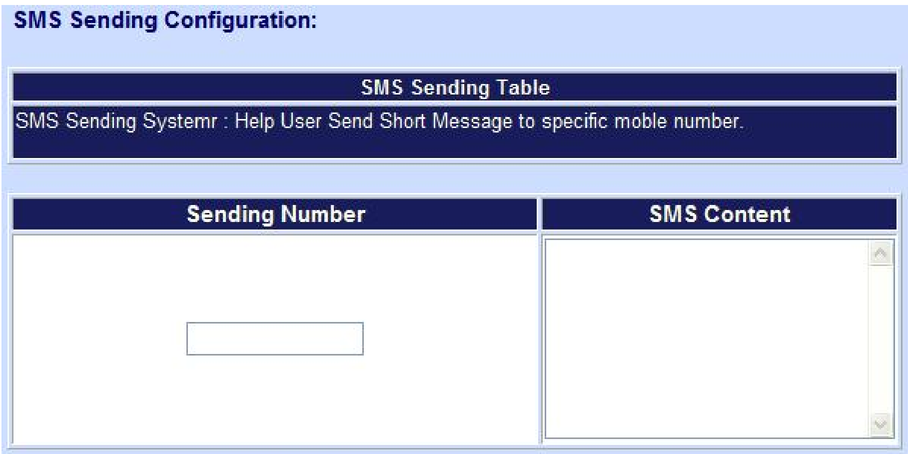



Figure 4-5. SMS sending setting

| SMS sending configuration | |
|---|---|
| **Sending Number** | The telephone number which an short message is sent to. |
| **SMS Content** | The SMS Content will be sent to the preset telephone number. If the SMS text is blank, an empty SMS is sent. The Maximum capacity is 40 characters. |

Table 4-3. SMS sending descriptions

## Receive SMS

This function is used to save the short messages on SIM card to a external file.

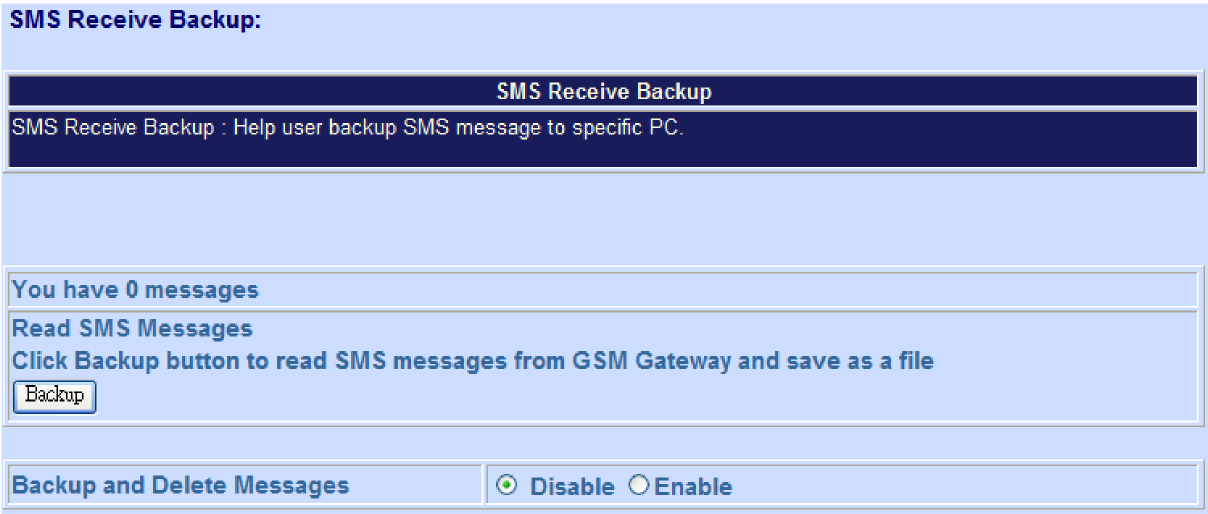



Figure 4-6. SMS Receive Backup setting

## Terminate Phonebook

**Terminate Phone Book:** The following phonebook can set to block or allow. When set to block, call from VoIP to GSM Network match the phone book will be block. When set to allow, only the phone number match the phone book will be allow.

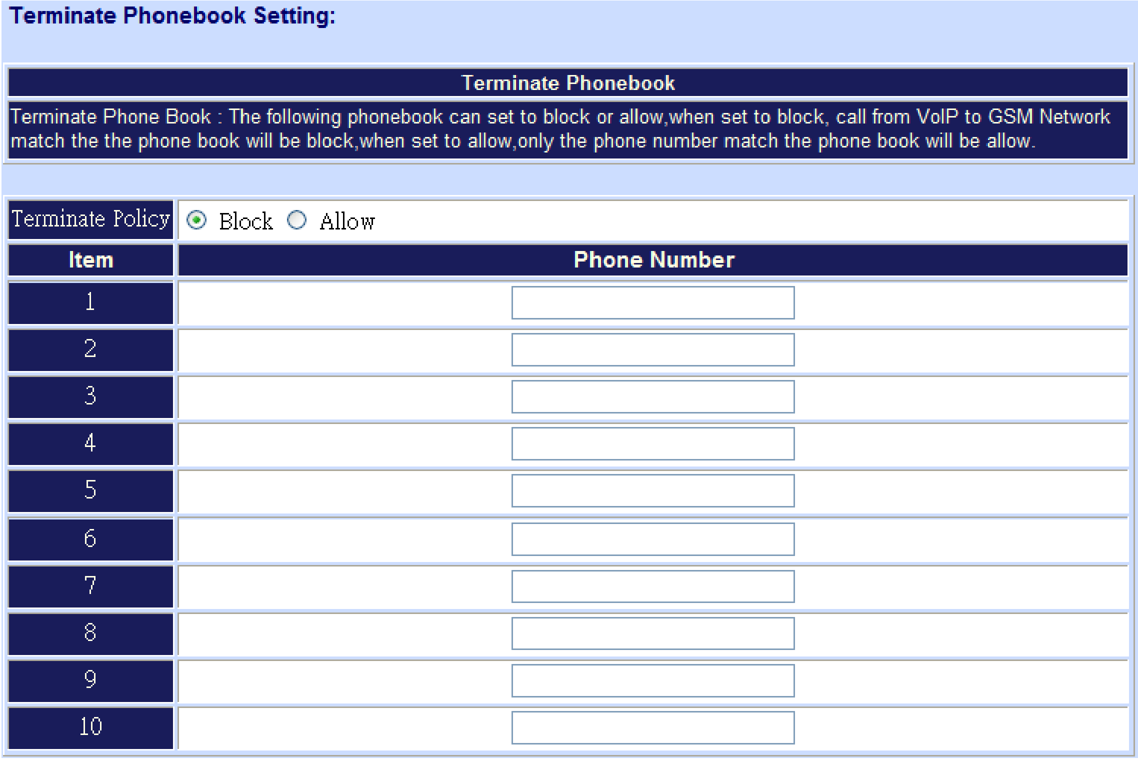


Figure 4-7. Terminate Black setting

## Originate Phonebook

**Originate Phonebook:** The following phonebook can set to block or allow. When set to block, phone number match phonebook can not call from GSM Network to VoIP, When set to allow, only phone number match phonebook call allow to make call.

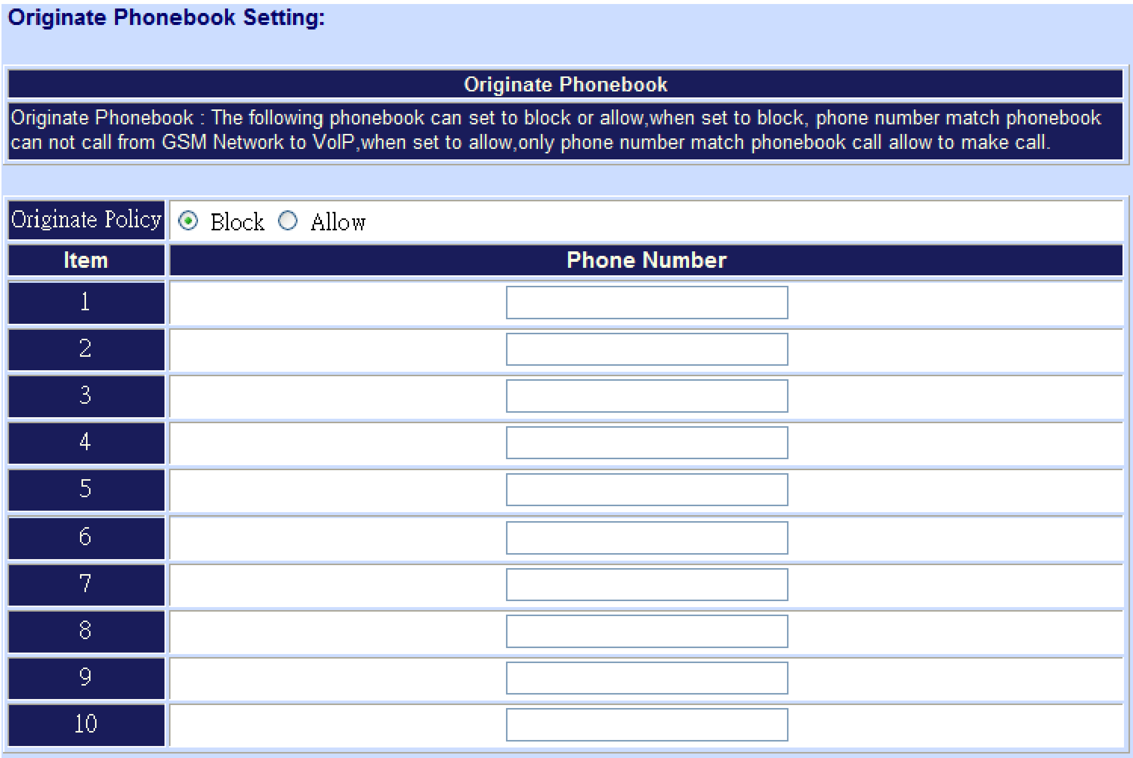


Figure 4-8. Originate Black setting

# Chapter 5

# Advance Setup

## Network Setup

In Network Setup, VIP-281GS provides user the major parts Network function to configure:

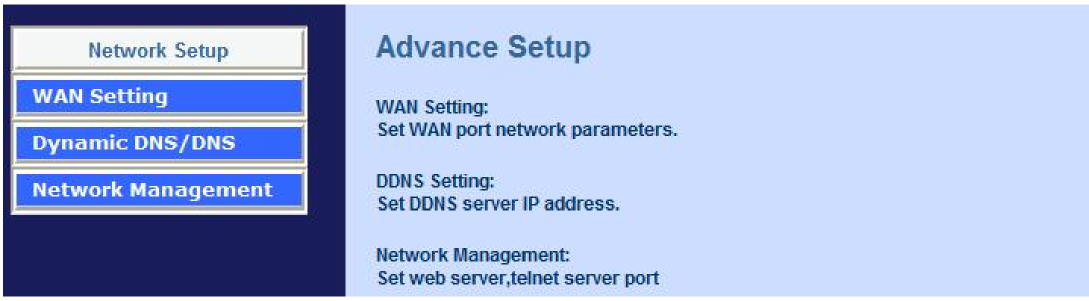



Figure 5-1. Network setup setting

### 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.

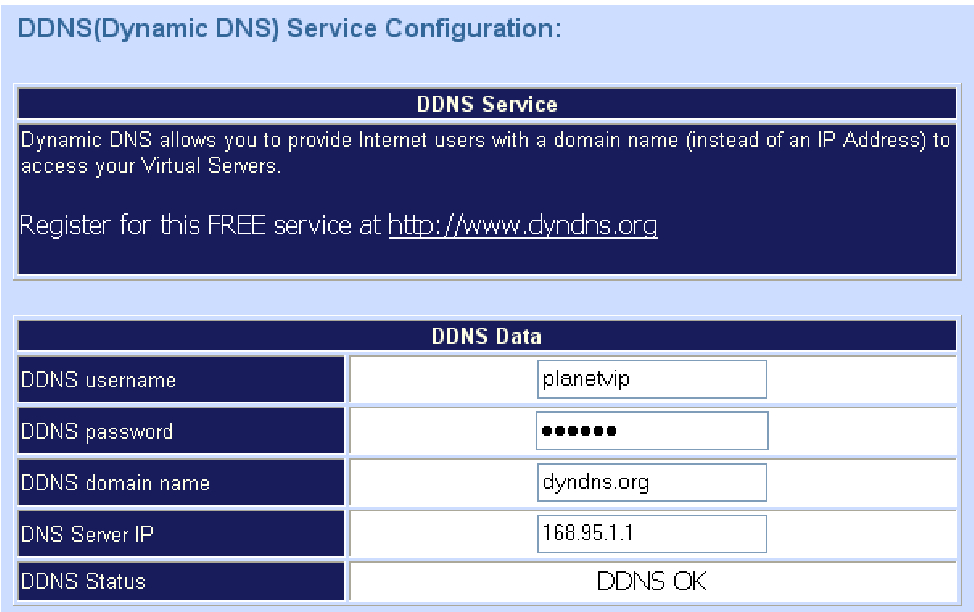



Figure 5-2. DDNS date setting

| Three methods are available for Internet Access | |
|---|---|
| **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 |

