Signamax™ Connectivity Systems

GSM GATEWAY

Model: 065-9066

User Manual

❖ Version: 1.01 –

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PREFACES

0.1 About This Manual

This manual is designed to assist users in using GSM Gateway. Information in this document has been carefully checked for accuracy; however, no guarantee is given as to the correctness of the contents. The information contained in this document is subject to change without notice.

0.2 Copyright Declarations

0.3 Trademarks

Products and Corporate names appearing in this manual may or not be registered tradem arks or copyrights of their respective companies, and are used only for identification or ex planation and to the owners' benefit, without to infringe.

0.4 Safety Instructions

The most careful attention has been devoted to quality standards in the manufacture of the Gateway. Safety is a major factor in the design of every set. But, safety is your responsibility too.

- Use only the required power voltage. Power Input: AC 100-240V, 50-60Hz

- To reduce the risk of electric shock, do not disassemble this product. Opening or removing covers may expose the Gateway to hazardous voltages. Incorrect reassembly can cause electric shock when this product is subsequently used.

- Never push objects of any kind into the equipment through housing slots since they may touch hazardous voltage points or short out parts those could result in a risk of electric shock. Never spill liquid of any kind on the product. If liquid is spilled, please refer to the proper service personnel.

- Use only Unshielded Twisted Pair (UTP) Category 5 Ethernet cable to RJ-45 port of the Gateway.

0.5 Warranty

We warrant to the original end user (purchaser) that the GSM gateway will be free from any defects in workmanship or materials for a period of one (1) years from the date of purchase from the dealer. Please keep your purchase receipt in a safe place as it serves as proof of date of purchase. During the warranty period, and upon proof of purchase, should the product have indications of failure due to faulty workmanship and/or materials, we will, at our discretion, repair or replace the defective products or components, without charge for either parts or labor, to whatever extent we deem necessary to re-store the product to proper operating condition. Any replacement will consist of a new or re-manufactured functionally equivalent product of equal value, and will be offered solely at our discretion. This warranty will not apply if the product is modified, misused, tampered with, damaged by an act of God, or subjected to abnormal working conditions. The warranty does not cover the bundled or licensed software of other vendors. Defects which do not significantly affect the usability of the product will not be covered by the warranty. We reserve the right to revise the manual and online documentation and to make changes from time to time in the contents hereof without obligation to notify any person of such revision or changes.

Note

Repair or replacement, as provided under this warranty, is the exclusive remedy of the purchaser. This warranty is in lieu of all other warranties, express or implied, including any implied warranty of merchantability or fitness for a particular use or purpose. We shall in no event be held liable for indirect or consequential damages of any kind of character to the purchaser.

To obtain the services of this warranty, contact us for your Return Material Authorization number (RMA). Products must be returned Postage Prepaid. It is recommended that the unit be insured when shipped. Any returned products without proof of purchase or those with an out-dated warranty will be repaired or replaced and the customer will be billed for parts and labor. All repaired or replaced products will be shipped by us to the corresponding return address, Postage Paid. This warranty gives you specific legal rights, and you may also have other rights that vary from country to country.

Introduce

GSM Gateway is designed for lowering company telephone bill in calling mobile numbers. This document describes the usage of GSM Gateway.

1.1 Overview

The 065-9066 Quad-Band GSM over VoIP gateway has been designed for user to make calls and receive calls from a cellular phone via the internet using VoIP (SIP/H.323).

1.2 Acronyms Table

Acronym:	Full Name:	Acronym:	Full Name:
ADC	Analog to Digital Converter	CODEC	Coder / Decoder
DAC	Digital to Analog Converter	DC	Direct Current
DDNS	Dynamic Domain Name System	DHCP	Dynamic Host Configuration Protocol
DMZ	Demilitarized Zone	DNS	Domain Name System
DTMF	Dual Tone Multi Frequency	FXS	Foreign Exchange Station
GMT	Greenwich Mean Time	GSM	Global System for Mobile Communications
IP	Internet Protocol	IPsec	Internet Protocol Security
L2TP	The Layer 2 Tunnel Protocol	LAN	Local Area Network
WAN	Wide Area Network	MAC	Media Access Control
MII	Media Independent Interface	NAT	Network Address Translation
NTP	Network Time Protocol	PPTP	Point-to-Point Tunneling Protocol
RTP	Real-Time Transport Protocol	RTCP	Real-Time Transport Control Protocol (also known as RTP control protocol)
SIP	Session Initiation Protocol	SLIC	Subscriber Line Interface Circuit
STUN	Simple Traversal of UDP through NATs	URI	Uniform Resource Identifier
TCP	Transmission Control Protocol	UDP	User Datagram Protocol
UPnP	Universal Plug and Play	VoIP	Voice Over Internet Protocol

1.3 Compare Table

Model Compare Table

Model	*FXS Port*	*PSTN*	*WAN Port*	*VoIP*
065-9066	1	1	1	v

*** manufacture by order (lead time : 60 days)**

1.4 Front Panel LED Indicators & Rear Panels

1.4.1 Outlook of Gateway

Front

Rear

1.4.2 Front Panel LED and Container Descriptions

LED	State	Description
Power	ON	GSM Gateway is Power On
	OFF	GSM Gateway is Power Off
WAN	ON	Network connection established
	Flashing	Data traffic on cable network
	OFF	Waiting for network connection
Line	ON	Line is busy
	Flashing	Ring Indication
	OFF	Line is not enabled
Phone	ON	Telephone Set is Off-Hook
	Flashing	Ring Indication
	OFF	Telephone Set is On-Hook

GSM	On	GSM Network is found and working properly
	Flashing	Searching GSM Network
SMS	ON	Short message waiting Indicator
	Flashing	Sending short message

1.4.3 Rear Panel Descriptions

Port	**Description**
Phone	Phone port can be connected to analog 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.
GSM	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.
Power	A power supply cable is inserted

1.5 Features & General Specifications

065-9066 Common Features and Specifications

Features

- 2-wire, FXS interface (for analog phone or PBX CO line) and PSTN Line
- SMS Server for SMS sending & receiving
- Dialed number restriction, evaluation and modification
- Easy & comfortable maintenance, configuration and upgrade

General Specification

- Compatible with European, US, Brazil and Japan GSM networks (900/1800/1900 MHz)
- SIM: supports SIM card (3V)
- 1 WAN port, 1 FXS port, 1 PSTN port
- Radio interface: Quad-Band EGSM 900/1800/850/1900
- AC power: AC100V-240V, DC12V/1.5A,50/60 Hz
- Temperature: 0°C ~ 40°C (Operation)
- Humidity: up to 90% non-condensing
- Emission: FCC Part 15 Class B, CE Mark
- Dimension: 170 x 100 x 35 mm
- Weight: 200g

Configuration & Management

- Web-based Graphical User Interface
- Remote management over the IP Network
- FTP firmware upgrade
- Backup and Restore Configuration file
- Syslog client support
- Auto-Provision

065-9066

Additional Features

- Calls from cellular over VoIP
- Calls from GSM network to the 065-9066 provides a VoIP dial tone
- Follow me feature for calls from VoIP network.
- Calls that enters FXS port with no answer will be sent to GSM network.

IP Specifications

- 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)
- WAN: Support PPPoE client, DHCP client, Fix IP Address, DDNS client
- Support MWI (Message Waiting Indicator) by SIP Notify.

Call Features

- Voice channels status display
- Direct Dialing Mode : peer to peer call (support IP Address Call or Domain Name Call)
- Register Call Mode : register to SIP Proxy Server or H.323 Gatekeeper
- Adjustable volume : - 9 db ~ 9 db
- Silence Compression / VAD
- Auto Dial for speed
- Dynamic Jitter Buffer
- Hot-Line and Warm-Line Support

Installation and Setup

2.1 Package Content

Please check enclosed product and its accessories before installation. (Refer to the item number). These contents are from pre-released product. The contents for the final product might change a little bit.

Appurtenances:

CD ROM

CD Include in all product user manual and datasheet.

RJ-45 cable

Internet cable RJ-45 connect to NIC/Gateway/Router

Antenna

This Antenna frequency is 900MHz/1800,1900Mhz for automobile.

Power supply

Power Supply,input:100-240V output:+12V (Europe/UK/US)

2.2 Installation

1. Install Gateway

1 Connect the 12V DC IN to the power outlet with power adaptor.

2 Connect Line to PSTN Line.

3 Connect Phone port to a telephone jack with the RJ-11 analog cable (Phone / PBX Trunk Line.)

4 Connect the antenna to the Antenna Connector.

5 Insert SIM card to the gateway

Warning: to avoid the product damaged, please insert SIM card before power-on, and power-off first if it is necessary to take SIM card out of the product.

2. Setting up the network environment for configuration

- To be able to enter the configuration system via web or telnet.

1. Connect the Ethernet cable (with RJ-45 connector) to WAN port.

GSM Gateway ----- RJ45 directly link ----- PC

2. Change the IP address to 192.168.1.2(2~254 is ok)
3. Change the subnet mask to 255.255.255.0
4. Change the gateway and the preferred DNS server to 192.168.1.1

IP configurations above please refer to page 15

3. After Network Configuration is done.

Connecting to an External Ethernet Hub or Switch:

1 Connect the Ethernet cable (with RJ-45 connector) to WAN port.

2 Connect the other end of the Ethernet cable to DSL/Cable modem or the external Ethernet hub or switch.

[Notice: If It's not able to access the GSM Gateway via Internet
Please follow step.2 to enter gateway, the values are special premade settings]

Port	**Description**
Phone	FXS port can be connected to analog telephone sets or Trunk Line of PBX.
Line	Line is used to connect to a PSTN line of carrier.
SIM	After Inserting SIM card ,the gateway is able to work as a mobile phone.
Antenna Connector	Connect the antenna to the connector
WAN	**For Setting** Connect directly to your PC with RJ 45 **For WAN** Connect to the network with an Ethernet cable. This port allows your GW 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.
RES	Push this button until 3 seconds, and GW will be set to factory default configuration.
AC power(DC in 12V)	A power supply cable is inserted

The hardware installation is now complete. The following sections will guide you through setting up your management PC and connecting to the Web User Interface.

2.3 Setup

There are 2 way to setting gateway – **Web User Interface, Telnet**

2.3.1 Factory Default setting

- **WAN** Port IP address : 192.168.1.1
- Default login authentication **username** : **admin, password** : **admin**

065-9066 (VoIP feature)

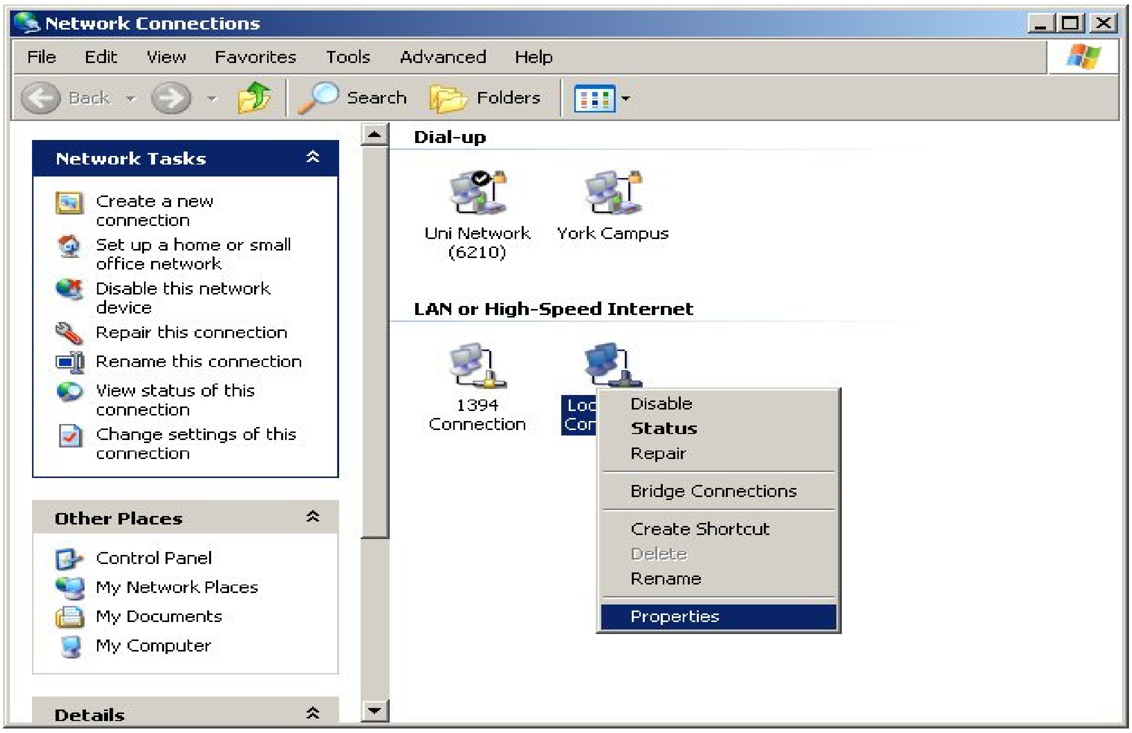
- VoIP Number Port_1~Port_2 number:**100,200**
- VoIP default setting was **H.323** signal protocol, **Direct Mode**, **Fast-Start** and **G.723** codec.

2.3.2 Setting Up Network

Checking the Network IP Configuration

The following explains how to setup the Transmission Control Protocol/Internet Protocol (TCP/IP) in Windows 2000/XP. For more detailed information on TCP/IP setup, refer to the Windows 2000/XP help files. For other operating systems refer to the user manuals.

1. On the desktop, Please enter start -> control panel -> network setting." Click Properties. The Network screen will open.

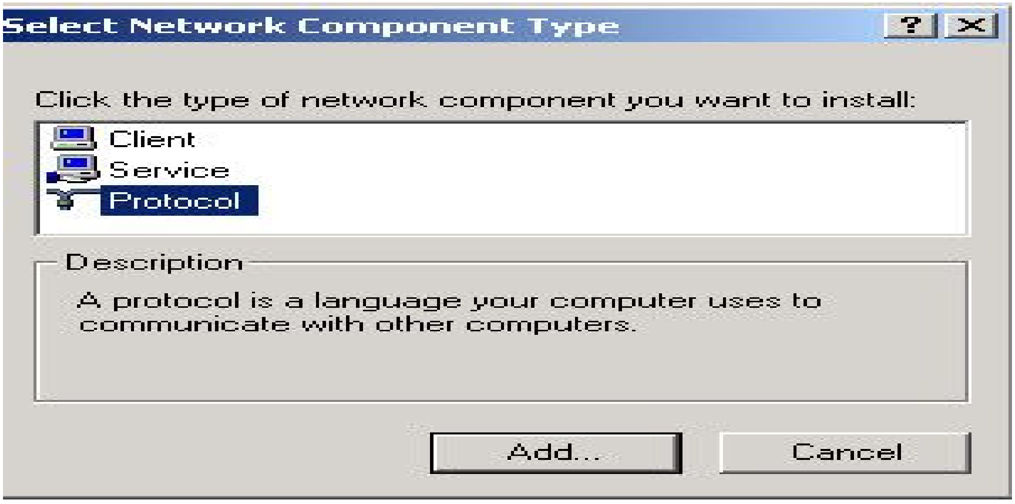


(Your particular system will be different from the screen shown here.)

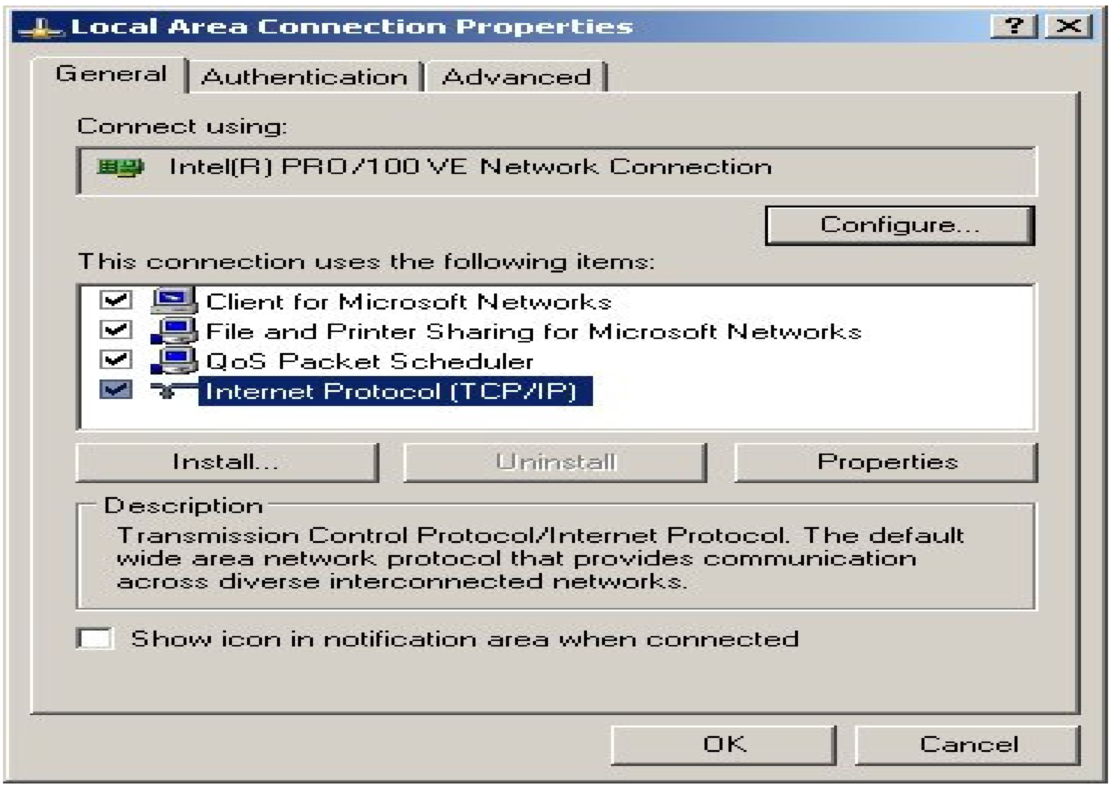
Check that you have an Ethernet network card installed. If not, refer to the card manufacturer's documentation and install the card and drivers.

If your card is installed,

1. Click the Add button. The Select Network Component Type dialog box will open. The box will show four options: *Client, Adapter, Protocol, and Service.*



2. Select Protocol and click the Add button. The Select Network Protocol dialog box will open.
3. Select Microsoft in the left scrolling window then selects TCP/IP in the right, and click OK.". You will be returned to the Network dialog box.



Configuring the TCP/IP Protocol

1. On the Network dialog box Configuration card, select TCP/IP and then click Properties." The TCP/IP Properties dialog box will open.

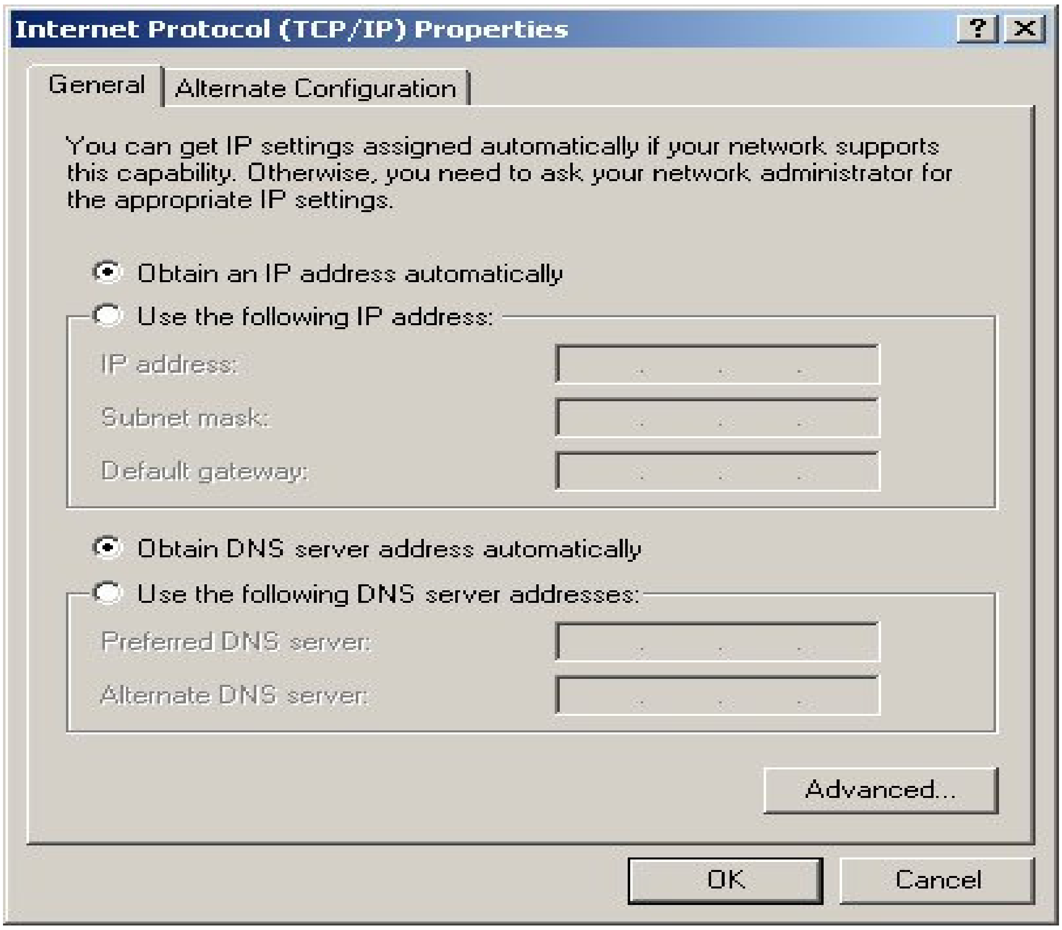
2. On the IP Address tab, Change the IP address to **192.168.1.2**(2~254 is ok) the subnet mask to **255.255.255.0**, the gateway and the preferred DNS server to **192.168.1.1**

5. click OK. A dialog box will pop up asking you to restart the PC. Click Yes".

Checking TCP/IP settings

1. After completing the previous steps, click Start -> Run -> and type ipconfig /all. The IP Configuration window will open. If the PC does not show an IP address in the 192.168.1.2 to 192.168.1.254 range, click the ipconfig /release button to release the current configuration. Wait a few seconds and click "ipconfig/renew" to get a new IP configuration from the router.

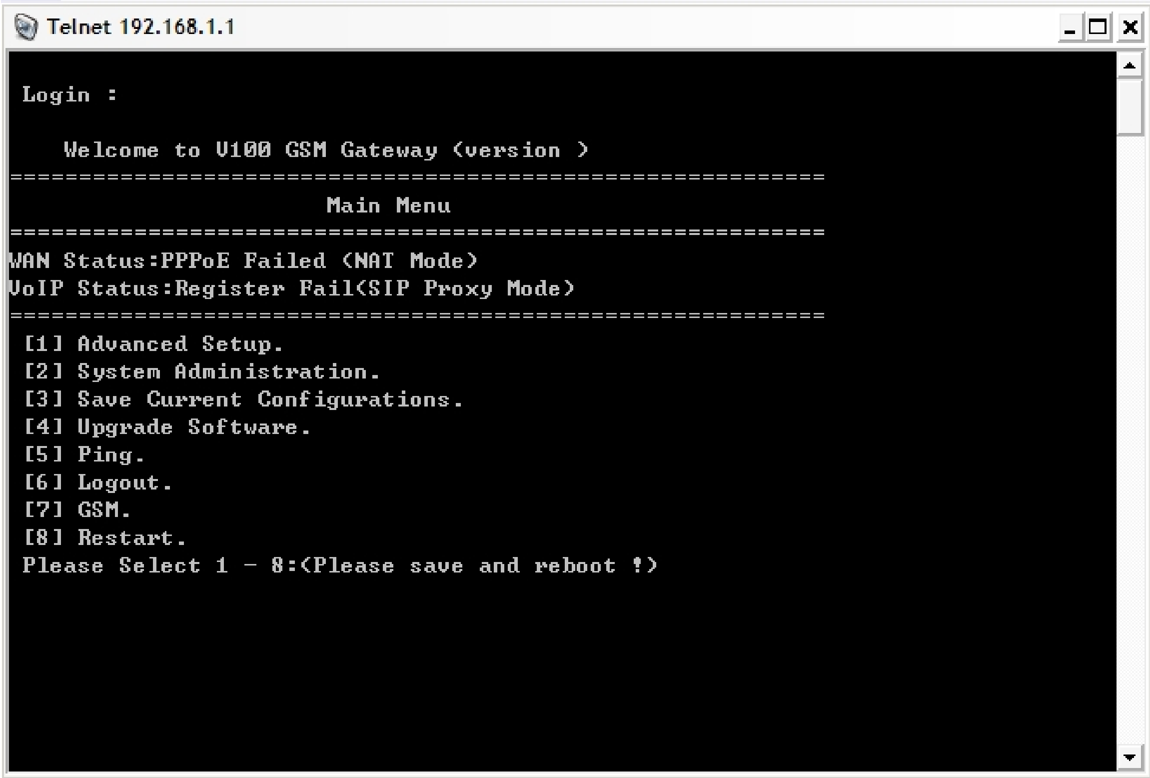
2. If the IP configuration is correct, you will be able to use the PING diagnostic utility built into Microsoft Windows to ping the router. Click Start -> Programs -> MS-DOS Prompt. A command mode window will open.Type "ping 192.168.1.1" (default IP of the router) to check the network connectivity. If both hardware and software are correct, your computer will receive a response from the router as shown on the next page. If not, verify that the Ethernet cable is connected to the router properly and the Ethernet port LED on the front panel is lit.



2.3.2 Telnet

Connect WAN port to Internet or PC and gateway at the same subnet. you can use telnet remote to configure your gateway.

1. Connect Gateway online (Wan)
2. Remote Gateway by Telnet. If telnet successful, you will see Login display.
(For Example: telnet 192.168.1.1)
3. Input Password (Gateway Access password, Default: admin), If login successful, you will enter the welcome display.



4. Gateway Telnet Setting Table, Use 1~9 a~z select setting, “ESC” is back setting.

Item	Setting Option
Main	[1] Advanced Setup. [2] System Administration. [3] Save Current Configurations. [4] Upgrade Software. [5] Ping. [6] Logout. [7] GSM [8] Restart.
[1]Advanced Setup	1.WAN Setting 2.DNS/Dynamic DNS Setting

	3.Network Management 4.VoIP Basic 5.Dialing Plan 6.VoIP Advance Setting 7.Hot Line Setting 8.Port Status 9.Busy Tone Learning a. Show DNS mapping
[1]Advanced Setup ……1.WAN Setting	1.Change WAN Type to DHCP 2.Change WAN Type to Fixed IP 3.Change PPPoE Username 4.Change PPPoE Password
[1]Advanced Setup ……2.NS/Dynamic DNS Setting	1.Change DDNS username 2.Change DDNS password 3.Change DDNS domain name 4.Change DNS server IP 5.Enable/Disable Get DNS Server IP 6.Change DNS server IP
[1]Advanced Setup …3Network Management	1.Change web server port 2.Change telnet server port
[1]Advanced Setup ……4.VoIP Basic	1.Change VoIP Protocol to H.323 2.Change Port Number/Account/Password 3.Enable/Disable Public account 4.SIP hunting setting 5.Change SIP Proxy Server IP Address/DNS 6.Use net2phone 7.Change Register Interval(seconds) 8.Enable/Disable SIP authentication 9.NAT Pass Method a. STUN Server address b. SIP realm c. Outbound Proxy Server address d.Change SIP Local Port
[1]Advanced Setup ……5.Dialing Plan	1.Add Outbound Direct Call 2.Delete Outbound Direct Call 3.Add Inbound Direct Call 4.Delete Inbound Direct Call
[1]Advanced Setup ……6.VoIP Advance Setting	**(1)Sip Advance** 1.Set DTMF Relay Mode 2.Change FAX Mode 3.Enable/Disable VoIP Encryption 4.VoIP Encryption Port Setting **(2)Telephone Advance** 1.VAD(Silence Compression)On/Off

2.Change Codec
3.Enable/Disable UK PSTN Tone Detection?
4.Enable/Disable Dial Complete Tone
5.Dial Termination Key Setting
6.FXS Parameters Setting
1.Change FXS Impedance
2.Change Phone In Volume
3.Change Phone Out Volume
4.Flash Detection
5.Ring Frequency
6.FXS Battery reversal generation

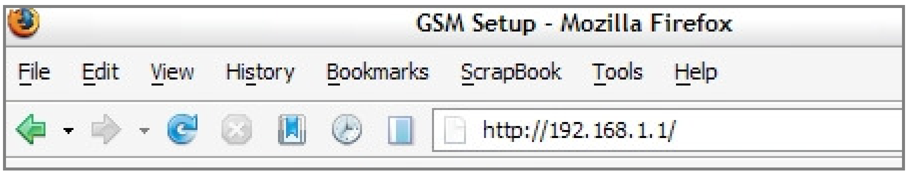
(3)Network Advance
1.Disable Smart QOS
2.Bandwidth Control
3.G.723 Bandwidth
4.G.729 Bandwidth
5.Set IP TOS

[1]Advanced Setup ……7.Hot Line Setting	1.Change Port1 Hot Line Number 2.Change Port2 Hot Line Number…….(To your own port)
[2] System Administration.	1.Save Configuration 2.Access Control 3.Set to Default 4.System Information 5.NTP Setting 6.Syslog Setting

2.3.4 Web User Interface

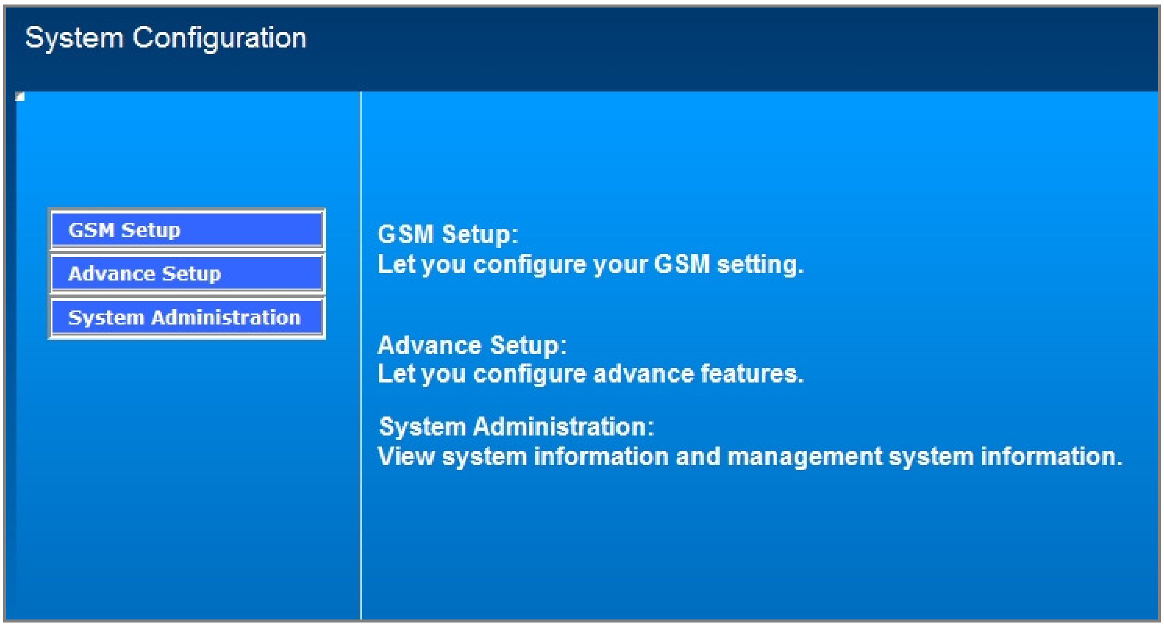
Connecting to the Web Configuration via a Web Browser