Table 5-1. DDNS date descriptions

## Network Management

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

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

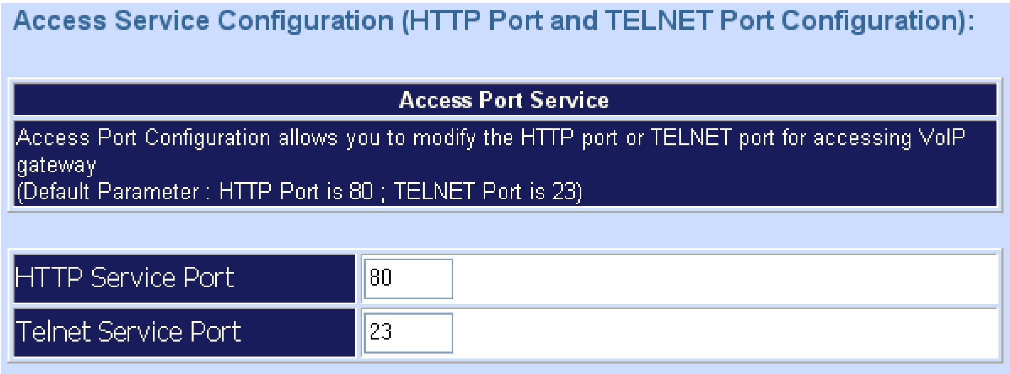



Figure 5-3. Access port service setting

# VoIP Setup

GSM Gateway support 2 VoIP protocol - H.323 / SIP, you can register to H.323 Gatekeeper or SIP proxy server. Gateway is **not a softswitch**, it only can use 1 VoIP protocol (SIP/H.323) at the same time! If you don't register GK or Proxy server, you can make Peer to Peer call by IP address or domain name (Setting Dialing plan).

In VoIP Setup, VIP-281GS provides user the major parts VoIP functions to configure:

| 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-281GS 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. |

Table 5-2. VoIP setup descriptions

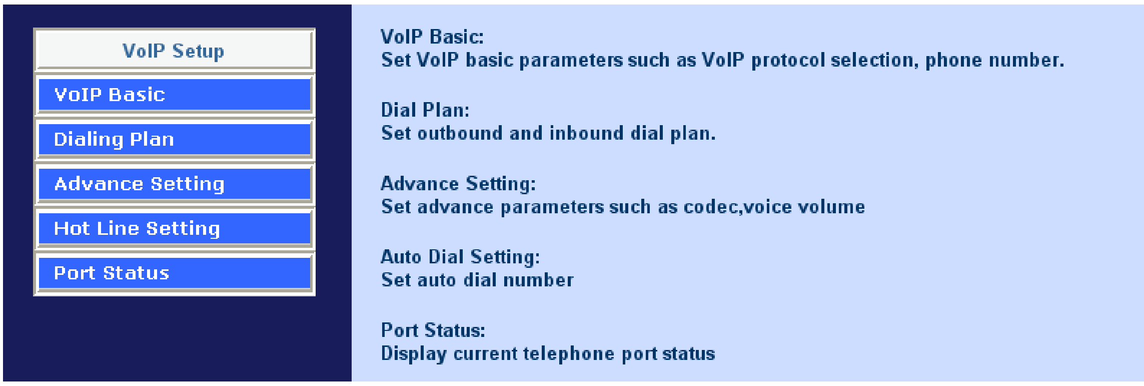



Figure 5-4. VoIP setup setting

## VoIP Basic Configuration to H.323 protocol

Gateway H.323 protocol support H.323 (v2/v3/v4), H.225, Q.931, H.245 and RTP/RTCP. Don't support **H.235 security**, can't use H.235 security Authentication Username / Password. H.323 protocol is not good at pass NAT/Firewall; the best way is installed gateway on Public IP Address when it uses H.323.

**Configure the numbering with FXS/GSM ports.**

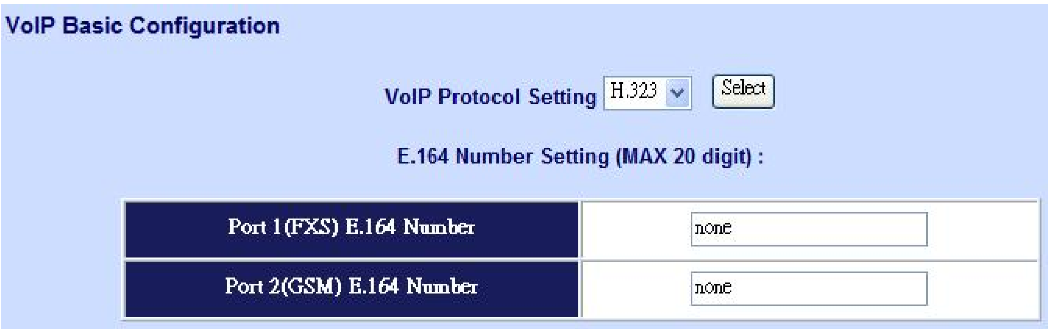



Figure 5-5. E.164 number setting

| E.164 number setting | |
|---|---|
| **FXS Number** | The representation number is the phone number of the telephone that is connected to FXS port. |
| **GSM Number** | The representation number is the phone number of SIM CARD |

Table 5-3. E.164 number descriptions

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

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

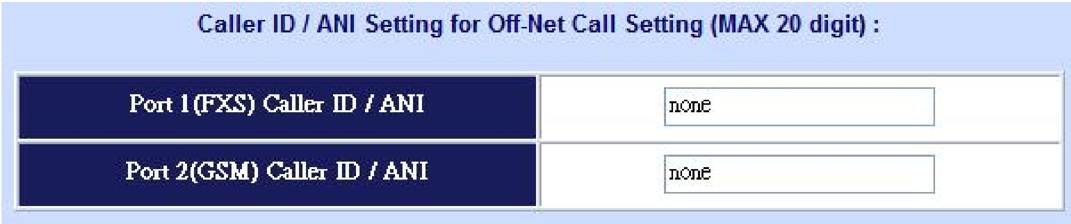



Figure 5-6. Caller ID setting

**Register to H.323 Gatekeeper**

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

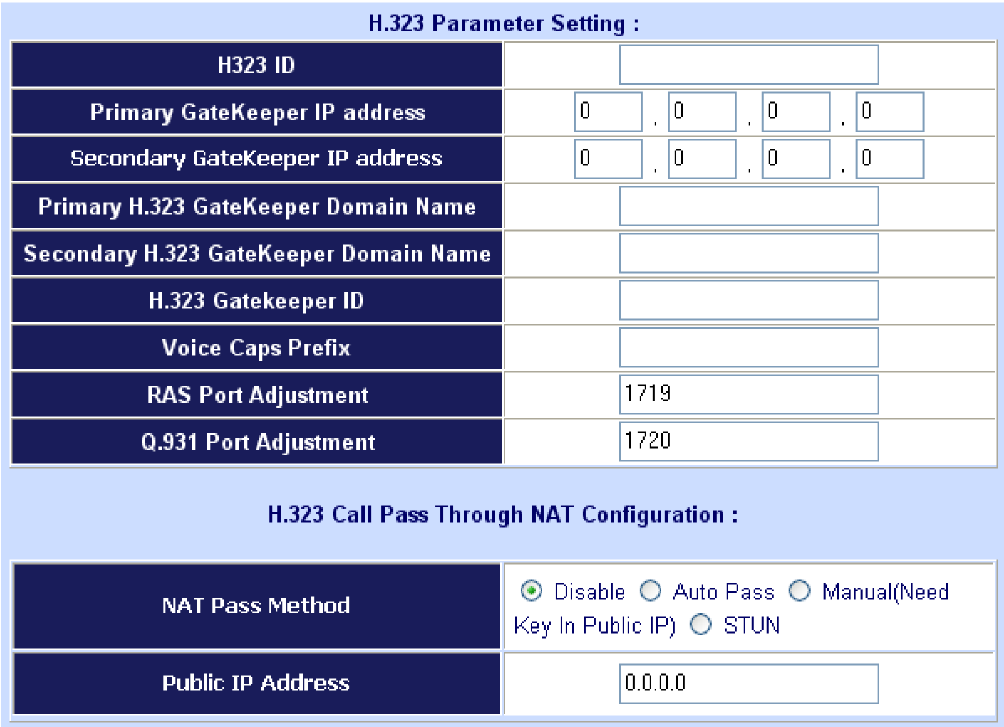



Figure 5-7. H.323 parameter setting

| **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** | 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 |
| **Secondary Gatekeeper** | |

| | |
|---|---|
| **IP Address** | to register to primary gatekeeper, i.e. if both the primary gatekeeper and second gatekeeper |
| **Primary Gatekeeper Domain Name** | Let user use Domain Name of H.323 Gatekeeper. |
| **Secondary Gatekeeper Domain Name** | |
| **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<br>**2. Auto Detection:** When the Gateway register to GNU Gatekeeper, please select this option.<br>**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. |

Table 5-4. H.323 parameter descriptions

## 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 two kinds of directions: Outgoing (call out) and Incoming (call in).

| | |
|---|---|
| **Outgoing Dial Plan** | Peer-to-Peer call mode: Effective<br>Registering to H.323 Gatekeeper mode: Effective |
| **Incoming Dial Plan** | Peer-to-Peer call mode: Effective<br>Registering to H.323 Gatekeeper mode:<br>The leading number would **register** to H.323 Gatekeeper |

Table 5-5. Dial plan descriptions

| In the “Outgoing Dial Plan Configurations” settings: Maximum Entries : 50 | |
|---|---|
| **Outbound number** | The leading digits of the call out dialing number. |
| **Length of Number** | It has two text fields need filled: “Min Length” and “Max Length” is the min/max allowed length you can dial. |
| **Delete Length** | The number of digits that will be stripped from beginning of the dialed number. |
| **Prefix no.** | The digits that will be added to the beginning of the dialed number. |
| **Destination IP / DNS** | The IP address / Domain Name of the destination gateway that owns this phone number. |

Table 5-6. Outgoing dial plan descriptions

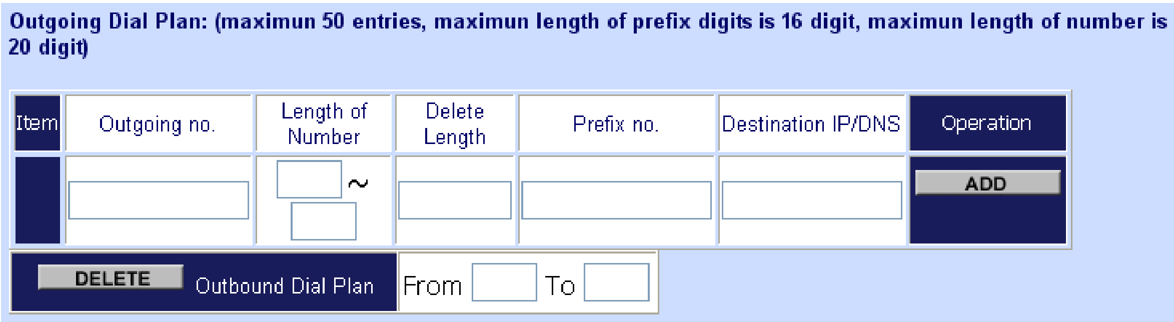



Figure 5-8. Outgoing dial plan setting

## 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

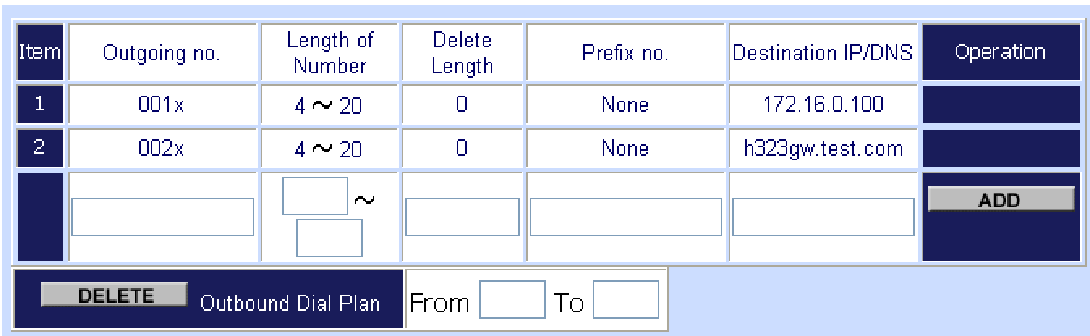
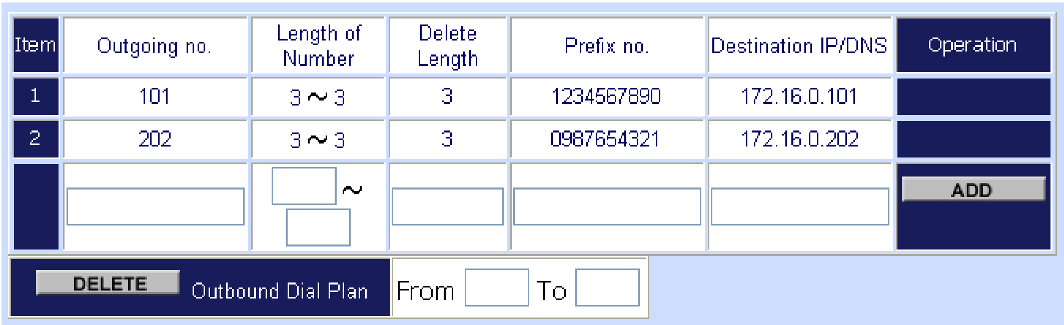