1. Launch the Web browser (IE or Firefox). Enter **http://192.168.1.1** into the browser **Address** window and press the Enter Key



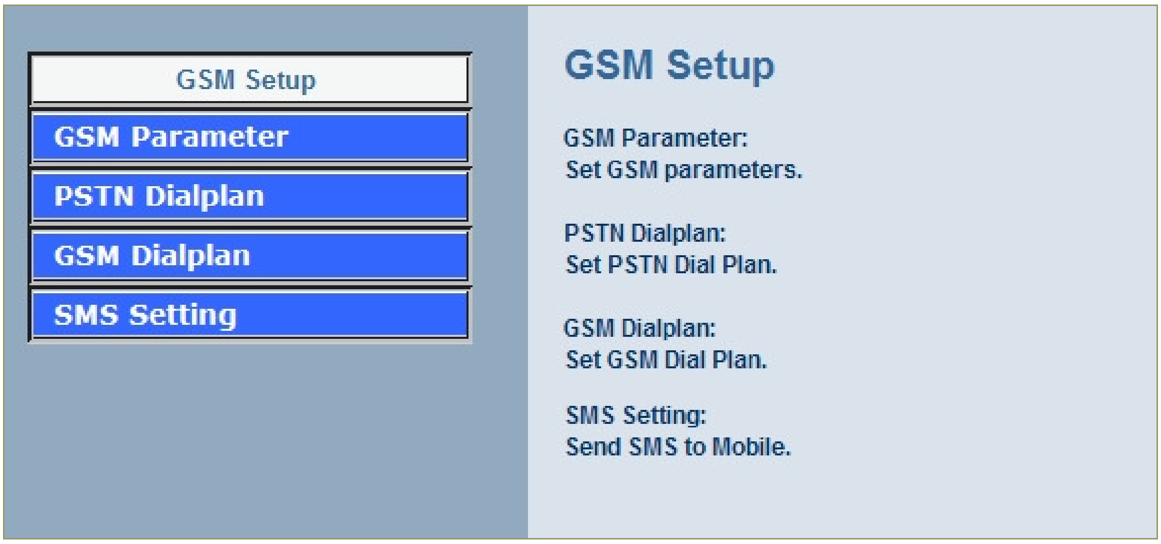
2. An authentication dialog box will open.

3. If this is a first time setup of the router, type "**admin**" as the User Name and the Password field as "**admin**". Click **OK.**(Default username/Password is "admin")
4. The Web Configuration Setup Main Menu will open. On the main page [GSM Setup], [Advanced Setup] and [System Information] were displayed.



GSM SETUP

3.1 GSM SETUP



GSM Parameter	GSM Parameter allows you to modify the option of GSM network.
PSTN Dailplan	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.
SMS setting	The Option is used to send short message to mobile phones

3.1.1 GSM Parameter

GSM Parameter Table Configuration:

GSM Parameter Table	
GSM Parameter table	
PIN Code Protection	○ Enable ⊙ Disable PIN:
Failsafe Mechanism (FXS rely on PSTN)	○ Enable ⊙ Disable
Baby Call	○ Enable ⊙ Disable Delay Time: 0 Phone no.:
Follow Me	○ Enable ⊙ Disable Delay Time: 0 Phone no.:
FXS Battery Reverse	○ Enable ⊙ Disable
Talking Time Limit	0 mins

- **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.
- **Follow ME**: When the calls come to PSTN, 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.

3.1.2 PSTN Dialplan

Routing Configuration:

PSTN Routing Table

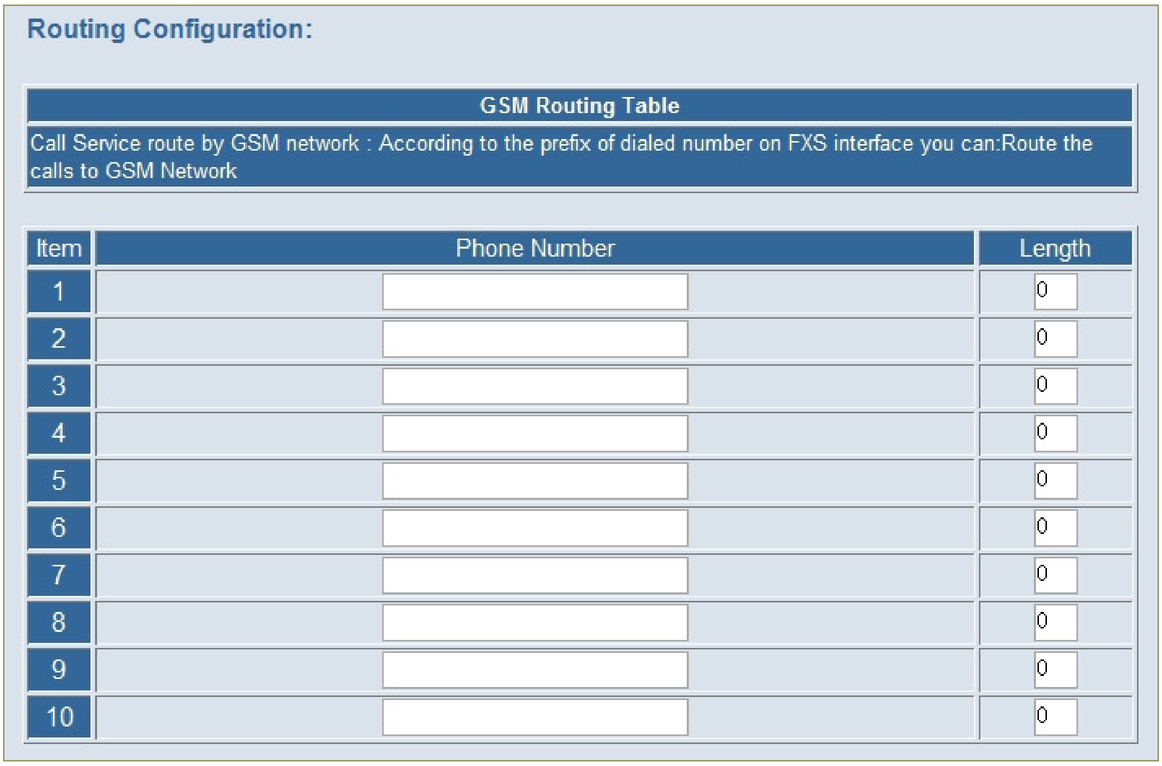
Call Service route by PSTN network : According to the prefix of dialed number on FXS interface you can:Route the calls to PSTN Network

Item	Phone Number
1	1x
2	
3	
4	
5	
6	
7	
8	
9	
10	

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.

For examples:
Emergent calls, like 911
Zone Numbers, like 02x (the phone numbers start with 02)

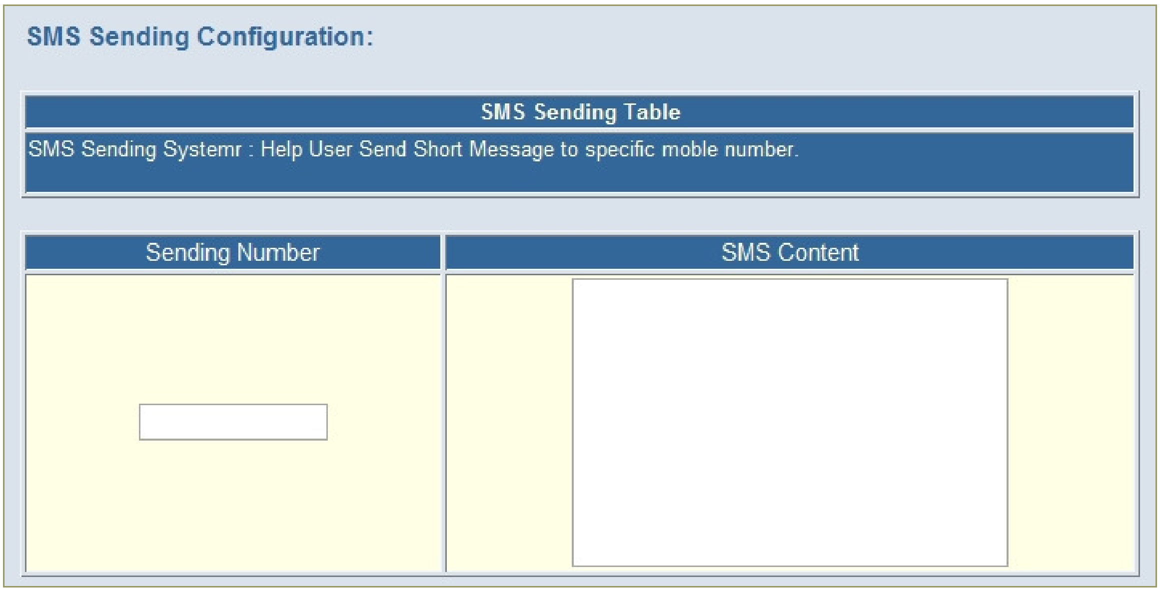
3.1.3 GSM Dialplan



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

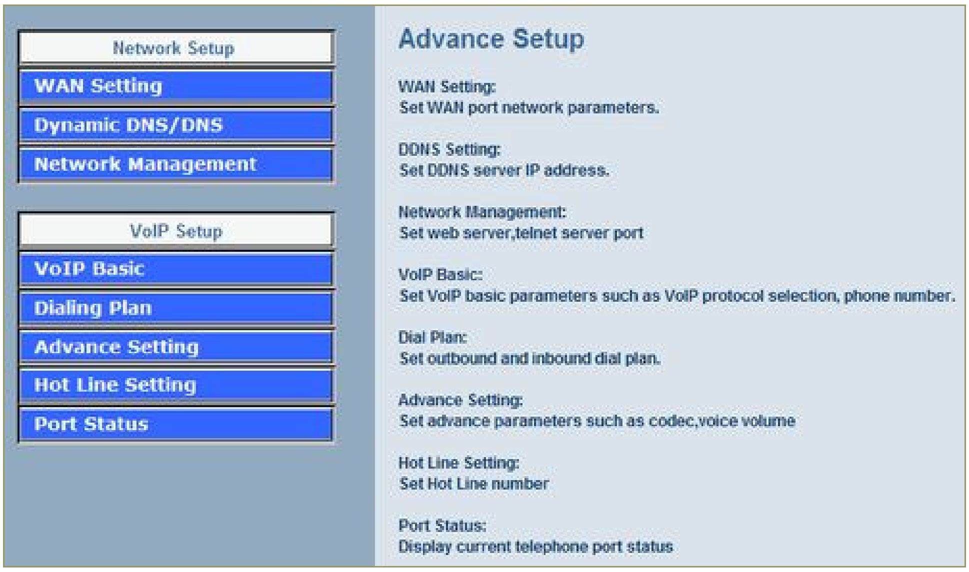
For examples:
09x All telephone numbers start with 09
0919x All telephone numbers start with 0919

3.1.4 SMS Setting



- **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.

Advanced Configuration



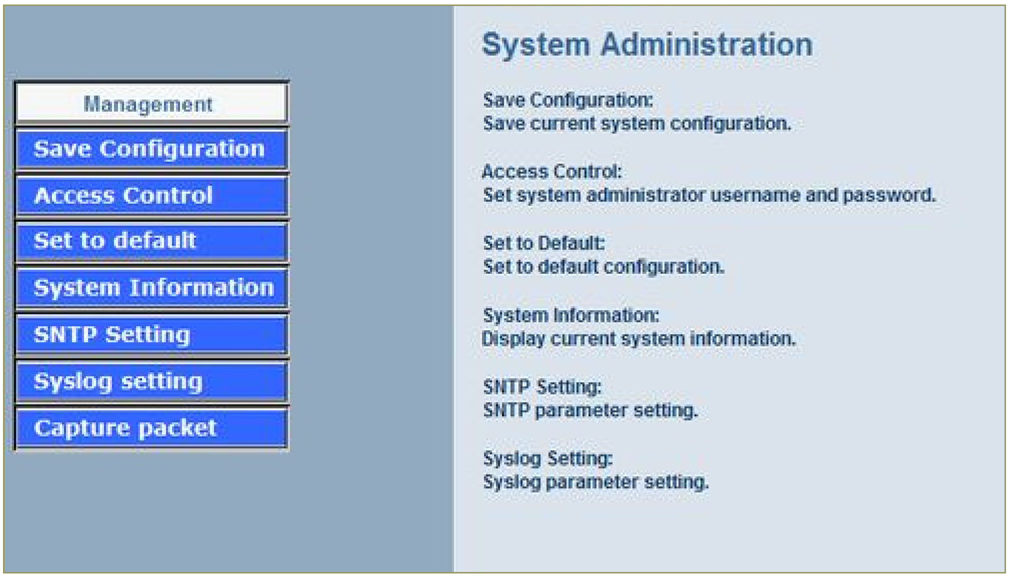
Network Setup

WAN Setting	Sets/changes the WAN port Type like "Fixed IP", "DHCP Client" or "PPPoE".
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. For example : Setting HTTP port : 8080 Setting TELNET port is : 8081 (Default HTTP :80, TELNET: 23)

VoIP Setup

VoIP Basic	The S Series Gateway support 2 / 4 / 8 / 16 / 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	VoIP Gateway support for silence compression, DTMF Relay, Codec Selection, FAX mode Option, H323 Register Type and H.323 Fast-Start/Normal-Start function. 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

System Administration:



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 gateway with the default configuration.
System Information	Display Software version, WAN Type, VoIP Status, VoIP Codec, Phone Interface and System Tim.
SNTP Setting	SNTP (Simple Network Time Protocol) Configuration for synchronizing gateway clocks in the global Internet.
Syslog Setting	Gateway can sends log information to Syslog Server by UDP ports 514.
Capture Packets	The gateway supports packets capture and save the packets to your PC. User can use Network Protocol Analyzer "Ethereal" to analysis the packets. (Free download from http://www.ethereal.com/)

4.1 Network Configuration

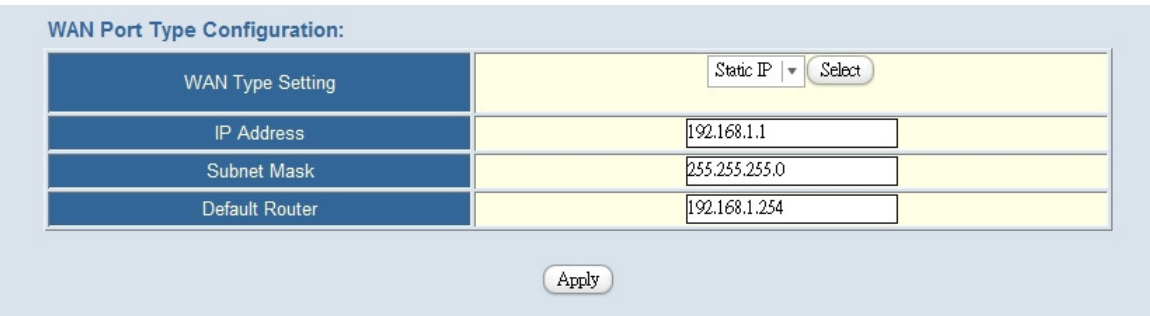
4.1.1 WAN Port Type Setup

For most users, Internet access is the primary application. The **GSM** 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**", the following setup page will be show. Three methods are available for Internet Access.

- **Static IP**
- **PPPoE**
- **DHCP**

Static IP:

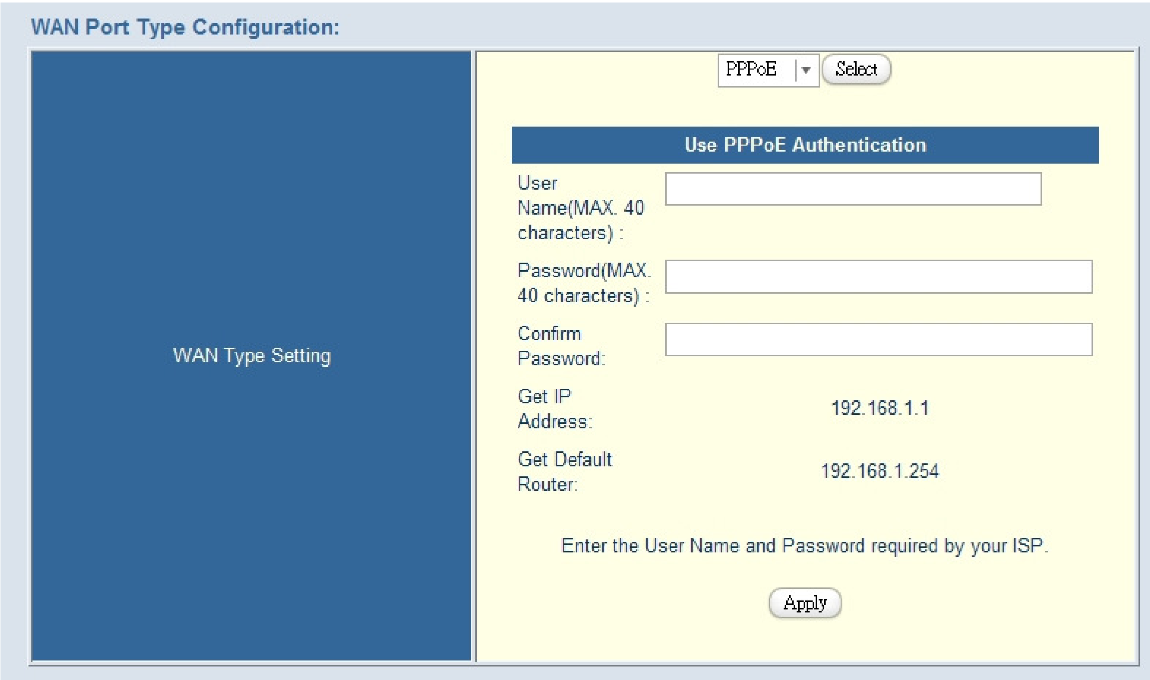
You are a leased line user with a fixed IP address; fill out the following items with the information provided by your ISP.



- **IP Address**: check with your ISP provider
- **Subnet mask**: check with your ISP provider
- **Default Gateway**: check with your ISP provider

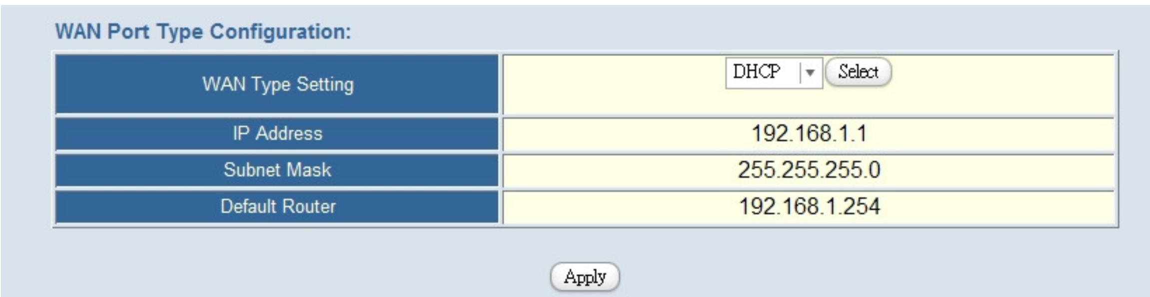
PPPoE for ADSL

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)

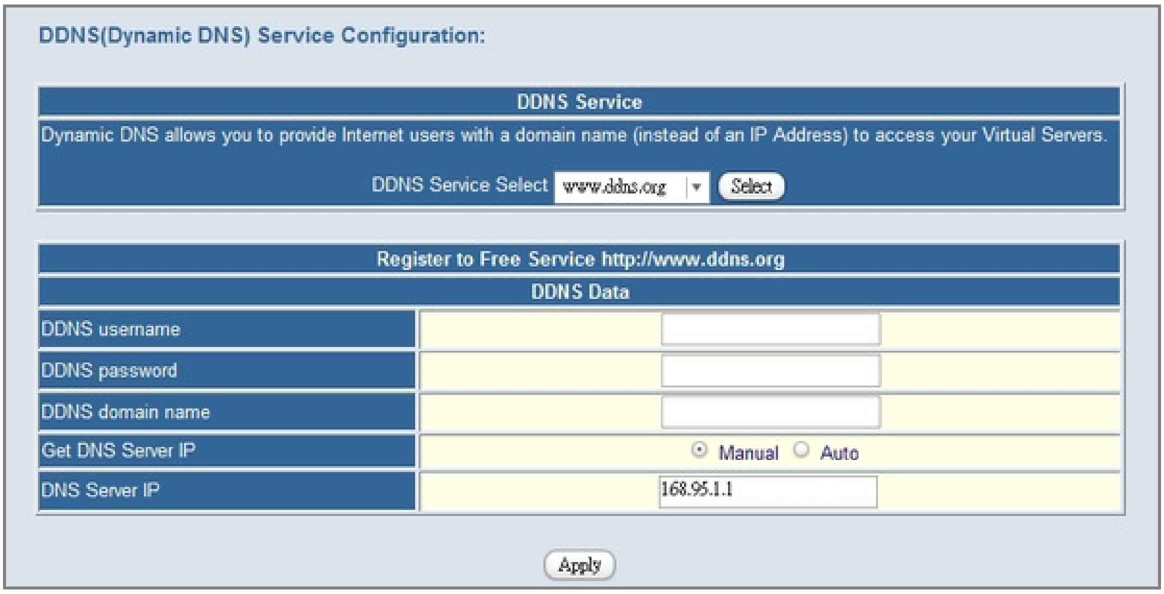
4.1.2 Dynamic DNS

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.

How to use DDNS

First: you should register a new DDNS service account from this web site:
http://www.dyndns.com/newacct

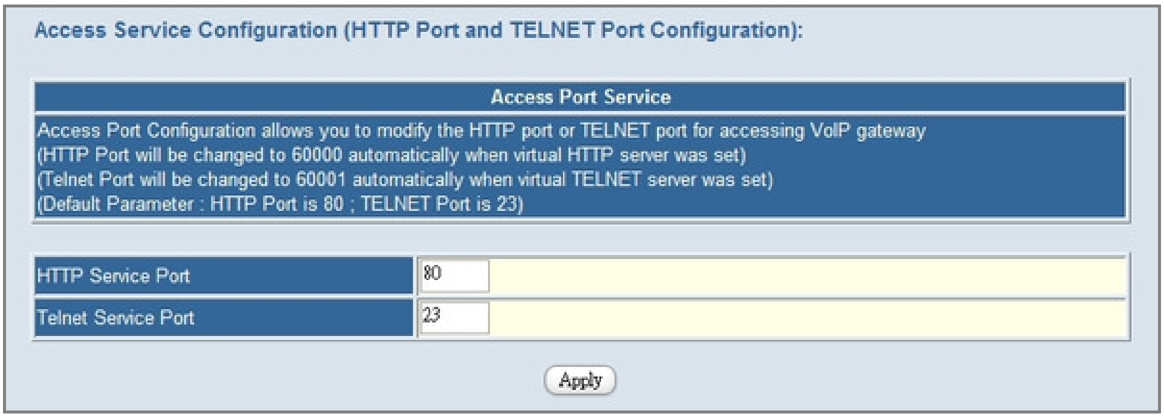
(Attention, if you use static IP address, you can't set DDNS in gateway. Use DDNS and Static IP at the same time, the dyndns will stop your DDNS service. Dyndns support DDNS service is Free, one account can create 5 different DDNS Domain Name)



- ❖ **User Name**: Input your DDNS User Name
- ❖ **Password**: Input your DDNS Password
- ❖ **Domain Name**: Input you set from your DDNS.(ie.gateway.gotdns.com)
- ❖ **DNS Server IP**: Input your DNS Server IP.

4.1.3 Network Management

Network Management,, access port configuration allows you to modify the HTTP port or TELNET port for accessing VoIP gateway
(Default Parameter : HTTP Port is 80 ; TELNET Port is 23)



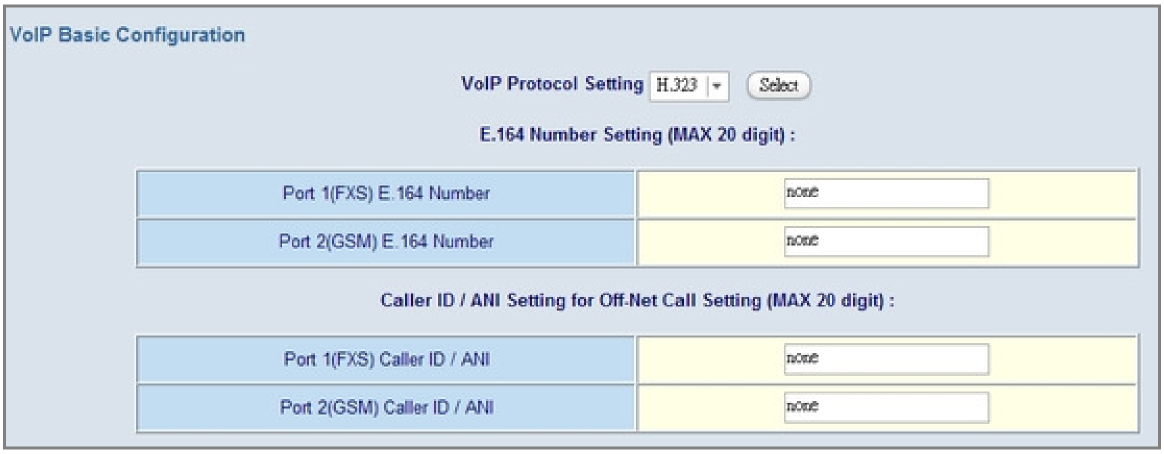
- **Http Server Port:** Input you want to change Web access port (Default is 80)
- **Telnet Server Port**: Input you want to change telnet access port (Default is 23)

4.2 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).

4.2.1 H.323 Setup

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 use H.323.If you want to under NAT, gateway support NAT pass function when you use the same S Series Gateway. Other band gateway doesn't promise this function can work fine!



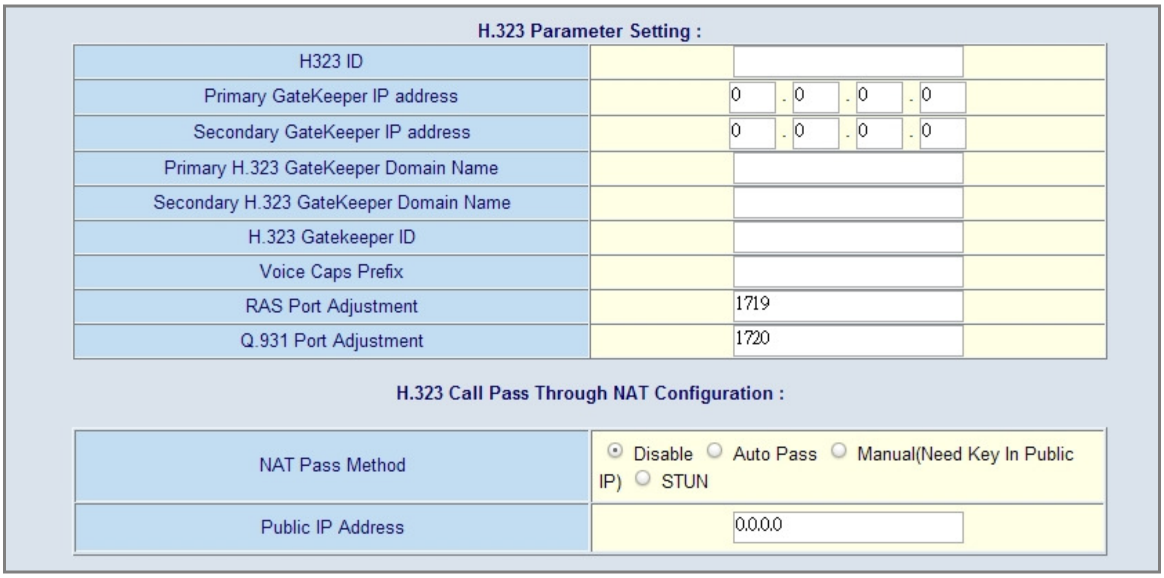
1. Configure the numbering with FXS / GSM ports.

- **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
- (Port number is in comparison with gateway port number. White Port socket is "GSM" port, Black Port socket is "FXS" port.)

2. 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.

4. Register to H.323 Gatekeeper
(If user does not have Gatekeeper, Please go to Dialing Plan Policy)

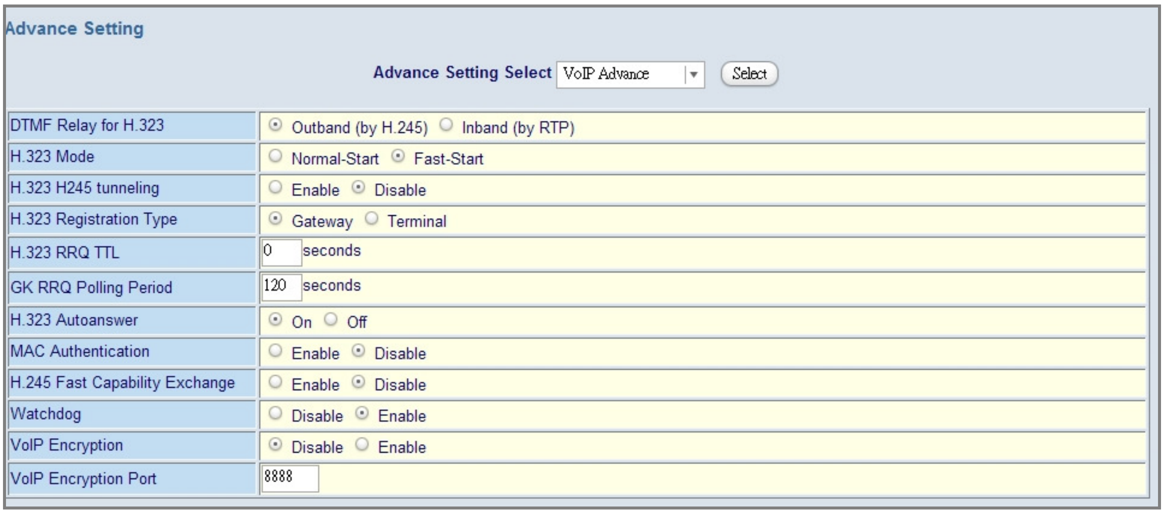


H.323 Parameters Label	
H.323 ID	Sets the unique name of this Gateway, that is communicated as part of H.323 messaging..
Primary Gatekeeper IP Address	There are two gatekeeper address fields, one is primary, the other secondary. If this gateway does not want to register to any gatekeeper, just set value 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 to register to primary gatekeeper, i.e. if both the primary gatekeeper and second gatekeeper addresses are present, the gateway will try to register to these two gatekeepers respectively. The gateway can have the gatekeeper backup function by this way.
Secondary Gatekeeper IP Address	
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 voice cap 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 Pass Through NAT method	1. Disable : The Gateway operates in public IP address 2. Auto Detection: When the Gateway register to GNU Gatekeeper / H.323 Gatekeeper (SK Series), 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.