Figure 5-9. Outgoing dial plan setting

## 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

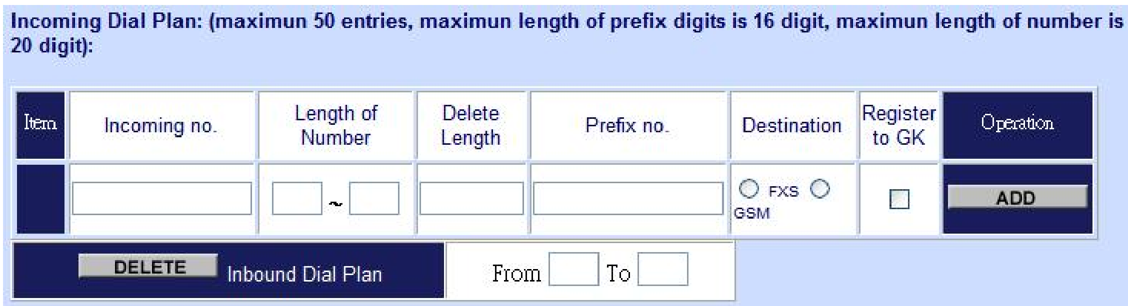


| Item | Outgoing no. | Length of Number | Delete Length | Prefix no. | Destination IP/DNS | Operation |
|---|---|---|---|---|---|---|
| 1 | 101 | 3 ~ 3 | 3 | 1234567890 | 172.16.0.101 | |
| 2 | 202 | 3 ~ 3 | 3 | 0987654321 | 172.16.0.202 | |
| | | ~ | | | | ADD |
| DELETE Outbound Dial Plan | From | To | | | | |

Figure 5-10. Outgoing dial plan setting

| In the “Incoming Dial Plan Configurations” settings: Maximum Entries : 50 | |
|---|---|
| **Inbound number** | The leading digits of the dialing number. |
| **Length of Number** | It has two text fields need filled: “Min Length” and “Max Length” is the min/max allowed length you can dial. |
| **Delete Length** | The number of digits that will be stripped from beginning of the dialed number. |
| **Prefix no.** | The digits that will be added to the beginning of the dialed number. |

Table 5-7. Incoming dial plan descriptions



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 | Register to GK | Operation |
|---|---|---|---|---|---|---|---|
| | | ~ | | | FXS GSM | | ADD |
| DELETE Inbound Dial Plan | From | To | | | | | |

Figure 5-11. Incoming dial plan setting

## Scenario description: Termination call to GSM for one-shoot call

GSM Port: SIM card was connected to GSM Gateway and standby for incoming/outgoing calls properly.

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

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

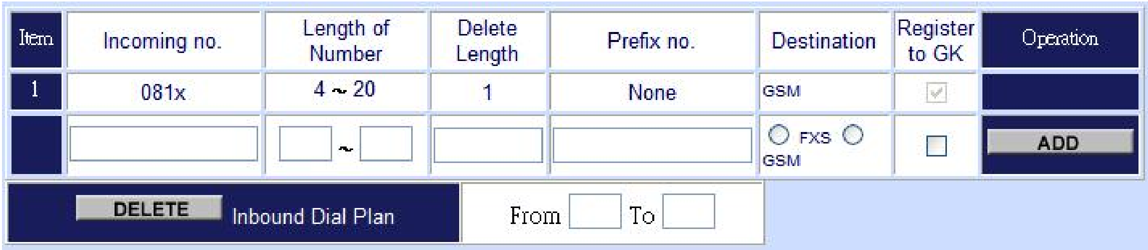



Figure 5-12. Incoming dial plan setting

## Advance Setting to H.323 protocol

In Advanced Setting , VIP-281GS provides user three major parts function to configure:

One is “**VoIP Advance**”, the other are “**Telephone Advance**” , “**Network Advance**” and “**Tone Table Setting**”

- **Advance Setting**

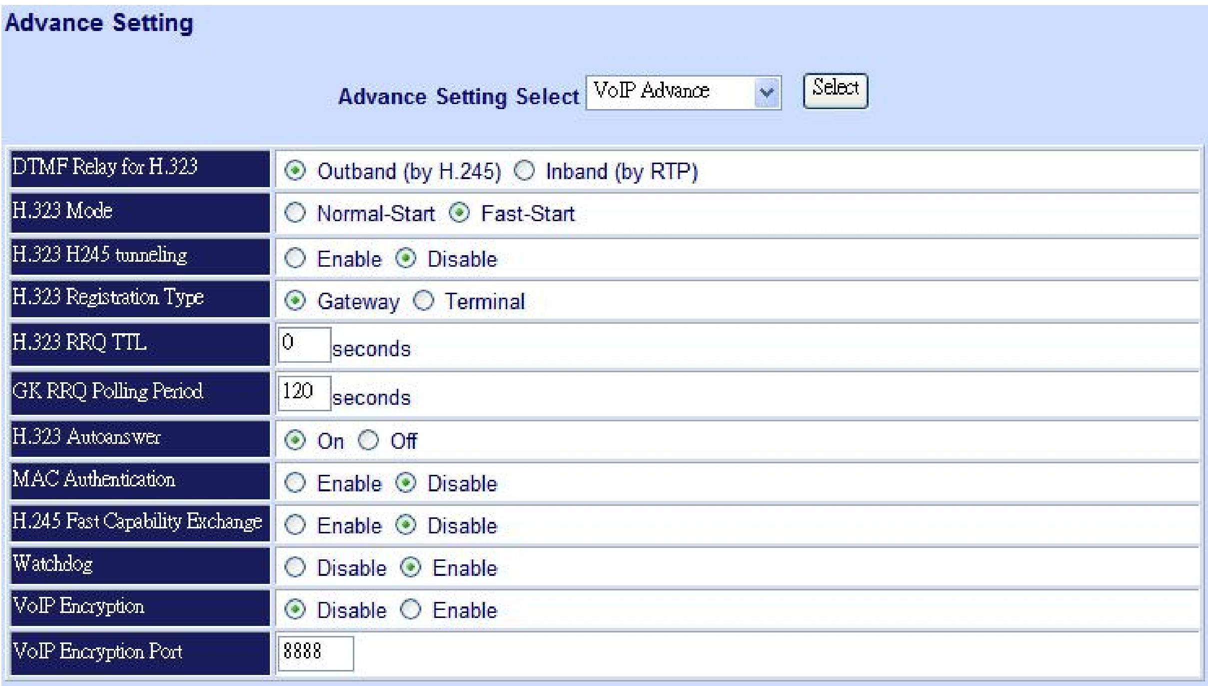



Figure 5-13. VoIP Advance setting

| **H.323 VoIP Advance Configurtion** | |
|---|---|
| **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 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 H.245 Tunneling when establishing a VoIP call The default is disabled (using fast start mode). |
| **H.323 Registration type** | There are 2 choices for this setting. “Gateway” means it will act as the VoIP gateway. “Terminal” means it will act as the IP phone terminal. |
| **H.323 RRQ TTL** | 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 Autoanswer** | When a VoIP call is incoming, the Gateway will ring a specific phone set. The H.323 call signaling part could be connected or alerting during this ringing period. If this selection is enabled, the H.323 signaling part is connected during the ringing period. The benefit of this situation is that the remote side could hear the status of the specific port. That is, the remote side will hear ring back tone if the Gateway is really ringing the phone set. If the phone set is busy, the remote side will hear busy tone. The disadvantage of this situation is that the H.323 connected time is not the real voice call connected time. So, if billing is recorded for this Gateway, this function should be disabled. |
| **MAC Authentication** | Some Gatekeeper register need UA send MAC address to Authentication, you need enable this function.(Default is disable). |
| **Watchdog** | When your gateway shutdown, or something happen that made gateway can’t work fine. Watchdog will reboot your gateway automatically when it can’t work. |

Table 5-8. VoIP Advance descriptions

- **Telephone Advance**

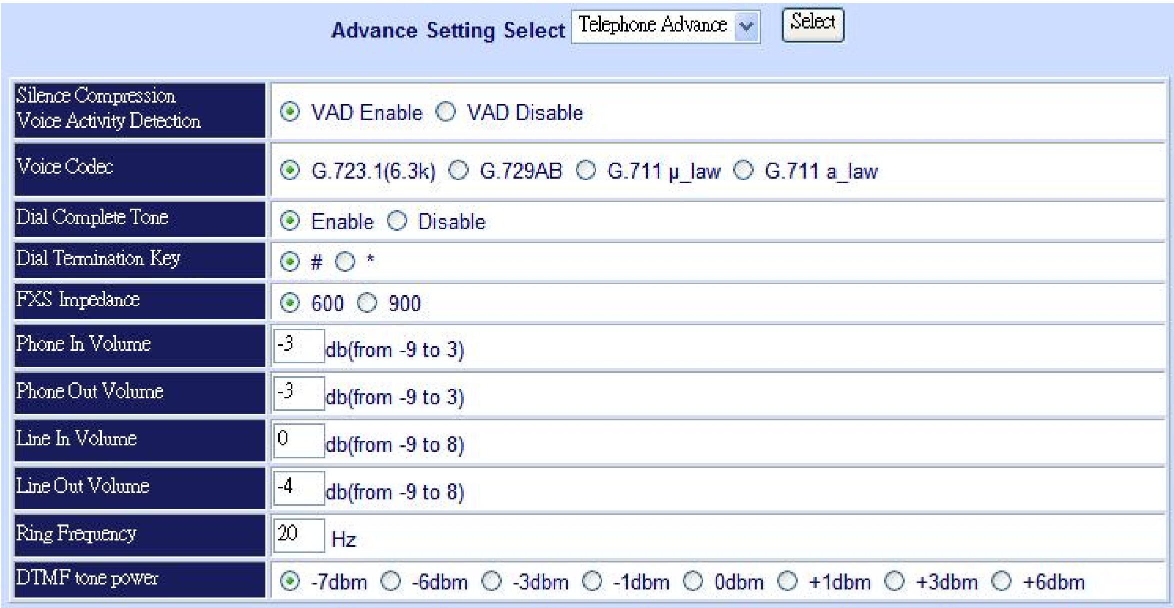



Figure 5-14. Telephone Advance setting

| **H.323 Telephone Advance Configuration** | |
|---|---|
| **Silence Compression (VAD)** | 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.<br>(If you use Asterisk, please disable Silence Compression, it maybe make you call disconnect.) |
| **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**. |
| **Dial Complete Tone** | When you use the VoIP call, you will hear "DuDu" voice that is dial complete tone. If you don't want to hear that tone, you can disable it. (Default is enabling). |
| **Dial Termination key** | Setting Termination key to speed up VoIP dial. Select "*" or "#" to Termination key. |
| **FXS Impedance** | The FXS provides 600/900 OHM impedances for selection. |
| **Phone (Line) in/out volume:** | You can adjust the Phone (Line) in/out volume, range from -9db to 9db<br>(If you adjust too bigger, maybe generation some ECHO or noise) |
| **Ring Frequency** | You can configure how long the Ring Frequency do you want to use. |
| **DTMF tone power** | Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db. |

| | |
|---|---|
| **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. |

Table 5-9. Telephone Advance descriptions

◆ **Network Advance**

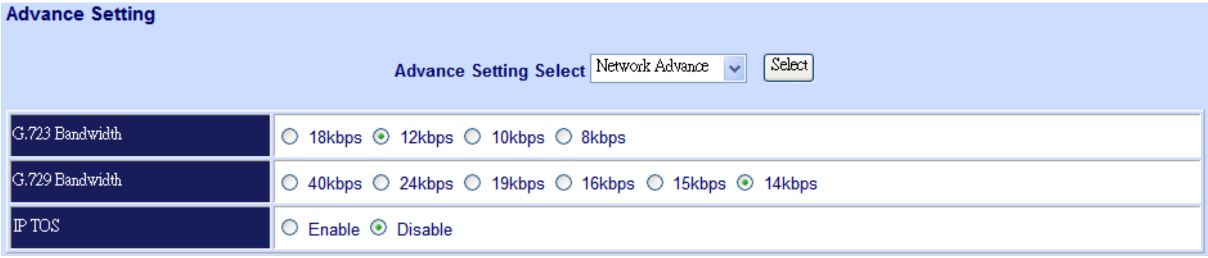



Figure 5-15. Network Advance setting

| **H.323 Netwrok Advance Configuration** | |
|---|---|
| **G.723/G.729 Bandwidth** | Setting G.723 / G.729 voice compression size. Quality and Packet size can adjust by you want. |
| **IP TOS** | Enable / Disable Type of Service in IP packets. |

Table 5-10. Network Advance descriptions

## VoIP Basic Configuration to SIP Protocol

Gateway SIP support SIP(RFC3261), SDP(RFC2327), RFC2833, STUN(RFC3489), Symmetric RTP, outbound proxy, ENUM(RFC2916),and RTP/RTCP.SIP NAT pass through Function can support 80% NAT/Firewall that you don't setting DMZ/Virtual server in router or Firewall.