H.323 VoIP Advanced Configuration

There are many H.323, VoIP, Codec and other more detail Setting, you can set in "**Advance Setting**". For SIP and H.323, there are a little different in advance setting. There are 3 different parts to setting about VoIP, Telephone and network.

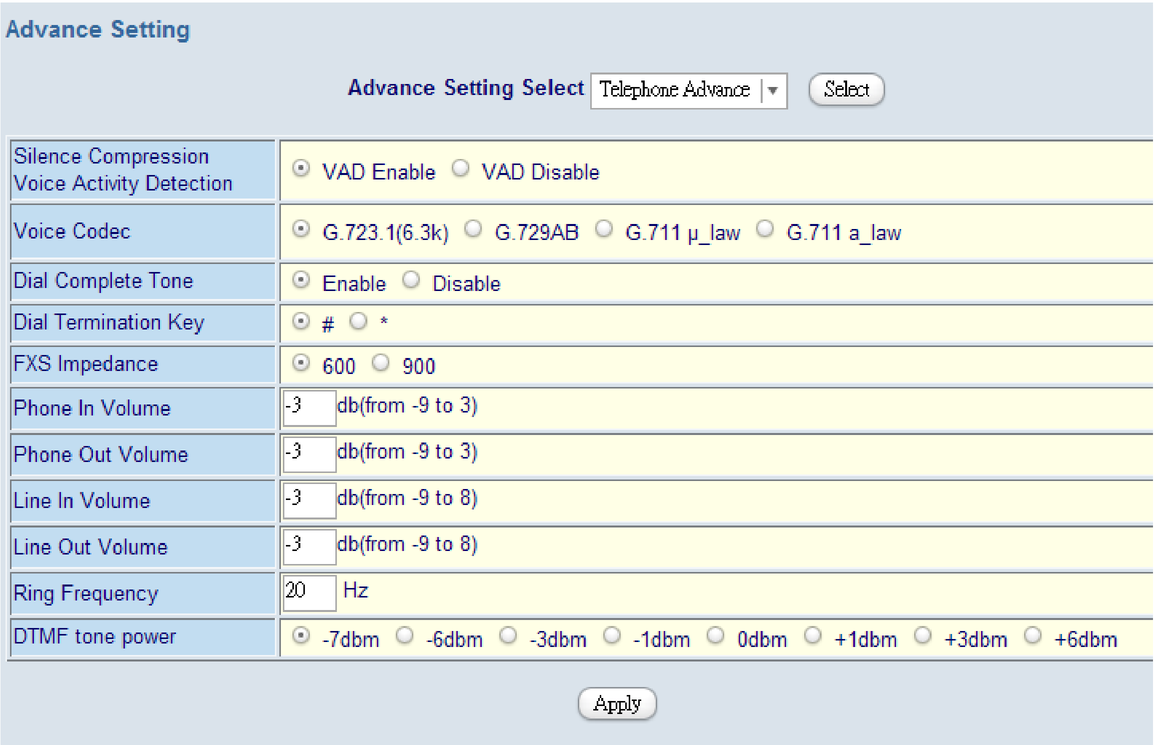
[Advance Setting]



Item	Description
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.

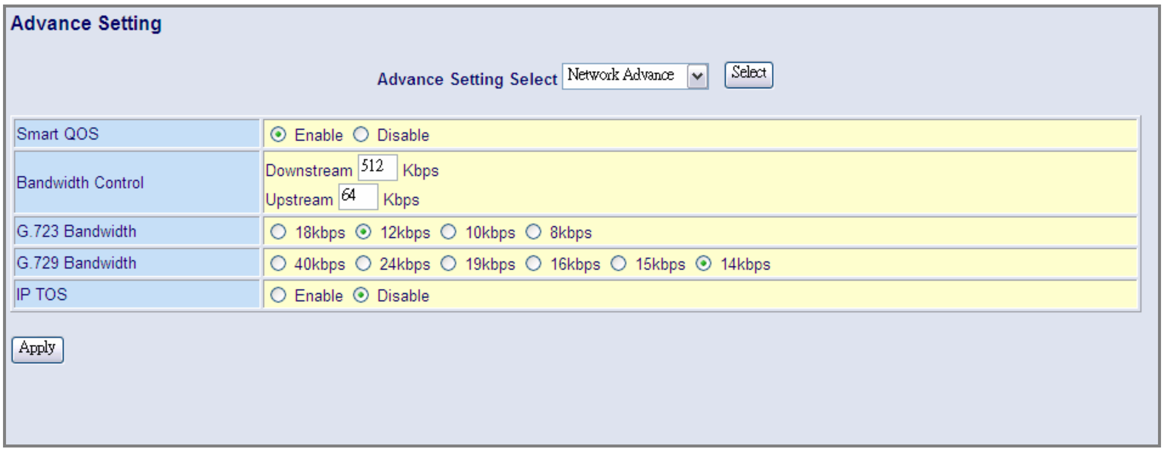
[Telephone Advance]



Item	Description
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.
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.

[Network Advance]

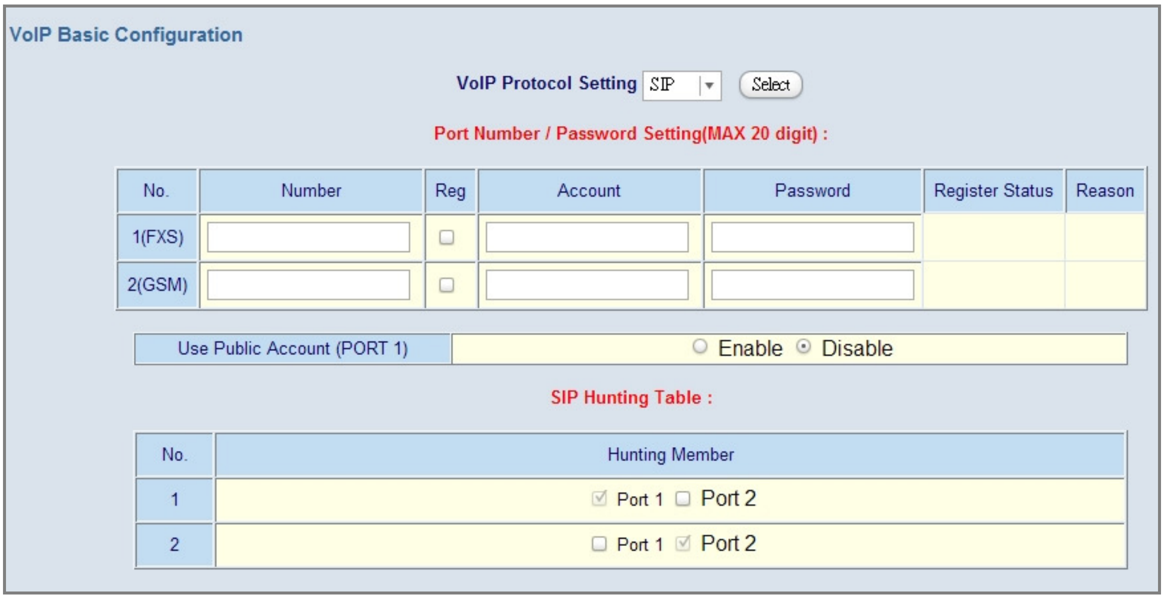


Item	Description
Smart-QoS:	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control:	You can configure your bandwidth what the Max byte of download and upload of 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)

4.2.2 SIP Setup

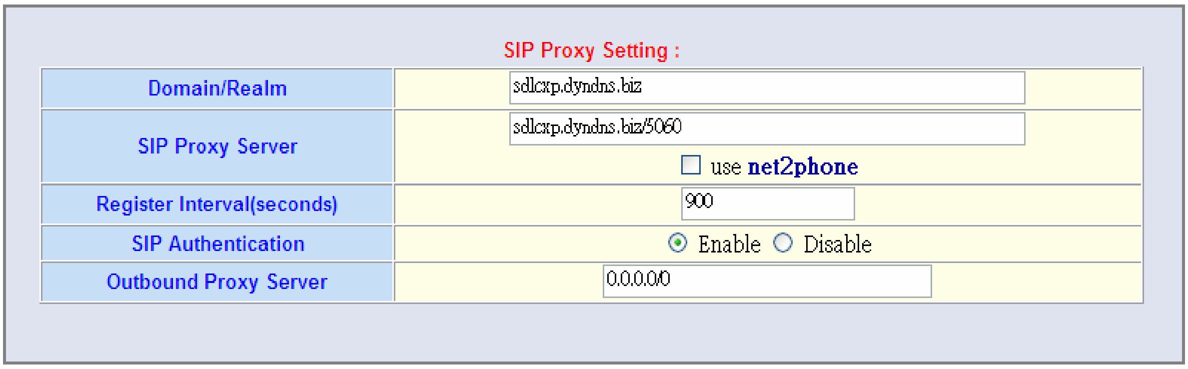
Gateway SIP support SIP(RFC3261), SDP(RFC2327), RFC2833, STUN(RFC3489), Symmetric RTP, outbound proxy, ENUM(RFC2916),and RTP/RTCP.SIP NAT pass through Function can support 80% NAT/Firewall that you don't setting DMZ/Virtual server in router or Firewall.



1. Select "SIP Protocol"
2. SIP number / account (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 server don't support this, number/Account are the same, please input the same username)

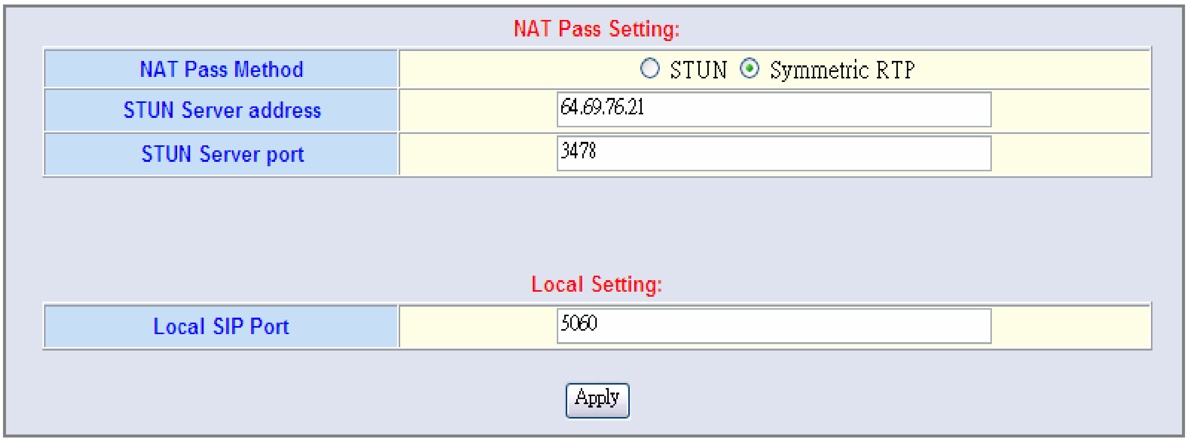
- **Number**: Input SIP Number(Username), if your server support account and number (different),input the number, else number/account are the same username.
- **Reg**: let your sip account register SIP Server, click this option.
- **Account**: Input SIP account(Username), if your server support account 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.

3. SIP Proxy Server setting, setting SIP proxy server register information.
(If user does not need register SIP Proxy Server, Please go to Dialing Plan Policy)



SIP Proxy Server Label	
SIP Proxy Server Setting	1. 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). 2. Use Net2Phone Service Provider
SIP Domain	1. Enter the SIP realm in this field
Register Interval Setting	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

4. 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.

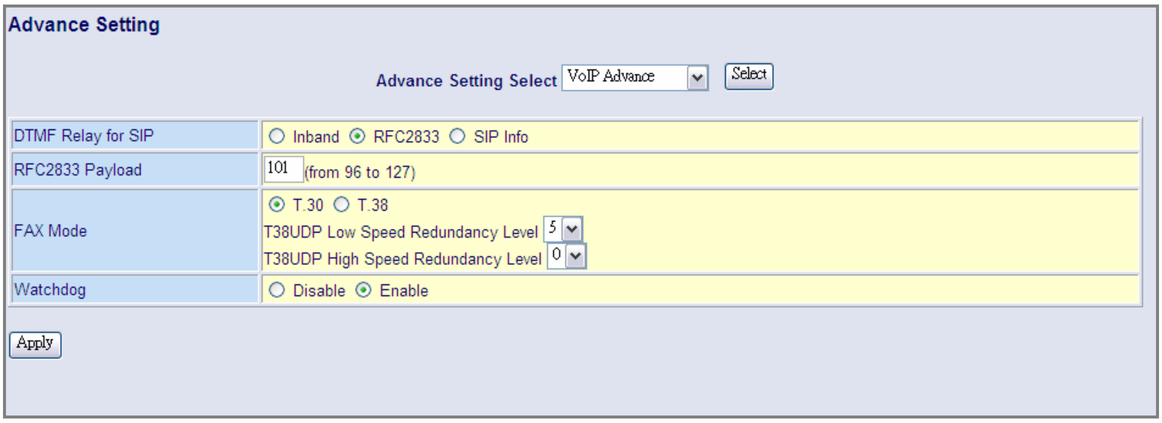


- **Symmetric RTP**: default use Nat pass function.
- **STUN Client**: setting your STUN server information, default STUN server is FWD STUN server.
- **Outbound Proxy Support**: Setting your Outbound Proxy server information.
- **Local SIP Port**:: setting local use SIP port, default is 5060.

SIP VoIP Advanced Configuration

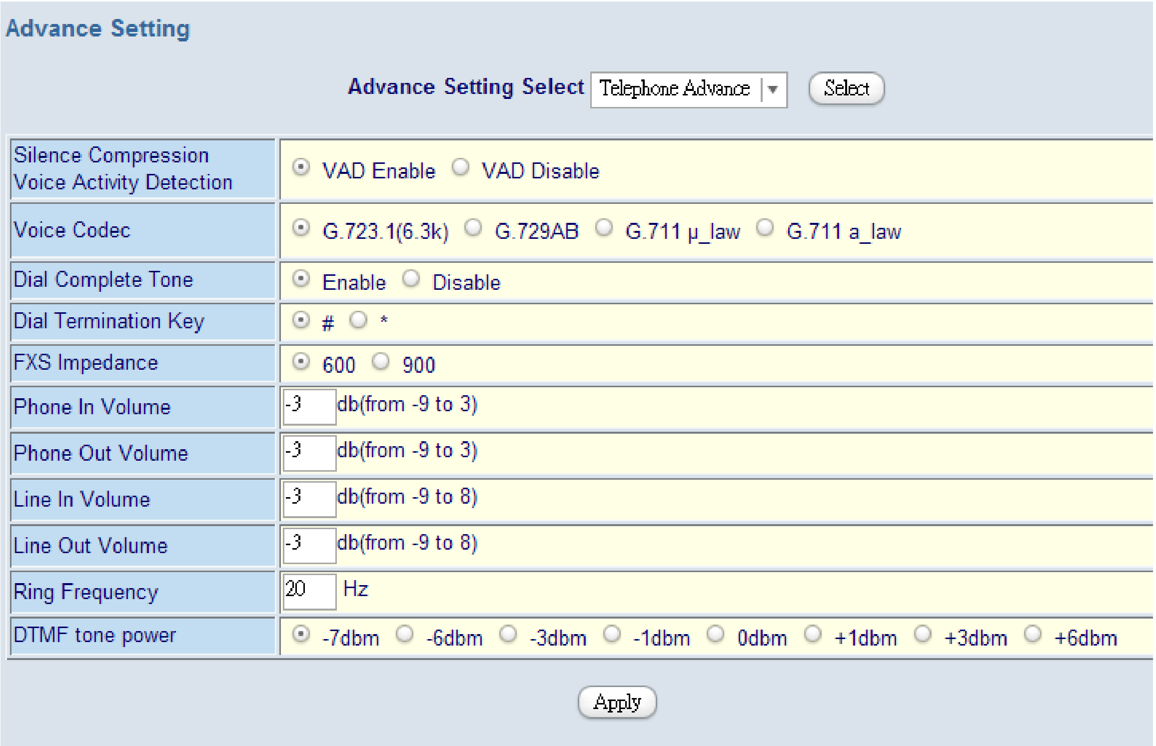
There are many SIP VoIP, Codec and other more detail Setting, you can set in "**Advance Setting**". For SIP and H.323, there are a little different in advance setting. There are 3 different parts to setting about VoIP, Telephone and network.

[VoIP Advance]



Item	Description
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.
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 (T.38 mode isn't support all gateway, different band use T.38 have a little change, it maybe let T.38 FAX Error)
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.

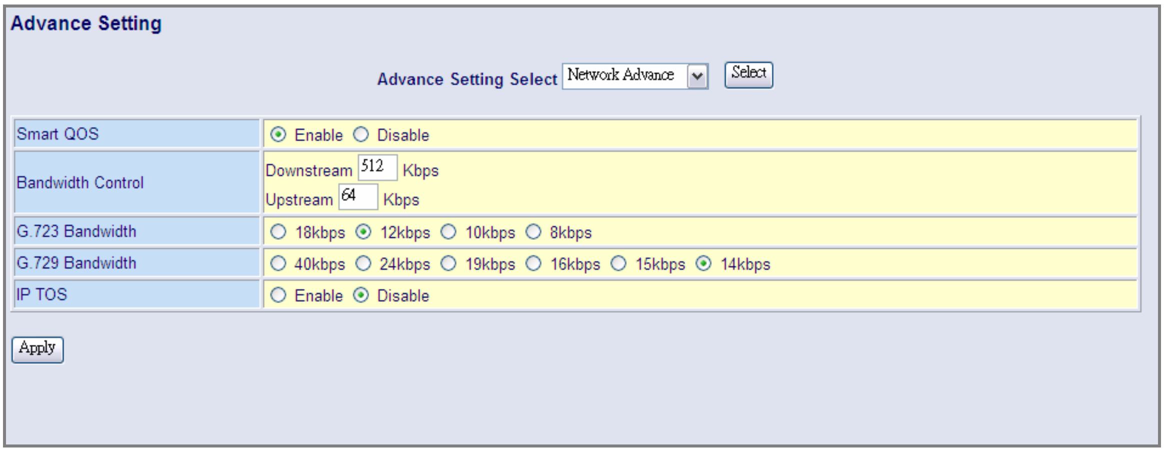
[Telephone Advance]



Item	Description
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.
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.

[Network Advance]



Item	Description
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)

4.2.3 Direct call (Peer to Peer) setup

If you don't registered Gatekeeper or SIP proxy server, you can make call by Peer to Peer. For SIP or H.323, setting the dialing plan, and can make direct call.

Overview of the Dialing Plan

The "**Dialing plan**" need setting when the user use the method of Peer-to-Peer H.323 (SIP) VoIP call or registering H.323 Gatekeeper (SIP Proxy Server) Mode. The H.323(SIP) Dialing Plan has two kinds of directions: Outgoing (call out) and Incoming (call in).

1. **Outgoing Dial Plan:**

   Peer-to-Peer Call Mode: Effective

   Registering to H.323 Gatekeeper (SIP Proxy Server) Mode: Effective

2. **Incoming Dial Plan:**

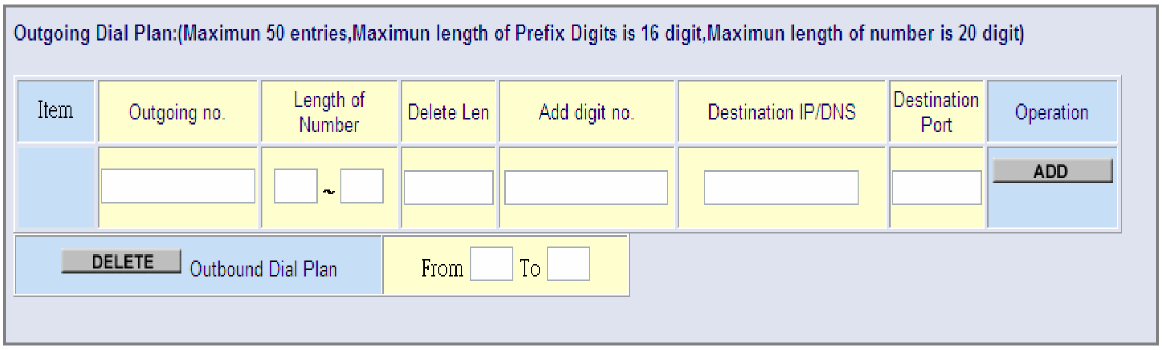
   Peer-to-Peer Call Mode: Effective

   Registering to H.323 Gatekeeper (SIP Proxy Server) Mode:

   The leading number would register to H.323 Gatekeeper (SIP Proxy Server)

When you use direct call, you must setting your VoIP protocol firstly. Use direct call, you should setting the same protocol both of UA. Both of UA must support dial plan function. Some ATA don't support Dialing plan, it maybe let direct call failed.

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 port of the destination gateway use.(Default is 5060)

Example1: Normally Dial

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 Len	Add digit no.	Destination IP/DNS	Destination Port	Operation
1	001x	4 ~ 20	0	None	215.214.1.1	5060	
2	002x	4 ~ 20	0	None	h323.gw.net	5060	
		~					ADD

DELETE Outbound Dial Plan	From To

1.001x leading call out, call to Destination IP address: 211.22.3.14
2.002x leading call out, call to Destination Domain Name: h.323.gw.net

Example2: Speed Dial

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 Len	Add digit no.	Destination IP/DNS	Destination Port	Operation
1	401	3 ~ 3	3	1334588712	211.22.3.14	5060	
2	402	3 ~ 3	3	2212345612	211.21.2.76	5060	
		~					ADD

DELETE Outbound Dial Plan	From To

1. If user dial "401",
Gateway automatically dial "1334588712" to Destination IP address: 211.22.3.14
2. If user dial "402",
Gateway automatically dial "2212345612" to Destination IP address: 211.21.2.76
In the "Incoming Dial Plan Configurations" settings: Maximum Entries : 50

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 Len	Add Digit no.	Destination tele port	Operation
		~				ADD

DELETE Inbound Dial Plan	From To

- "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 "Tel-port"; this is for local dial plan setting phone number.

4.2.4 Other VoIP Setting

Hot Line:

You can setting hot line. when the call incoming the hot line port, it will call hot line number automatically. The hot line call 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.

Hot Line Number Setting (Hotline Setting)

Hotline Delay	⦿ Disable ○ Enable
Hotline Delay Time(Max. 20 sec)	3 sec
Port 1 number	None
Port 2 number	None

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

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:

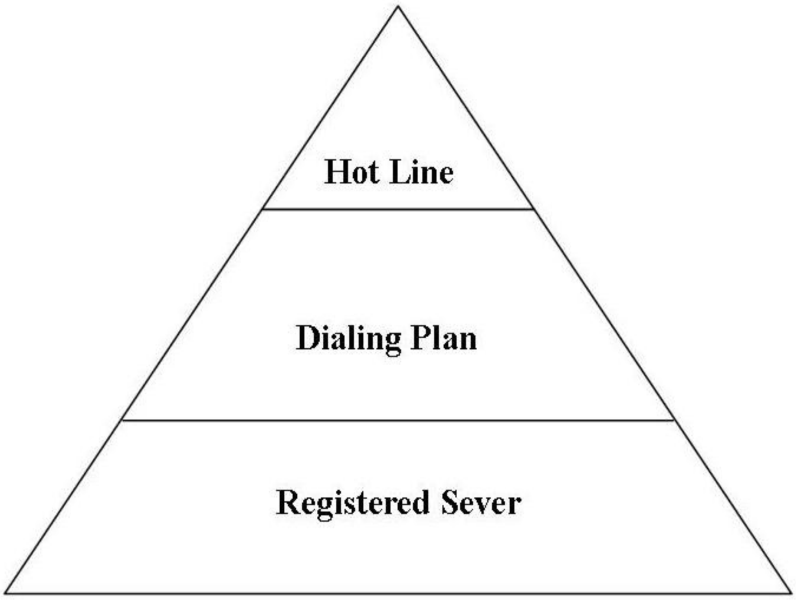
Port No.	Type	Status	Codec	Direction	Dial No.	Caller No.	Dest/Source	IN	OUT	Duration
1	FXS	onhook	none	none	none	none	none	0	0	0
2	GSM	onhook	none	none	none	none	none	0	0	0

❖ 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.

Call Priority:

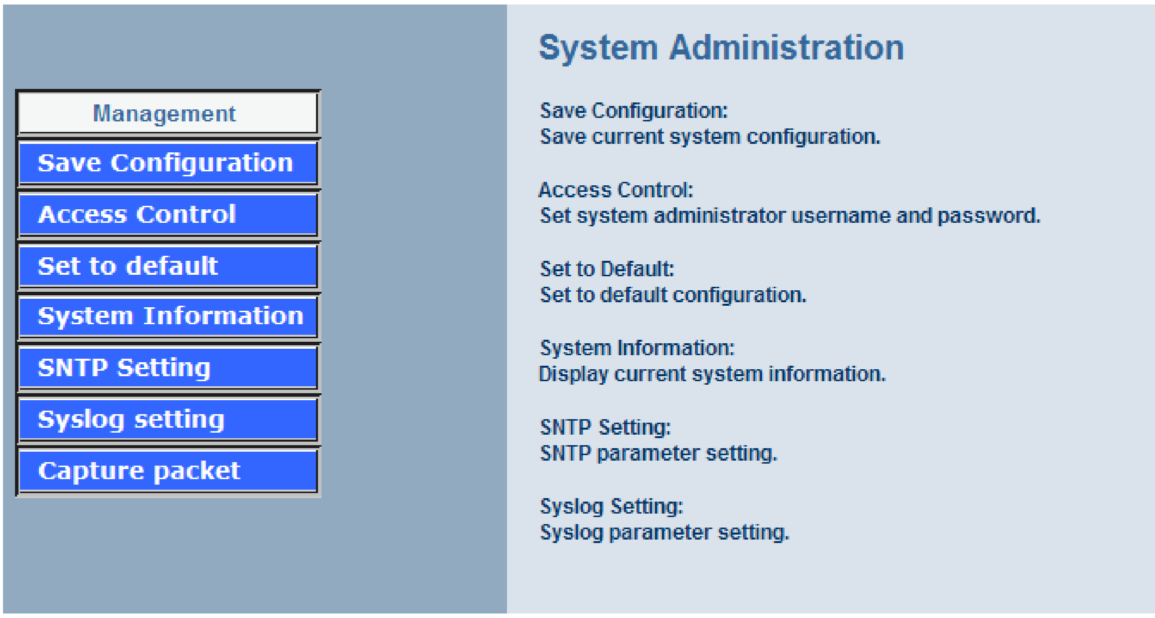
Gateway have a rule for call Priority, up to down is 1)Hot Line 2)Dialing plan 3)Registered server(SIP Proxy / H.323 Gatekeeper).When a VoIP call made, Gateway will process by Hot Line first, then it will check the dialing plan table, last fine Server(SIP/H.323).

For example, if I have a gateway , and It is registered a proxy server, I don't setting any others (Hot Line or dialing plan.). when I make a VoIP call, gateway will check Proxy server. Now,, I setting 1~2 dialing plan, and registered proxy server. When I call, gateway will check the dialing plan first, then find the proxy server. And so on…..

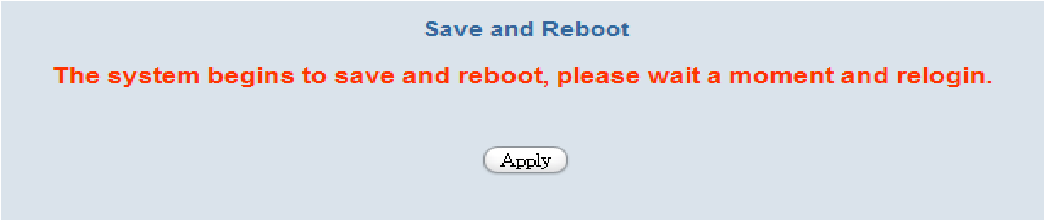


System Administrator

You can setting other gateway setting, like gateway time, Syslog that send CDR information to Syslog server, backup and restore configuration.

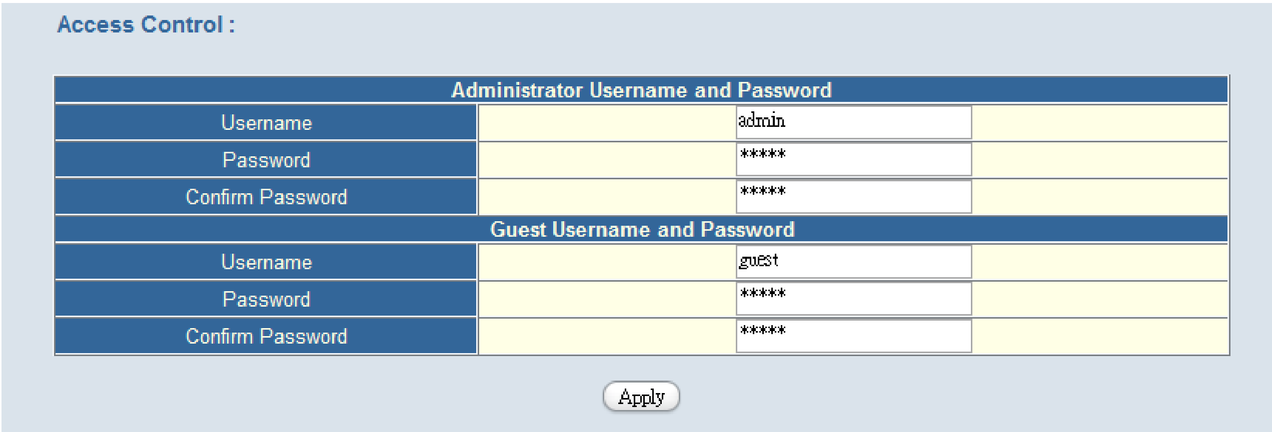


4.3.1 Save Configuration and Reboot



❖ Click "**Save Configuration and Reboot"** to save configuration and begin to restart. (When you set done, select "Reboot" option will auto save and reboot!)