**Select "SIP Protocol"**

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

**Note:** Support digits and character base SIP Account / username, some SIP Server use character username to login, and a number to call number (ie. VoIPBuster), if your servers don't support this, number/Account is the same, please input the same username, and now only support digits type for SIP number / username

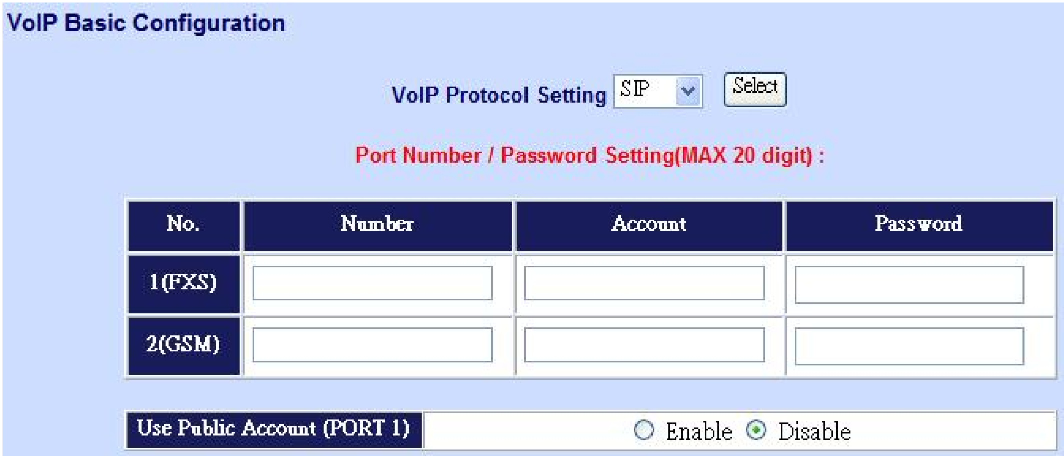



Figure 5-16. Port number setting

| Port Number / Password Setting | |
|---|---|
| **Number** | Input SIP number (Username), if your server support account and number (different), input the number, else number/account are the same username. |
| **Reg** | Let your sip account register SIP Server, click this option. |
| **Account** | Input SIP account (Username), if your server support account and number (different), input the number, else number/account are the same username. |
| **Password** | Input Password that ITSP support. |
| **Use Public Account** | This allows gateway can use single SIP account for multiple ports. User input the only one account in port one field for registering the ITSP. |

Table 5-11. Network Advance descriptions

**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.


Figure 5-17. SIP hunting table setting

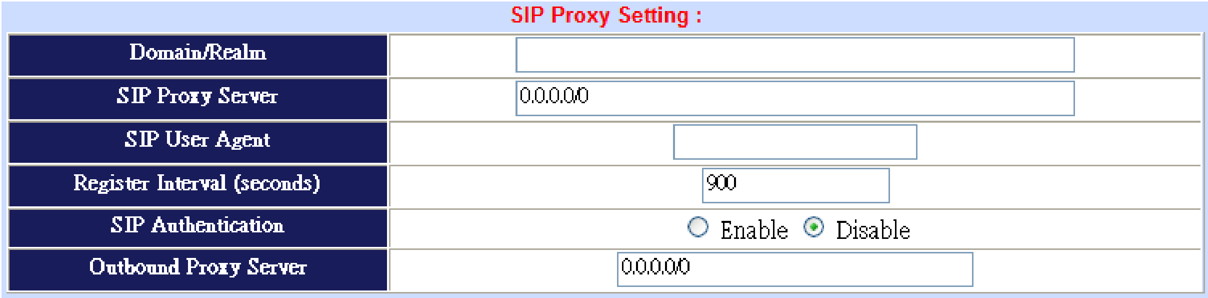



Figure 5-18. SIP proxy setting

| 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). |
| **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 |

Table 5-12. SIP proxy descriptions

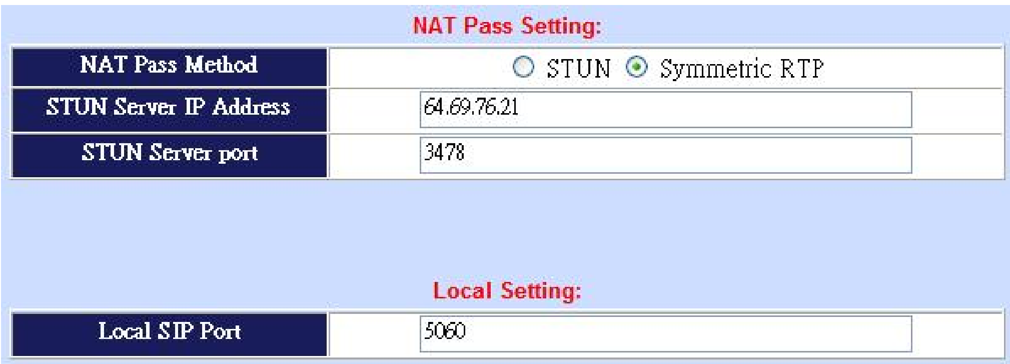



Figure 5-19. NAT pass setting

If your gateway under the NAT/Firewall, you should setting different NAT Pass function. if you setting STUN/Outbound Proxy, you should have a STUN/Outbound proxy server. If they can't pass NAT or one way talk happen, try to open "DMZ" and virtual server "5060" port in router.

| NAT Pass Setting | |
|---|---|
| **NAT Pass Method** | Default use Symmetric RTP pass function. |
| **STUN Client** | Setting your STUN server information, default STUN server is FWD STUN server. |
| **Outbound Proxy Support** | Setting your Outbound Proxy server information. |
| **Local SIP Port** | Setting local use SIP port, default is 5060. |

Table 5-13. SIP proxy descriptions

## 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<br>Registering to SIP Proxy Server Mode: Effective |
| **Incoming Dial Plan** | Peer-to-Peer call mode: Effective<br>Registering to SIP proxy server mode: The leading number would **register** to SIP proxy server |

Table 5-14. Dialing plan descriptions

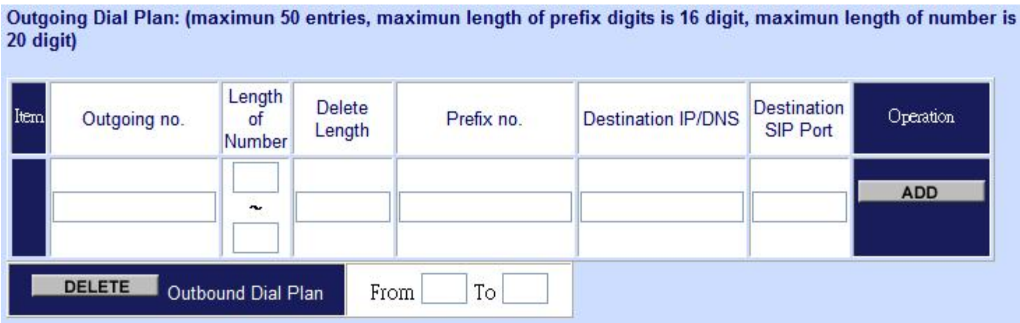



Figure 5-20. Outgoing dial plan setting

| In the "Outgoing Dial Plan Configurations" settings: Maximum Entries : 50 | |
|---|---|
| **Outbound number** | The leading digits of the call out dialing number. |
| **Length of Number** | It has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial. |
| **Delete Length** | The number of digits that will be stripped from beginning of the dialed number. |
| **Prefix no.** | The digits that will be added to the beginning of the dialed number. |

| | |
|---|---|
| **Destination IP / DNS** | The IP address / Domain Name of the destination gateway that owns this phone number. |
| **Destination SIP Port** | It is the UDP port of the remote SIP proxy, which usually refer to the SIP server on the ITSP side. |

Table 5-15. Outgoing dial plan descriptions

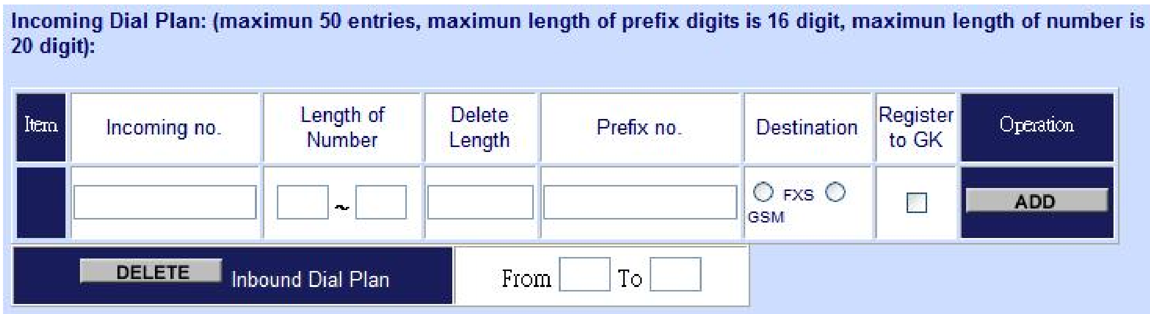



Figure 5-21. Incoming dial plan setting

| In the "Incoming Dial Plan Configurations" settings: Maximum Entries : 50 | |
|---|---|
| **Inbound number** | The leading digits of the dialing number. |
| **Length of Number** | It has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial. |
| **Delete Length** | The number of digits that will be stripped from beginning of the dialed number. |
| **Prefix no.** | The digits that will be added to the beginning of the dialed number. |

Table 5-16. Incoming dial plan descriptions

## Advance Setting to SIP protocol

In Advanced Setting, VIP-281GS provides user three major parts function to configure: One is "**VoIP Advance**", the other one is "**Telephone Advance**" , "**Network Advance**" and "**Tone Table Setting**"

- **VoIP Advance**

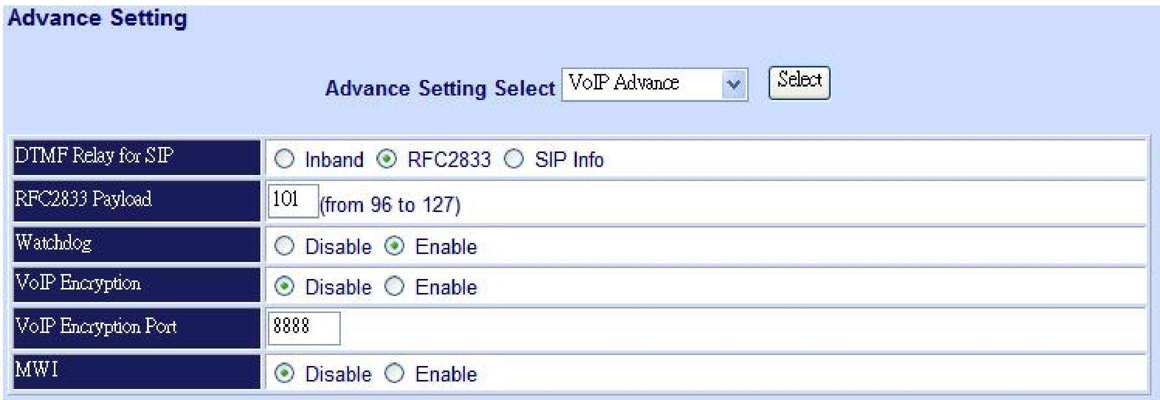



Figure 5-22. VoIP Advance setting

| SIP VoIP Advance Configurtion | |
|---|---|
| **DTMF Relay 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. |
| **RFC2833 Payload** | Adjust RFC2833 DTMF payload value; range from 96 to 127, default is 101. |
| **Watchdog** | When your gateway shutdown, or something happen that made gateway can't work fine. Watchdog will reboot your gateway automatically when it can't work. |

Table 5-17. VoIP Advance descriptions

◆ **Telephone Advance**

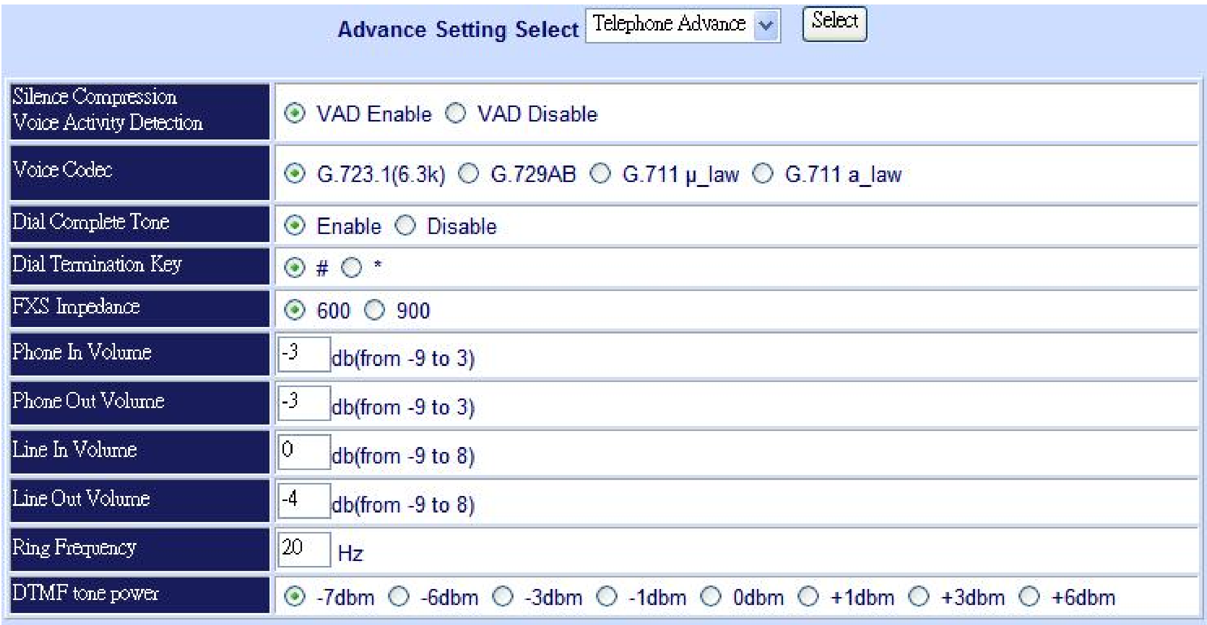



Figure 5-23. Telephone Advance setting

| SIP Telephone Advance Configuration | |
|---|---|
| **Silence Compression (VAD)** | 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. (If you use Asterisk, please disable Silence Compression, it maybe make you call disconnect.) |

| | |
|---|---|
| **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. |
| **Dial Complete Tone** | When you use the VoIP call, you will heard “DuDu” voice that is dial complete tone. If you don’t want to heard that tone , you can disable it.(default is enable). |
| **Dial Termination key** | Setting Termination key to speed up VoIP dial. Select “*” or “#” to Termination key. |
| **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. (If you adjust too bigger, maybe generation some ECHO or noise) |
| **Ring Frequency** | You can configure how long the Ring Frequency do you want to use. |
| **DTMF tone power** | Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db. |

Table 5-18. Telephone Advance descriptions

- **Network Advance**

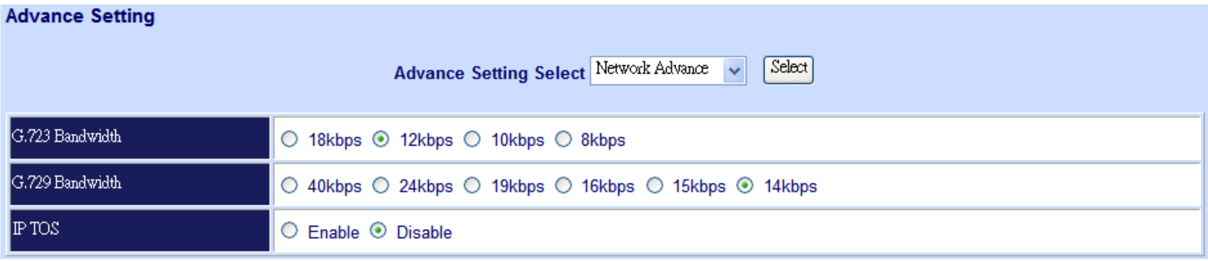



Figure 5-24. Network Advance setting

| 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** | You can configure your bandwidth what the Max byte of download and upload of ADSL modem rate. |
| **G.723/G.729 Bandwidth** | Setting G.723 / G.729 voice compression size. Quality and Packet size can adjust by you want. |
| **IP TOS** | Some Router support TOS(Type of Service), when you enable the TOS function, the router will process those packets firstly.(default is disable) |

Table 5-19. Network Advance descriptions

## Hot Line Setting

You can set hot line. When the call incoming the hot line port, it will call hot line number automatically. The hot line calls the number via VoIP, so you setting the hot line number must VoIP number. Usually, you want to incoming GSM calls transfer to FXS, you only setting the GSM hot line to FXS number.

**Port number:** Input FXS/GSM wants to call hot line number. The call will via VoIP, so the number must be the VoIP number.

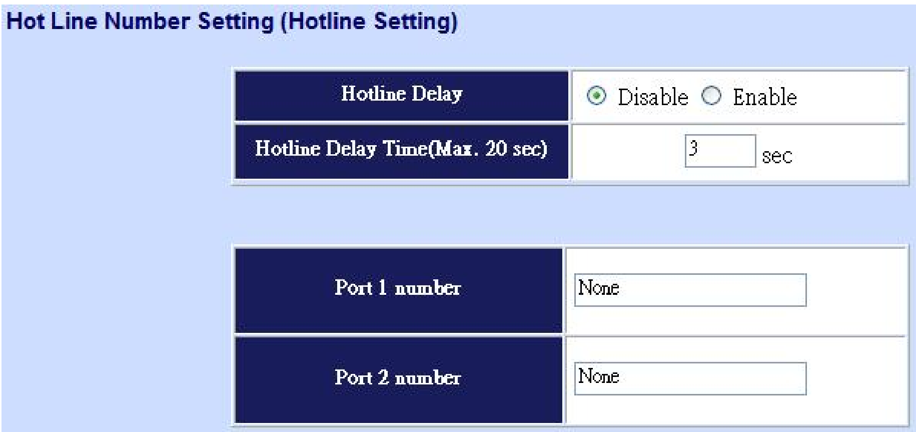



Figure 5-25. Hot line setting

## Port Status

Each of port show status table. You can view all port status. Like on/off hook, caller/callee IP, duration, and packet loss.

**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 | GSM | onhook | none | none | none | none | none | 0 | 0 | 0 |

Figure 5-26. Port status

# Chapter 6 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-281GS with the default configuration. |
| **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-281GS can send log information to Syslog Server by UDP ports 514. |
| **Capture Packets** | The VIP-281GS supports packets capture and save the packets to your PC. |

Table 6-1. Management descriptions

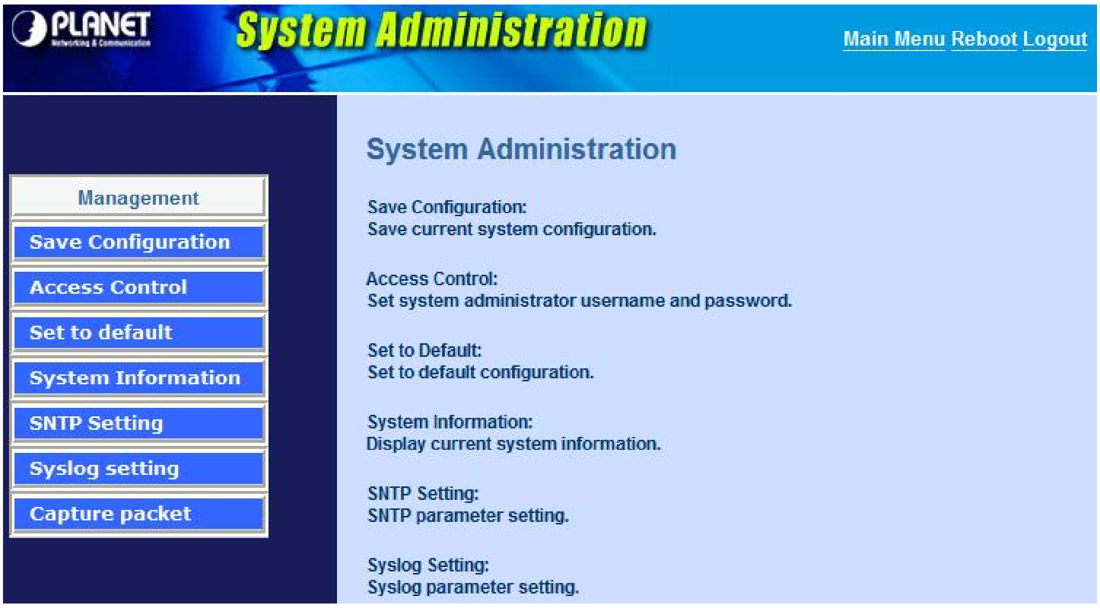



Figure 6-1. Management setting

## Save Configuration

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


Figure 6-2. Save setting

## 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**

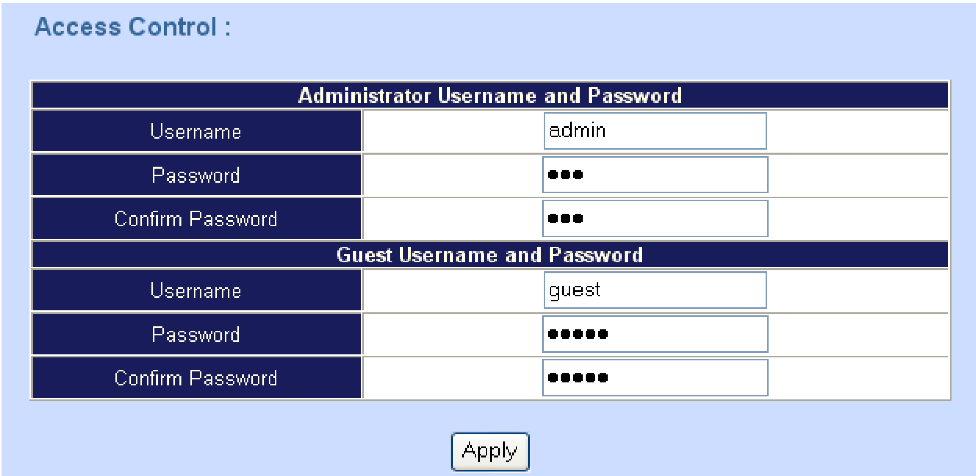



Figure 6-3. Access control setting

## 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.

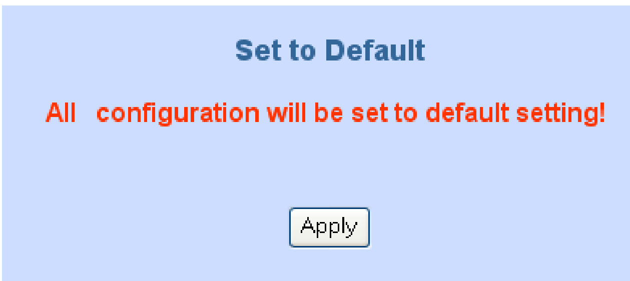
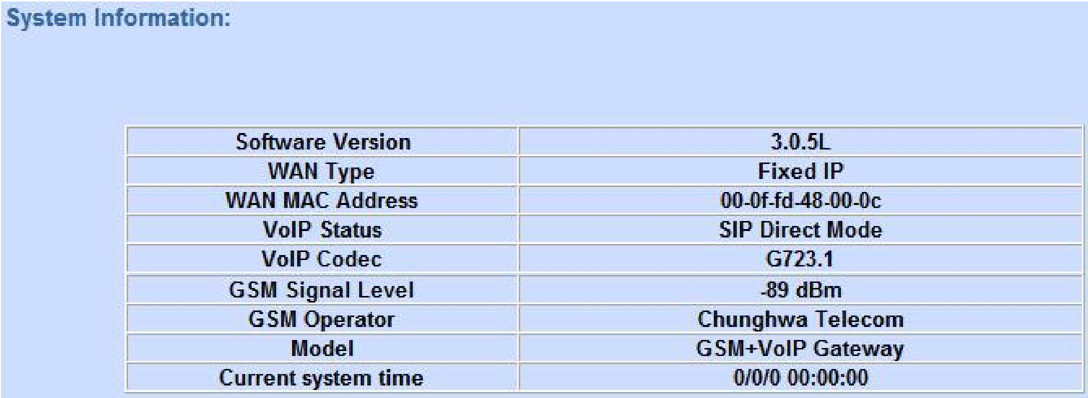