4.3.2 Access Control



❖ **Changing the Administrator Password**

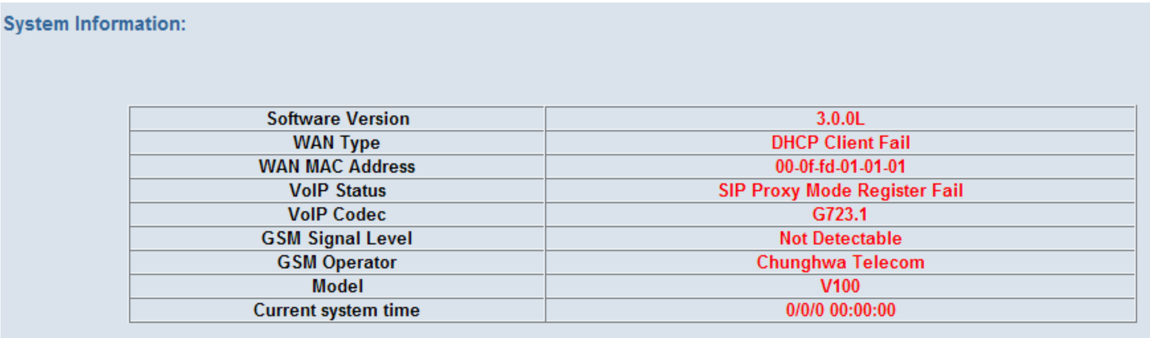
For security reasons, we strongly recommend that you set an administrator. password for the router. On first setup the router requires no password. If you don't set a password the router is open and can be logged into and settings changed by any user from the local network or the Internet.

❖ Click **Access Control Setup**, the following screen will open.
(Guest account , if you use guest account login , you only can view gateway setting, not change and configure any gateway setting, else you login by Admin account)

4.3.3 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.

4.3.4 System Information Display Function



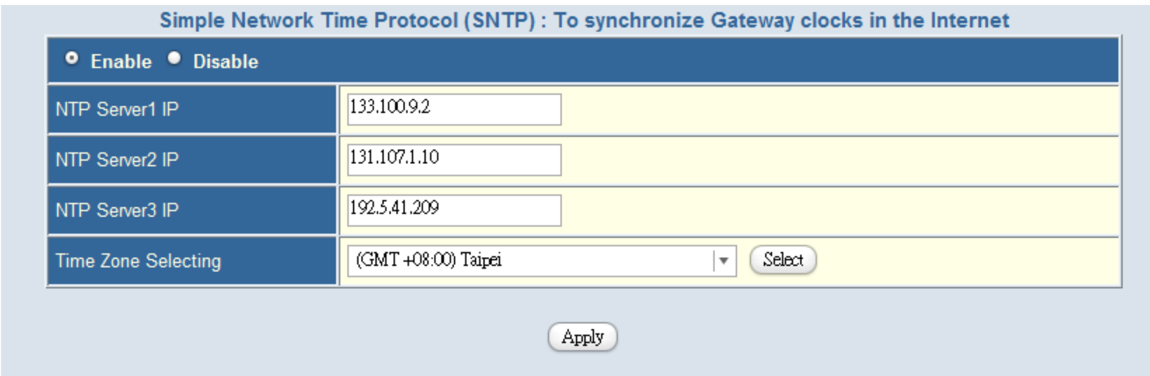
System Information:

Software Version	3.0.0L
WAN Type	DHCP Client Fail
WAN MAC Address	00-0f-fd-01-01-01
VoIP Status	SIP Proxy Mode Register Fail
VoIP Codec	G723.1
GSM Signal Level	Not Detectable
GSM Operator	Chunghwa Telecom
Model	V100
Current system time	0/0/0 00:00:00

❖ Click System Information Display to open the Online Status page. In the example, on the following page, both PPPoE connection is up on the WAN interface, H323 Status, MAC address, Register Status, etc.…

4.3.5 SNTP Setting Function

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



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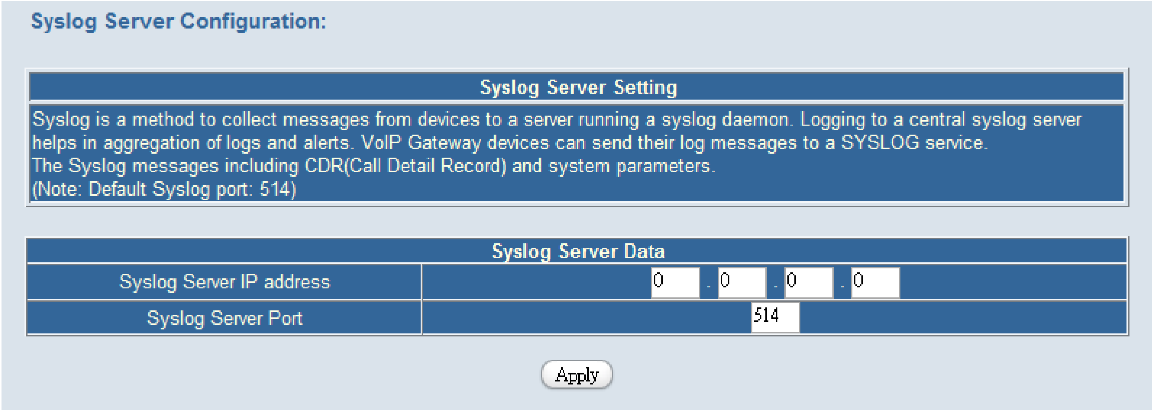
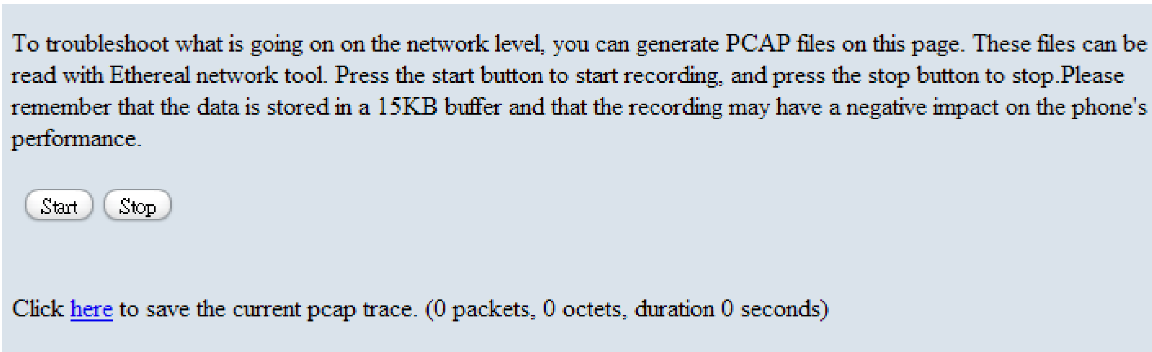
Use SNTP Setting—When checked, Gateway uses a Simple Network Time Protocol (SNTP) to set the date and time . The Gateway synchronizes the Gateway’s time after you select the time zone. *Use SNTP Setting*, Select the time zone which Gateway was at.

4.3.6 Syslog Setting Function



- Use Syslog server to record your Gateway log file. you can setting you syslog server IP address for this function. Syslog information include the CDR source!

4.3.7 Capture Packets Function



- Use "Capturer Packets" to record Gateway packets. You can start and stop the capture then save the file to PC Use the Ethereal Tool (www.ethereal.com) to analyze the packets.

(if gateway have interoperability problem, you can capture the packet, send to us . we can refer this packet to bebug.)

4.4 Update firmware

Gateway can upgrade Firmware via FTP, update firmware can add new function or fix some bug. If your gateway works fine, you don't need update any new firmware. The new firmware maybe let your gateway not stable. you can get the last version firmware on our web site or send support mail to us, we will mail firmware to you.

Firmware name is "v100.300", the first name v100 is mean the gateway module.
(Gateway update firmware only support use **telnet** via **FTP,** no other else upgrade function.)

FTP upgrade Requirement and Process

1. Environment Requirement

- PC with FTP Server (Server-U software, 3CDaemon,..)
- PC or Notebook witch connected to WAN port of Gateway.
- Put the image (firmware) named “v100.xxx” at the assigned folder in FTP Server.

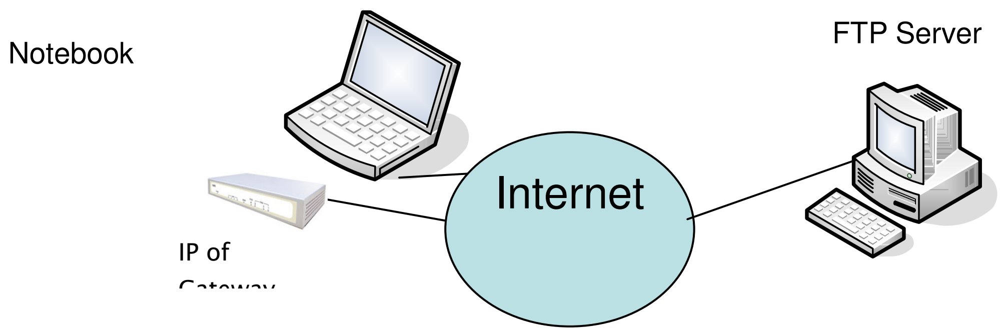
(for example: “v100.270” is version 2.7.0)

Note: Our company FTP server, you can use it to upgrade

Free FTP server : 61.218.109.83

username: share , password: 19730809

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



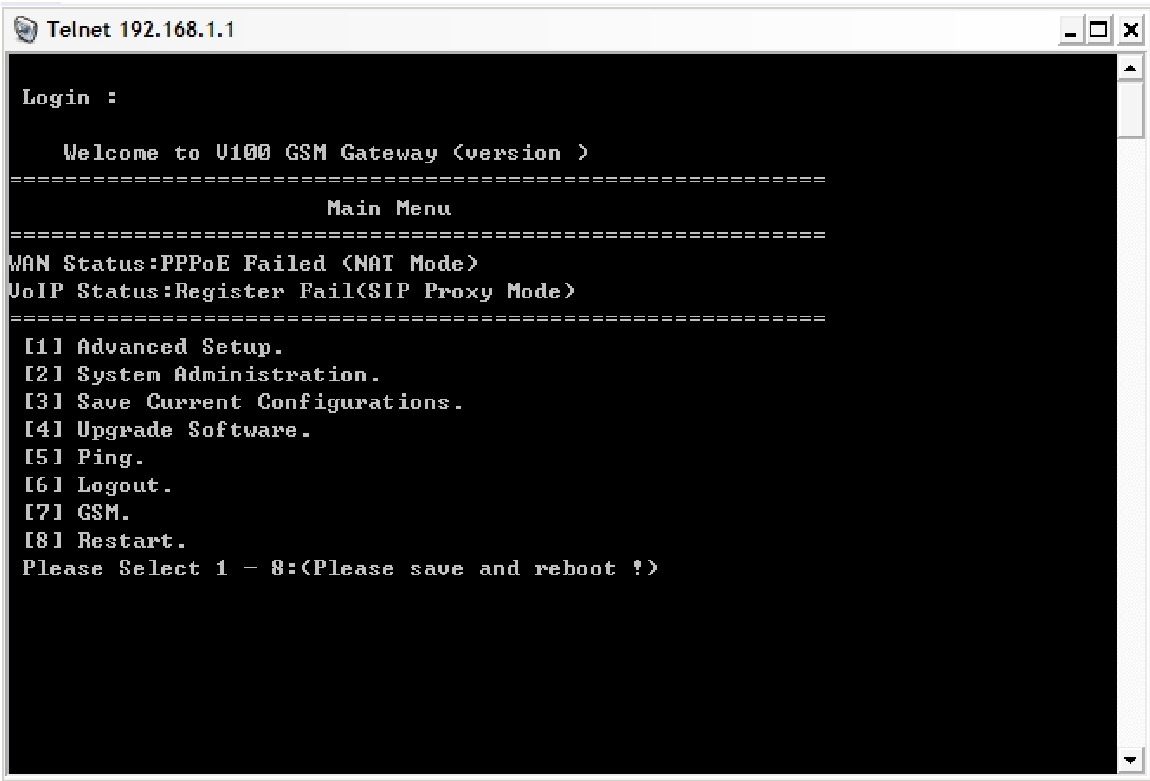
2. Upgrading Process

Notebook Telnet VoIP GW

[Open DOS mode]

C:> telnet [the IP of gateway]

Please select [4] Upgrade Software



Please input IP address of FTP server like as : 61.218.109.83
  Username : share
  Passswd : 19730809
  Imagename: s400.271

Upgrade (y/n) : y , then will write the firmware to flash.
(In different module or firmware , maybe have different change)

```
Connected to 61.218.109.83 port 21
[3] from 218.168.180.216 port 60002
220 (vsFTPd 1.2.0)
[Command] USER share
331 Please specify the password.
[Command] PASS xxxxxx
230 Login successful.
receiving bvw.15
[Command] TYPE I
200 Switching to Binary mode.
[4] going to listen 218.168.180.216 port 60002
[Command] PORT 218,168,180,216,234,99
200 PORT command successful. Consider using PASV.
[4] listener 0.0.0.0 port 60003
[Command] RETR bvw.15
150 Opening BINARY mode data connection for bvw.15 (1173940 bytes).
[4] Socket closed.
[5] accept from 61.218.109.83 port 20
Starting the file transfer
..........................................................................
1173940 bytes received in 39915 ms, (29.41Kbytes/sec), transfer succeeded
[5] Socket closed.
226 File send OK.
[3] Socket closed.
Upgrade(y/n) : y
```

After writing flash, Please reboot the Gateway.
If the new firmware (image) was most different with the previous version, please **push** the **hardware reset bottom to set to default.**
If the VoIP Gateway is in remote site, please use WEB configuration to **set to default.**

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Tel: 420-533 338805
Fax: 420-533 338883

Appendix

A FAQ List

1. What is the default administrator password to login to the gateway?
A: By default, your default username is “admin”, default password is “admin” to login to the router. For security, you should modify the password to protect your gateway against hacker attacks.

2. 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. Default username / password is admin / admin.

3. What is the default IP address?
A: The default WAN IP address is 192.168.1.1 with subnet mask 255.255.255.0.

4. What is different [set to default] and [Factory set to default]?
A: Factory set to default, you must push RST button until 5 second, gateway will clear all your setting, and let gateway Wan port become the factory default (192.168.1.1). 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.

5. Why can I call out when the gateway under the NAT?
A: VoIP product almost have 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.

6. Why does the one way talk happen?
A: Generally, one way talk happen when use the different codec between VoIP device make call. Please check and setting the same codec, most one way talk will be solved.

7. 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, then you can make a VoIP call.

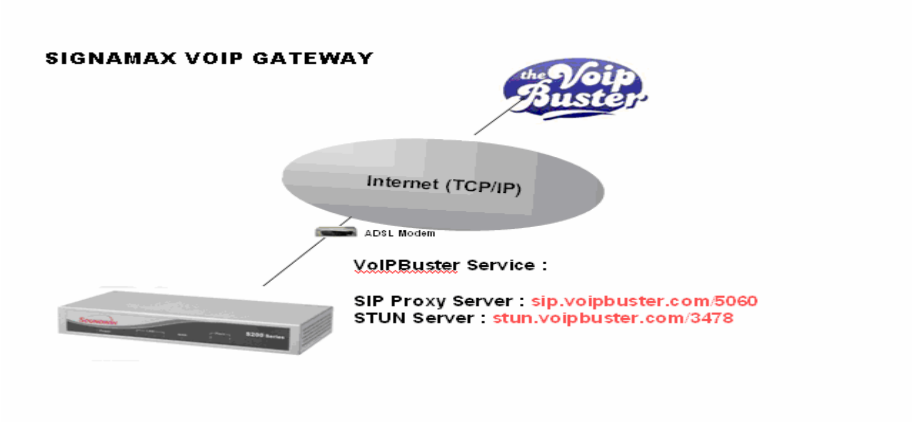
8. Why I use asterisk by G.729 sometimes disconnect happen?
A: In asterisk setting VAD must disable, if you open Silence Compression (VAD), it will make call disconnect happen, please disable the option when you use the asterisk.