Figure 6-4. Set to default setting

## 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.


System Information:

| | |
|---|---|
| Software Version | 3.0.5L |
| WAN Type | Fixed IP |
| WAN MAC Address | 00-0f-fd-48-00-0c |
| VoIP Status | SIP Direct Mode |
| VoIP Codec | G723.1 |
| GSM Signal Level | -89 dBm |
| GSM Operator | Chunghwa Telecom |
| Model | GSM+VoIP Gateway |
| Current system time | 0/0/0 00:00:00 |

Figure 6-5. System information

## SNTP Setting Function

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

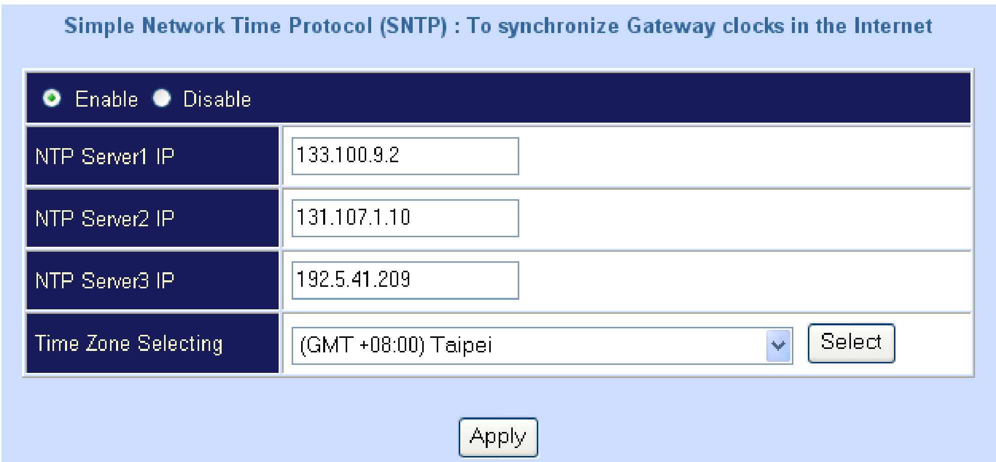



Figure 6-6. SNTP setting

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-281GS 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.

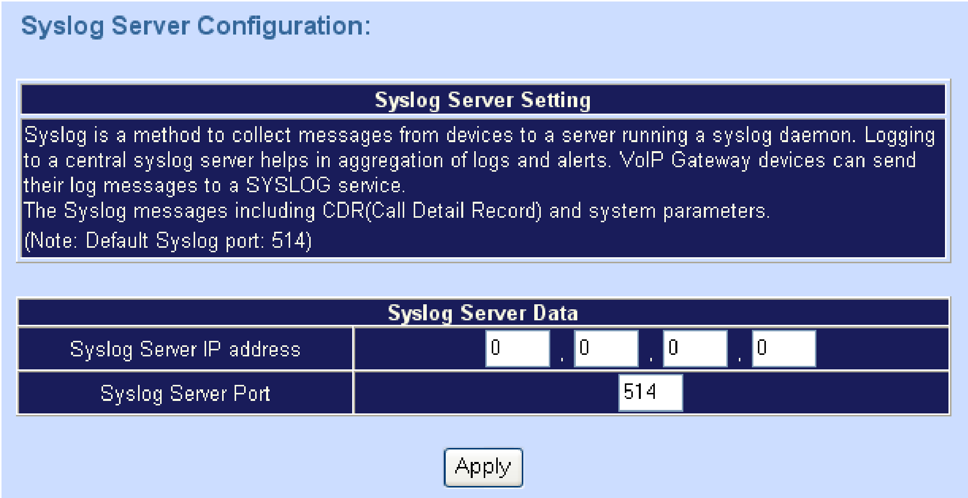



Figure 6-7. Syslog setting

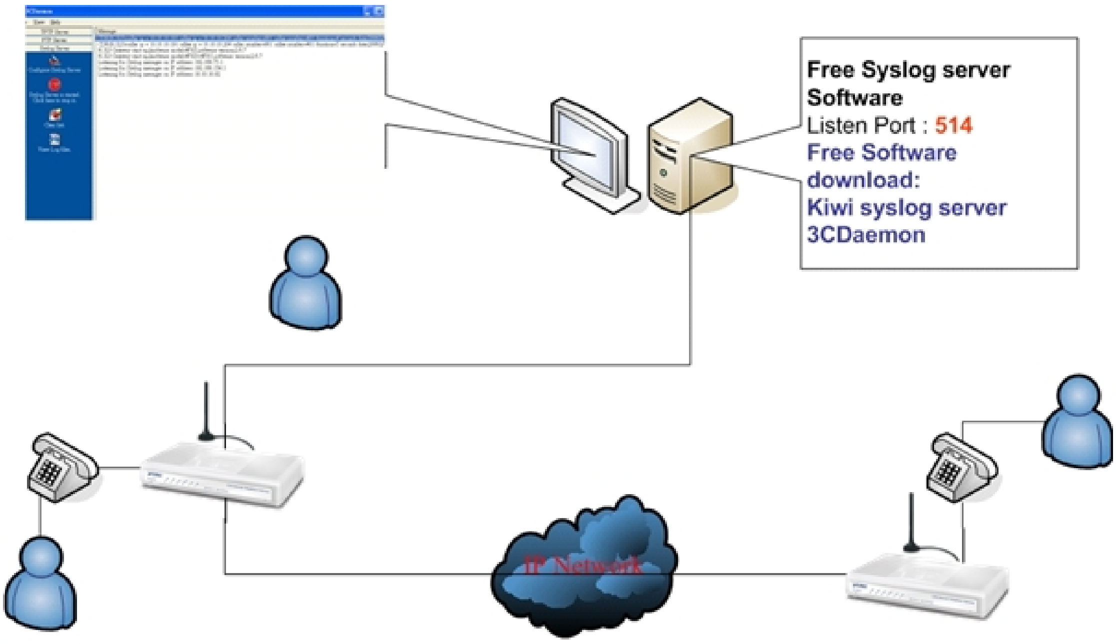



Figure 6-8. Syslog topology

## Capture packets Function

Use “Capturer Packets” to record VIP-281GS packets. Users can start and stop the capture then save the file to PC. Use the Ethereal Tool ([www.ethereal.com](www.ethereal.com)) to analyze the packets.

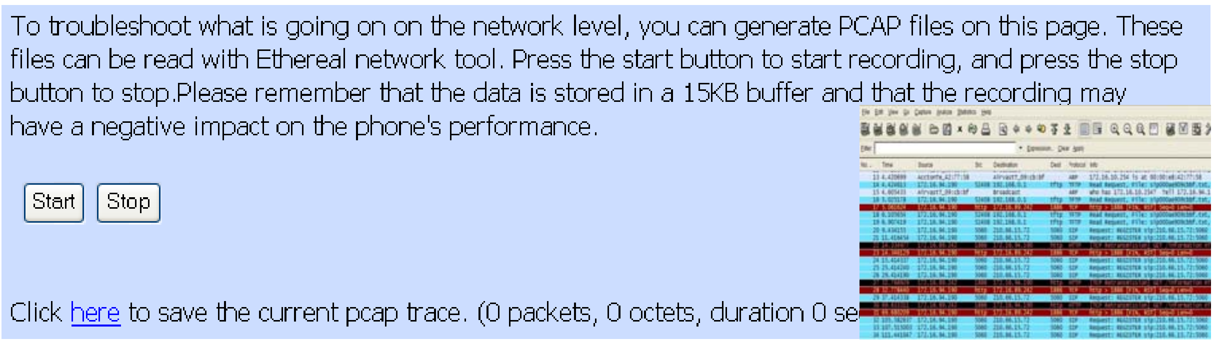



Figure 6-9. Capture packets setting

# Appendix A

## Voice communications

The chapter shows you the concept and command to help you configure your PLANET VIP-281GS through sample configuration. And provide several ways to make calls to desired destination in VIP-281GS. 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, **Phone** (**FXS**, Foreign exchange Station) on the printing of the RJ-11 port, and **GSM** on the printing of the SIM port, you should find that.

- **Phone port**

The Phone port allows the connection to an end node, like **telephone**, or **out-line of PBX system**. Phone 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 Phone port and you will hear the dial-tone from Phone port once the hand set off-hook.

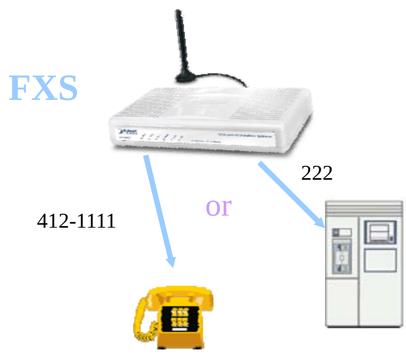



Figure A-1. Phone port topology

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

- **GSM port**

The GSM port allows can be inserted a SIM card that already has a fixed number; say 0912-111111. So the only connections for GSM port will be to your local PSTN or GSM network.
With your GSM connect to GSM network; the Internet Voice can then have a GSM call through this line/number (0912-111111). Or, locally, you can have an Internet Call through the line 0912-111111.

Your PBX users will need to know this number in the future.

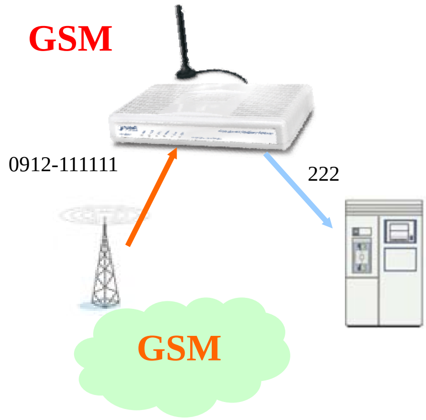



Figure A-2. GSM port topology

## Sample scenario_1: Peer to Peer GSM termination

In the following samples, we'll introduce the Peer to Peer GSM termination applications.

In this example, there are two VIP-281GS calling by IP address directly, both VIP-281GS have inserted the GSM SIM cards into SIM slots, the GSM number are 09127788(GSM_1) and 09583344(GSM_2).

The VoIP number of VIP-281GS_A are ext.100 (FXS) and ext.200 (GSM), the VoIP number of VIP-281GS_B are ext.300 (FXS) and ext.400 (GSM)

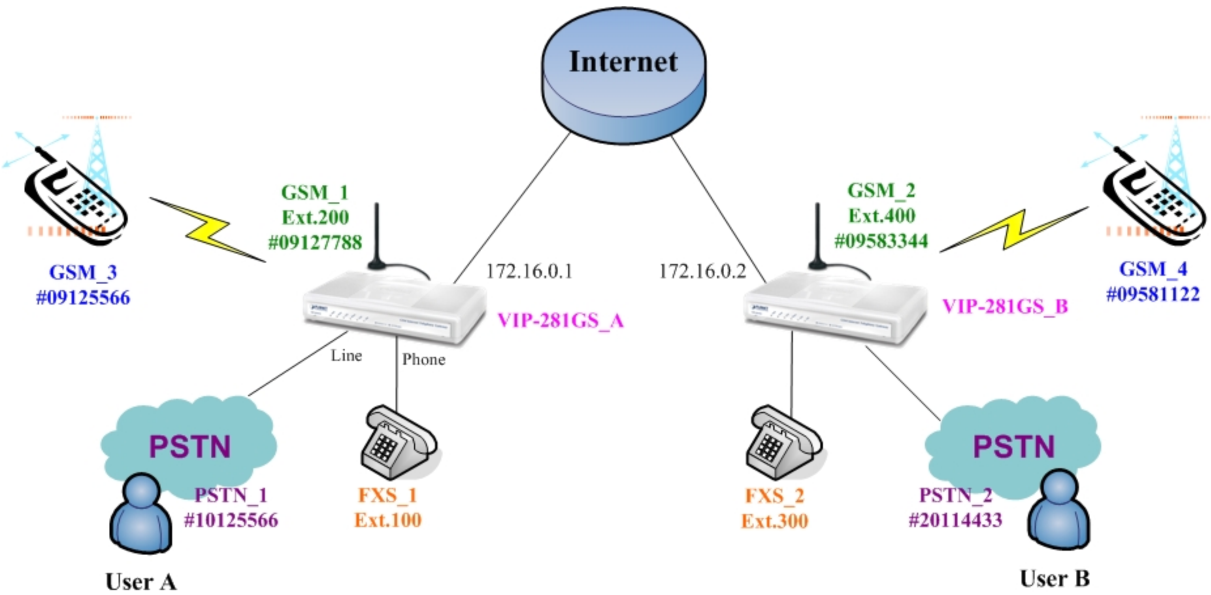
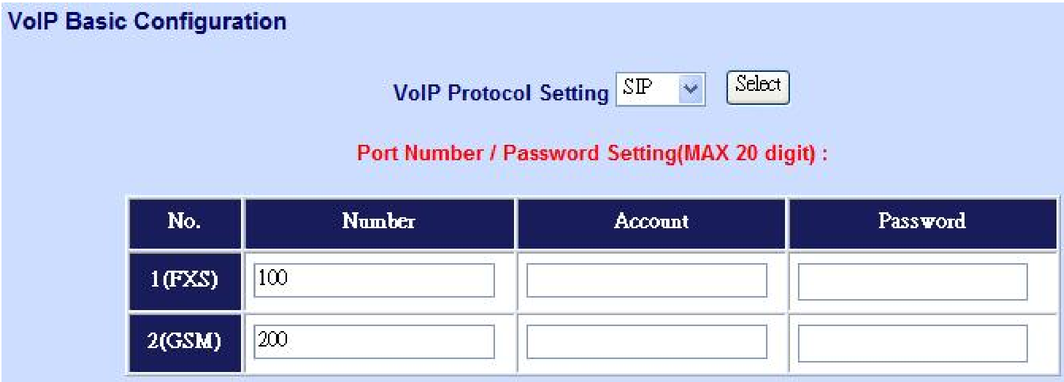



Figure A-3. Peer to Peer GSM topology

### Machine configuration on the VIP-281GS:

**STEP 1:**

Please log in VIP-281GS_A via web browser, browse to the **Advance Setup -> VoIP Basic** menu and set the VoIP number as 100 and 200, the sample configuration screen is shown below:

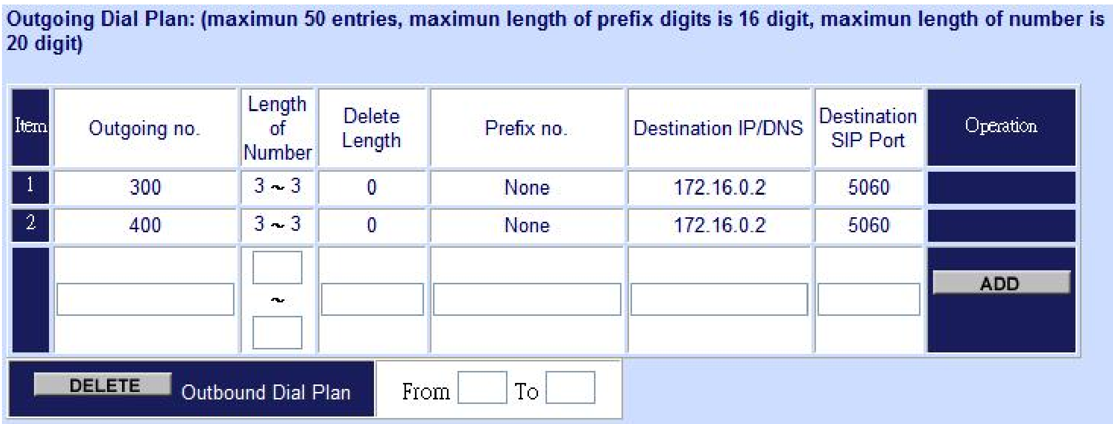



| No. | Number | Account | Password |
|---|---|---|---|
| 1(FXS) | 100 | | |
| 2(GSM) | 200 | | |

Figure A-4. VoIP basic settings

**STEP 2:**

Please browse to the **Dial Plan** menu and add the outgoing dial plan for calling to VIP-281GS_B, the sample configuration screen is shown:




| Item | Outgoing no. | Length of Number | Delete Length | Prefix no. | Destination IP/DNS | Destination SIP Port | Operation |
|---|---|---|---|---|---|---|---|
| 1 | 300 | 3 ~ 3 | 0 | None | 172.16.0.2 | 5060 | |
| 2 | 400 | 3 ~ 3 | 0 | None | 172.16.0.2 | 5060 | |
| | | ~ | | | | | ADD |

DELETE Outbound Dial Plan From To

Figure A-5. Outgoing dial plan settings

**STEP 3:**

Please browse to the **GSM Setup -> PSTN Dial plan** menu and set the PSTN outgoing number, the sample configuration screen is shown:

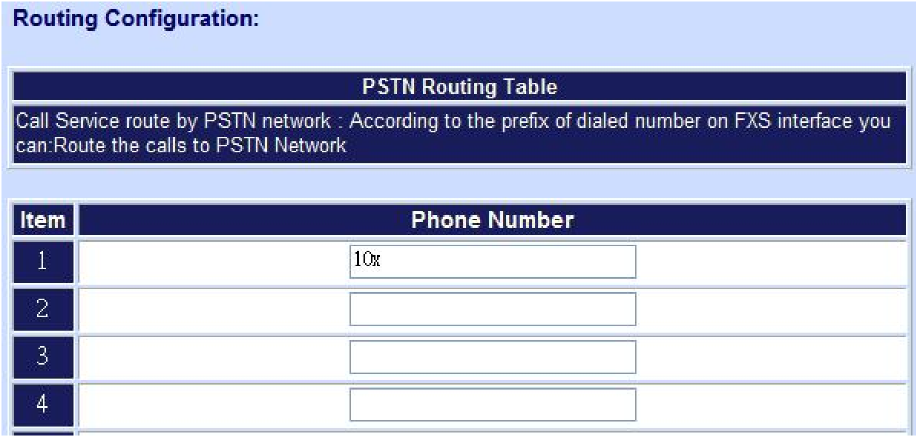



Figure A-6. PSTN Routing table

**STEP 4:**

Please browse to the **GSM Dial plan** menu and set the GSM outgoing number, the sample configuration screen is shown:

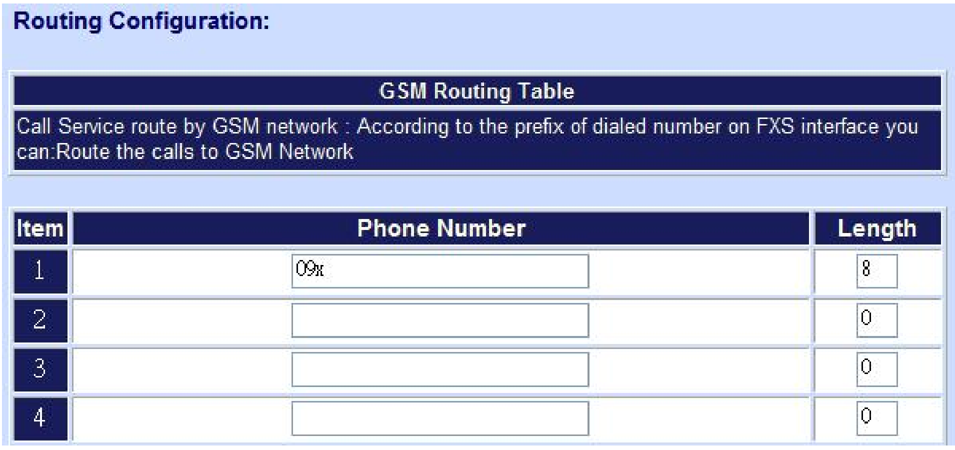



Figure A-7. GSM Routing table

**STEP 5:**

Repeat the same configuration steps on VIP-281GS_B.

## Test the scenario:

### A. FXS_1 call to GSM_4

1. FXS_1 pick up the telephone.
2. Dial the ext.400 to GSM port of VIP-281GS_B, and get the dial tone.
3. Dial the GSM number #09581122 to establish the voice communication with GSM_4.

### B. GSM_3 call to FXS_2

1. GSM_3 dial the GSM number #09127788 to GSM_1, and get the dial tone.
2. Dial the ext.300 to establish the voice communication with FXS_2.

### C. FXS_1 call to PSTN_1

1. FXS_1 pick up the telephone.
2. Dial the PSTN number #10125566 to establish the voice communication with PSTN_1.

## Sample scenario_2: Enterprise SIP + GSM termination

In the following samples, we'll introduce the SIP Proxy and GSM termination applications.

In this example, there are two VIP-281GS; the FXS and GSM ports are register to SIP Proxy Server (IP PBX).

The out-lines of PBX connect with Phone (FXS) ports of VIP-281GS. The extensions of PBX can make GSM calls via GSM ports of VIP-281GS.

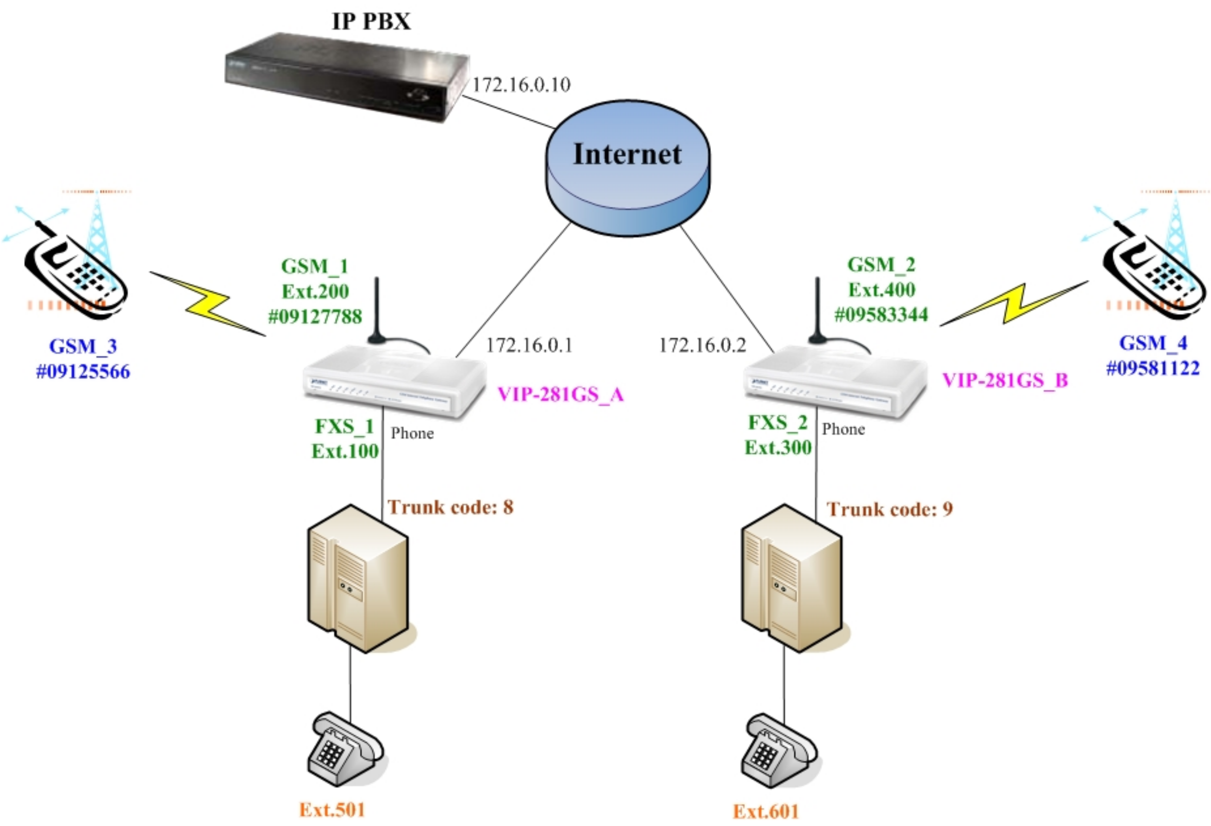



Figure A-8. Enterprise GSM Routing table

## Machine configuration on the VIP-281GS:

**STEP 1:**

Please log in VIP-281GS_A via web browser, browse to the **Advance Setup -> VoIP Basic** menu, set the VoIP registration number as 100/ 200 and the registration server address, the sample configuration screen is shown below:

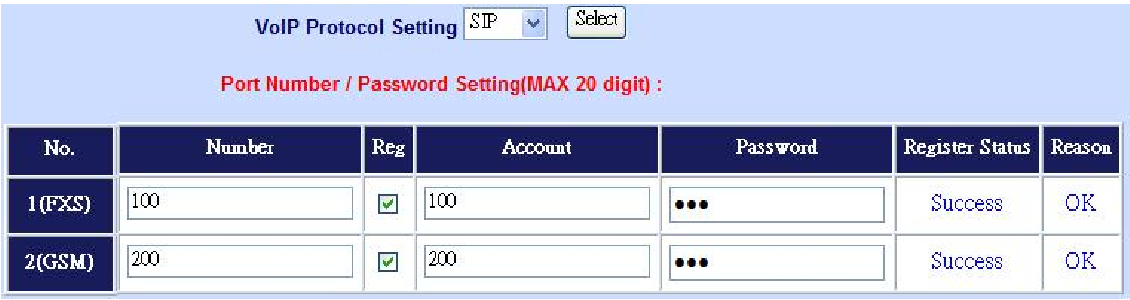



Figure A-9. Port number settings

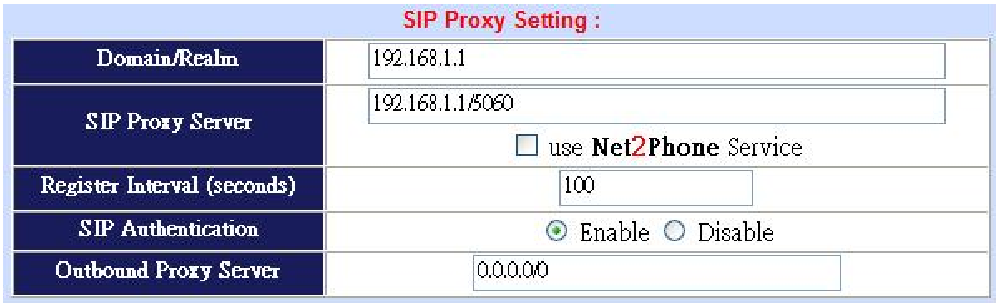



Figure A-10. SIP proxy settings

**STEP 2:**

Because the VIP-281GS have registered to IP PBX, all the VoIP calls will send to IP PBX, so that don't need to set the dial plan settings.

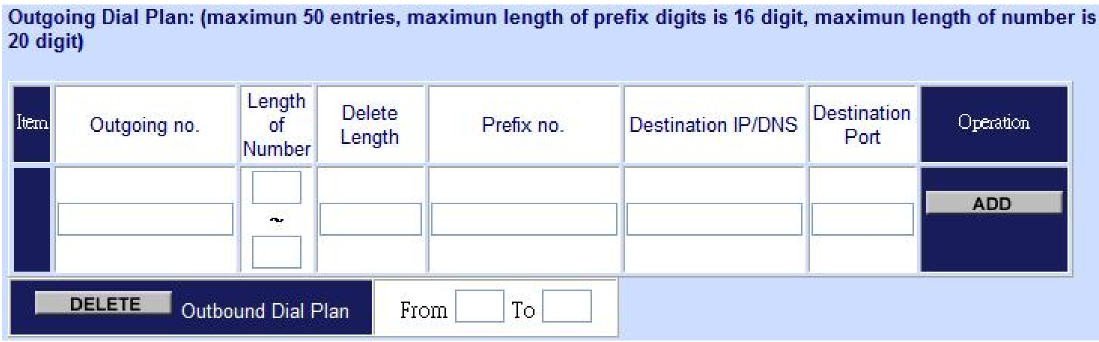



Figure A-11. Outgoing dial plan settings

**STEP 3:**

Please browse to the **GSM Dial plan** menu and set the GSM outgoing number, the sample configuration screen is shown:

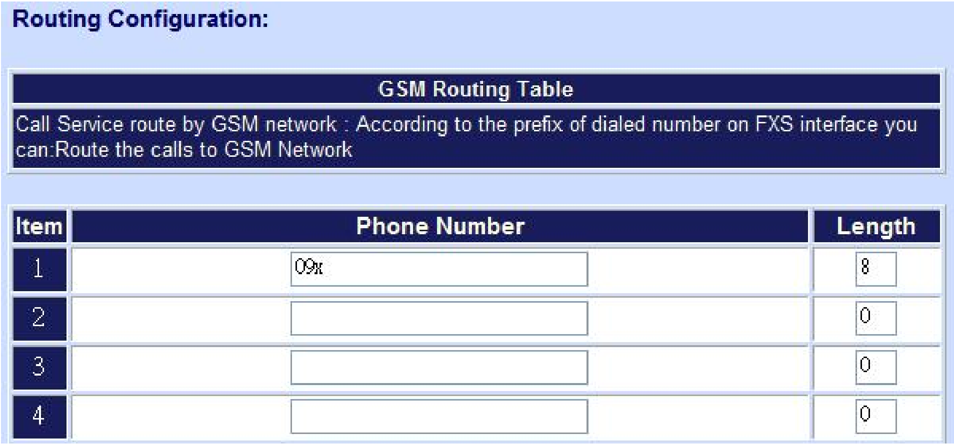



Figure A-12. GSM Routing settings

**STEP 4:**

Repeat the same configuration steps on VIP-281GS_B.

## Test the scenario:

### A. ext.501 call to GSM_3

1. Ext.501 picks up the telephone, and input the trunk code 8 to connect with FXS port of VIP-281GS_A.
2. Dial the GSM number #09125566 to establish the voice communication with GSM_3.

### B. ext.501 call to GSM_4

1. Ext.501 picks up the telephone, and input the trunk code 8 to connect with FXS port of VIP-281GS_A.
2. Dial the ext.400 to GSM port of VIP-281GS_B, and get the dial tone.
3. Dial the GSM number #09581122 to establish the voice communication with GSM_4.

# Appendix B

## FAQ

| |
|---|
| **Q1: 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.<br>**Note:** Default guest login username/password: **guest/guest** |
| **Q2: 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. |
| **Q3: 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. |
| **Q4: What is different [set to default] and [Factory set to default]?** |
| **A:** Factory set to default, you must push RST button until 5 second, and gateway will clear all your setting, and let gateway Wan port become the factory default (172.16.0.1). When you use setting to default by Web or telnet, it will clear all your setting, but the wan port setting will be saved. If you remote the gateway, after set to default, you can login gateway again. No reset the gateway wan port again. |
| **Q5: Why can I call out when the gateway under the NAT?** |
| **A:** VoIP product almost has NAT Pass through problem. By SIP, there are many NAT Pass through Function can solve 80% NAT Problem. You can choose STUN/Outbound Proxy/ Symmetric RTP to Pass through NAT, you don't set any other setting (DMZ/Virtual Server) by router side. If you use STUN/Outbound Proxy, you must have a STUN/Outbound Proxy Server to support. If they can't pass NAT, please open the DMZ/Virtual Server by Router/NAT/Firewall. |
| **Q6: Why does the one way talk happen?** |
| **A:** Generally, one way talk happen when use the different codec between VoIP devices make call. Please check and setting the same codec, most one way talk will be solved. |
| **Q7: Why can I call out by Gateway?** |
| **A:** Please chick your Gateway is registered SIP Proxy Server (ITSP), and chink your Internet works fine. Gateway can't make a call without Internet or SIP Account that from ITSP supply. You must have a SIP account or know the other Gateway IP/Domain Name, and then you can make a VoIP call. |
| **Q8: Why I use asterisk by G.729 sometimes disconnect happen?** |
| **A:** In asterisk setting VAD must disable, if you open Silence Compression (VAD), it will make call disconnect happen, please disable the option when you use the asterisk. |

| Q9: Why can I register and use after setting? |
| --- |
| **A:** After setting, please save configuration and reboot, after reboot you can use new configuration. |

# Appendix C

## 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-VIP281GS_vxxx.bin** ” at the assigned folder in FTP Server.

For example: “**FW-VIP281GS_v305.bin**” is version **3.0.5L**

Note: Free FTP server: 172.16.0.101

username: xxxx, password: xxxx

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


Figure C-1. Firmware upgrades topology

**2. Upgrading Process**

a) Notebook Telnet GSM GW -> open DOS mode ->C:> telnet 172.16.0.1 (Default WAN port IP)

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

```
 Login :

    Welcome to VIP-281GS GSM Gateway (version 3.0.5)
============================================================
                       Main Menu
============================================================
WAN Status:Fixed IP (NAT Mode)
VoIP Status:SIP Direct Mode
GSM Signal Level:-91 dBm
GSM Operator:Chunghwa Telecom
============================================================
 [1] Advanced Setup.
 [2] System Administration.
 [3] Save Current Configurations.
 [4] Upgrade Software.
 [5] Ping.
 [6] Logout.
 [7] GSM.
 [8] Restart.
 Please Select 1 - 8:_
```

Figure C-2. Main menu

c) Please input IP address of FTP server like as: 172.16.0.101, username: xxxx, passswd: xxxx, and image name: **FW-VIP281GS_v305.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 GSM 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 !
```

Figure C-3. Upgrade firmware procedures

# Appendix D

## VIP-281GS Specifications

| | |
|---|---|
| Product | H.323/SIP VoIP GSM Gateway |
| Model | VIP-281GS |
| Hardware | |
| WAN | 1 x 10/100Mbps RJ-45 port |
| FXS | 1 x RJ-11 connection |
| PSTN | 1 x RJ-11 connection |
| GSM | 1 x SIM connection |
| Protocols and Standard | |
| Standard | H.323 v2/v3/v4 and SIP (RFC 3261)<br>SDP (RFC 2327)<br>Symmetric RTP<br>STUN (RFC3489)<br>ENUM (RFC 2916)<br>RTP Payload for DTMF Digits (RFC2833)<br>Outbound Proxy Support. |
| Voice Codec | G.711(A-law /μ-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps) |
| Voice Standard | Voice activity detection (VAD)<br>Comfort noise generation (CNG)<br>G.165/G.168 Echo cancellation<br>Dynamic Jitter Buffer |
| Protocols | SIP 2.0 (RFC-3261), H.323, TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, PPPoE, DNS |
| Advanced Function | Smart QoS, IP TOS (IP Precedence) / DiffServ |
| Network and Configuration | |
| Access Mode | Static IP, PPPoE, DHCP |
| Management | Web, Telnet |
| LED Indications | System: 1, PWR<br>WAN: 1, LNK/ACT<br>Line: 1, In-Use/Ringing<br>Phone: 1, In-Use/Ringing<br>GSM: 1, In-Use/Standby<br>SMS: 1, Transmission |
| Dimension (W x D x H) | 180 x 110 x 25 mm |
| Operating Environment | 0~40 degree C, 0~90% humidity |
| Power Requirement | 12V DC |
| EMC/EMI | CE, FCC Class B |

# EC Declaration of Conformity

For the following equipment:

*Type of Product : 2-Port H.323 / SIP VoIP GSM Gateway
*Model Number : VIP-281GS

* Produced by:
Manufacturer's Name: **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

This product, which has been issued the test report listed as above in QuieTek Laboratory, is based on a single evaluation of one sample and confirmed to comply with the requirements of the following CE/LVD (Low-Voltage Directive; 73/23/EEC) standard.

73/23/EEC relating to electrical equipment designed for use within certain voltage limits and the Amendment Directive 93/68/EEC.

| | |
|---|---|
| ESD | EN 61000-4-2 |
| RS | EN 61000-4-3 |
| EFT/ Burst | EN 61000-4-4 |
| Surge Test | EN 61000-4-5 |
| CS | EN 61000-4-6 |
| Voltage Disp | EN 61000-4-11 |
| EMC (R&TTE, Article 3.1b) | |
| | EN 301 489-1 V1.6.1 |
| | EN 301 489-7 V1.3.1 |
| Radio spectrum (R&TTE, Article3.2) | |
| | EN 301 511 V9.0.2 selection of Test-cases form 3GPP TS 51.010 V7.3.1 |

**Responsible for marking this declaration if the:**

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

**Authorized representative established within the EU (if applicable):**

**Company Name:** **Planet Technology Corp.**

**Company Address:** **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

**Person responsible for making this declaration**

**Name, Surname** **Jonas Yang**

**Position / Title :** **Product Manager**

Jonas

**Taiwan** **20 Oct, 2007**
*Place* *Date* *Legal Signature*