9 Why can i register and use after setting?
A: After setting, please save configuration and reboot, after reboot you can use new

configuration.

B SIP Setting VoIP Buster

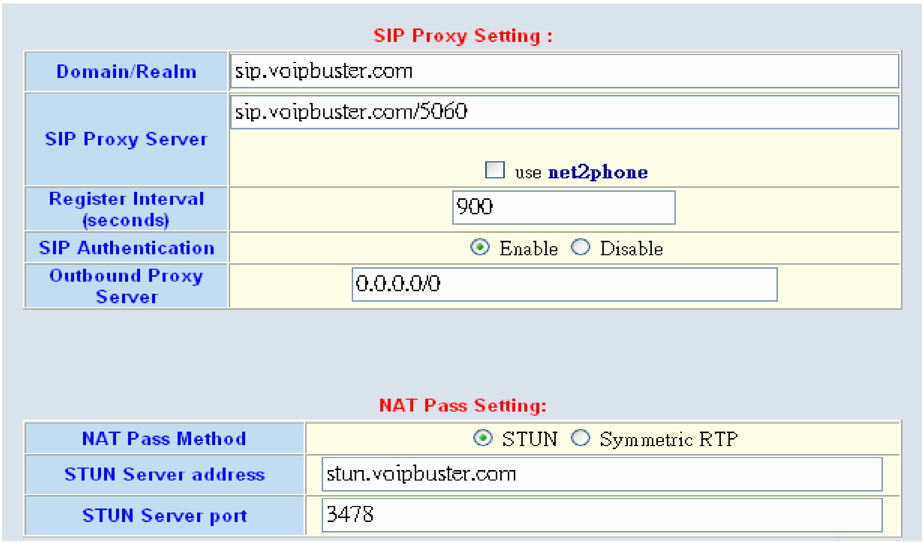
VoIPBuster Service Using VoIP Gateway



The Signamax GSM Gateway VoIP Gateway can register to VoIPBuster (http://www.voipbuster.com) VoIP service by SIP protocol and also can call SIP calls by VoIPbuster (http://www.voipbuster.com) service.

- **Gateway Setting**

1. **VoIPBuster SIP Proxy Server** : sip.voipbuster.com / 5060
2. **VoIPBuster STUN Server:** stun.voipbuster.com / 5060
3. VoIP Basic -> Setting SIP accounts and Set the Proxy Server and STUN server.



SIP Proxy Setting :

Domain/Realm	sip.voipbuster.com
SIP Proxy Server	sip.voipbuster.com/5060 ☐ 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	stun.voipbuster.com
STUN Server port	3478

- **How to dial the call?**

00 – country code – area code

For example Signamax company phone number is +420-53333805, the dial number is 0042053333805

- **VoIPBuster Provides Free Land Line (Fixed Line) Calls**

C Sip Speeds call

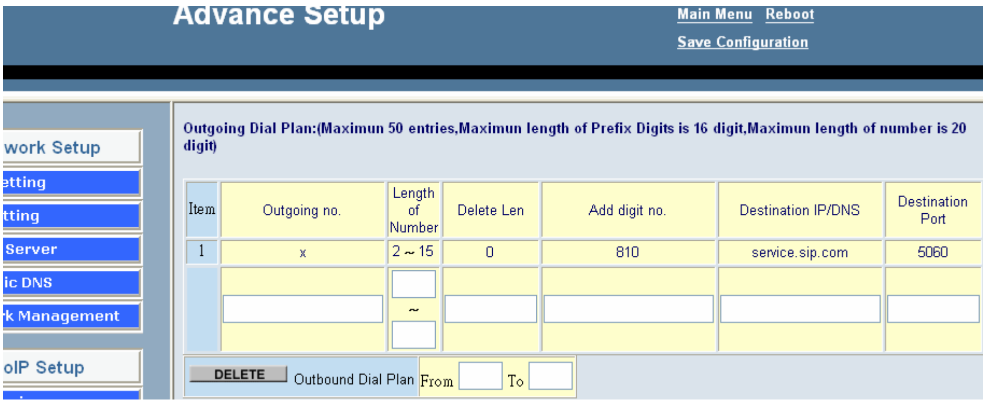
Speed Call Concept:
Cut your phone number down to fewer digit dialing!
Life is moving fast - you've got to dial fast. Now you can with Speed Dial. Dial the people you call most with just dialing fewer digits instead of dialing the full phone number.

- **SIP Register Mode**

Example: Gateway registers to sip proxy server: service.sip.com

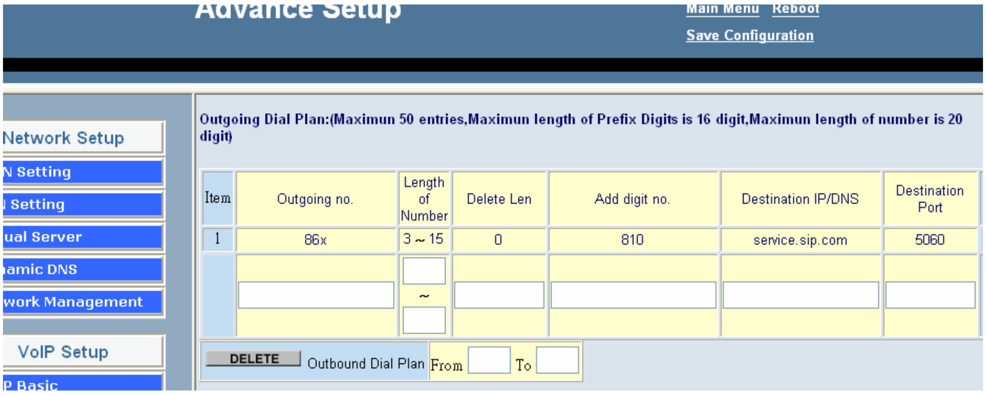
What's even better is that you can customize and manage your speed dial phone numbers in Dial Plan Setting on your gateway! Dial Plan allows you to set up to speed dial numbers that can be called with the fewer numbers.

Example 1: you want to dial any number instead of 810-any number



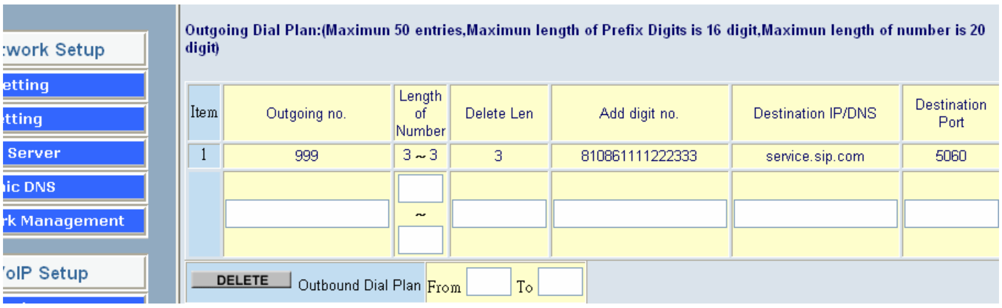
The destination IP address is the domain name of sip proxy server

Example 2: you want to dial 86-1111222333 instead of 810-86-1111222333

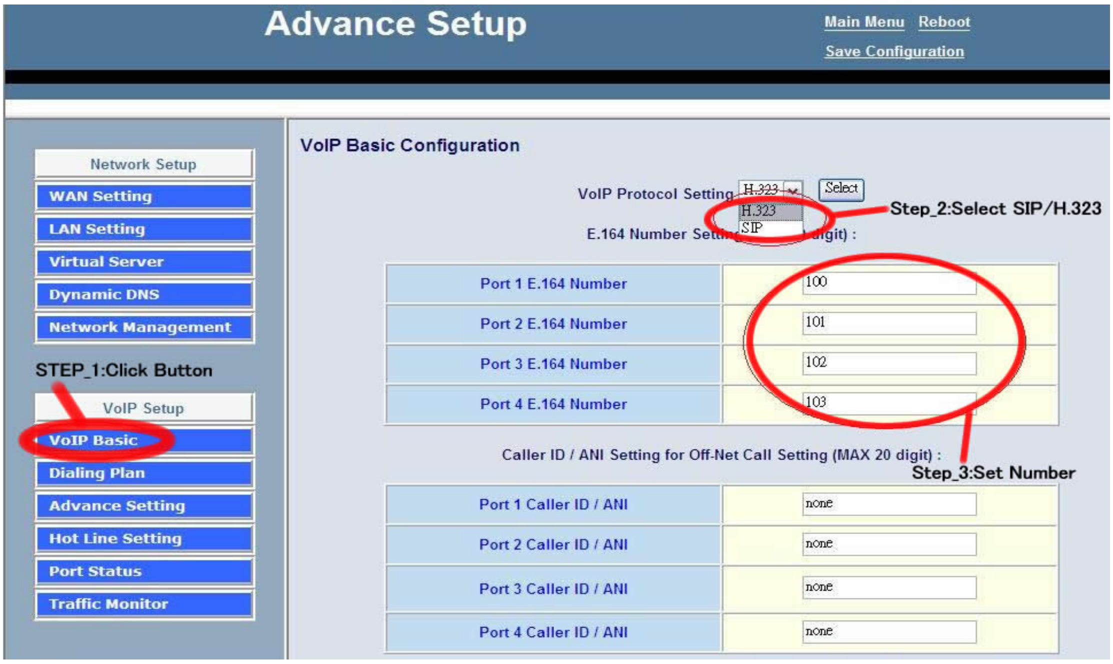


The destination IP address is the domain name of sip proxy server

Example 3: you want to dial 999 instead of 810-86111222333



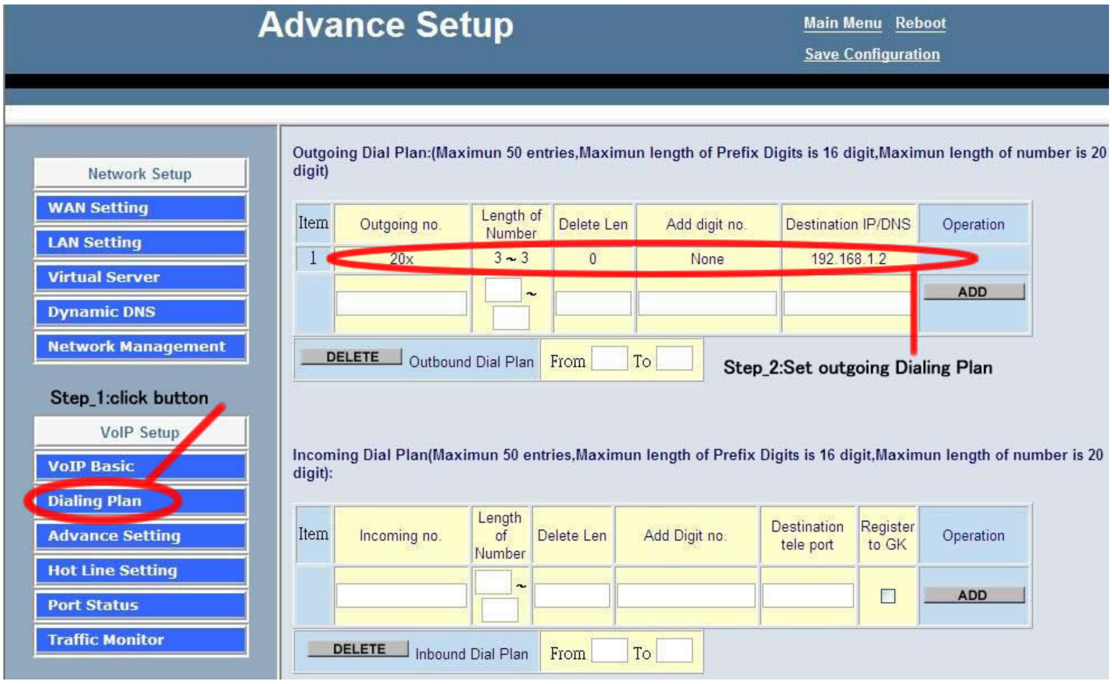
The destination IP address is the domain name of sip proxy serve



1. Choose "VoIP Basic". Login in web interface, and in "Advance Setting".

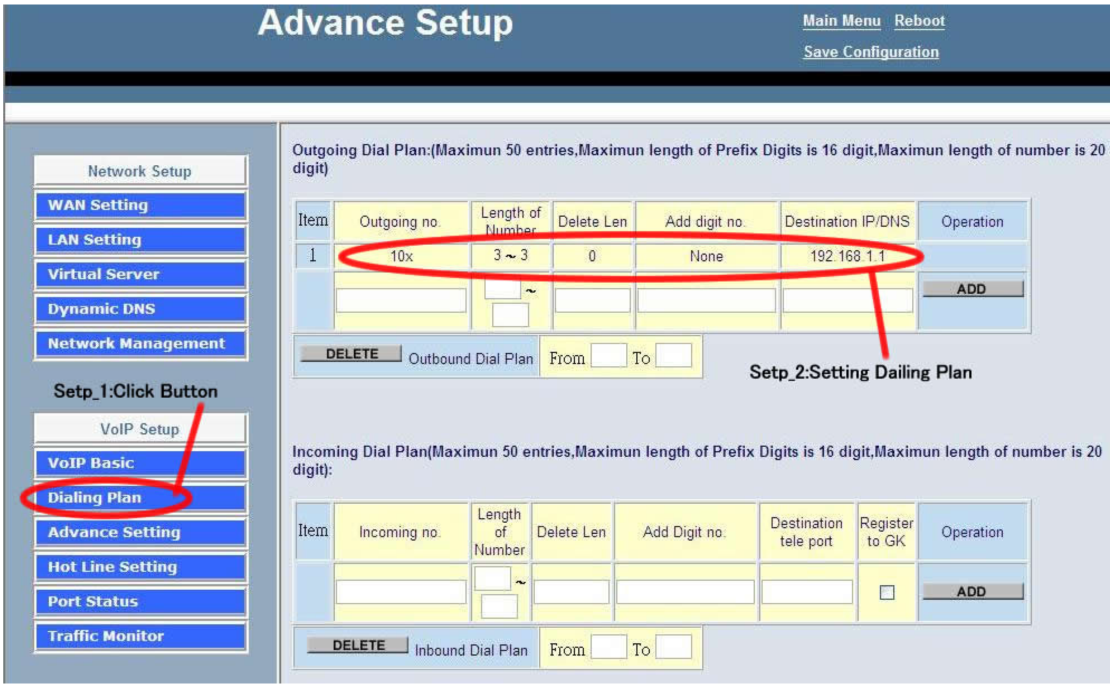
2. Select you wan to use protocol (SIP/H.323).
3. Input you want to use call number.

Setp_2:Setting Dialing plan



For Gateway_1 Setting

1. Choose "Dialing plan" and Setting Outgoing Dial plan.
2. Setting dial plan just like picture for demo."20x" the "x" mean wild card , it can be one of "0~9" number. And length "3~3", when you input 3 number and the call will be made. Destination is the Gateway_2 IP address.



For Gateway_2 Setting

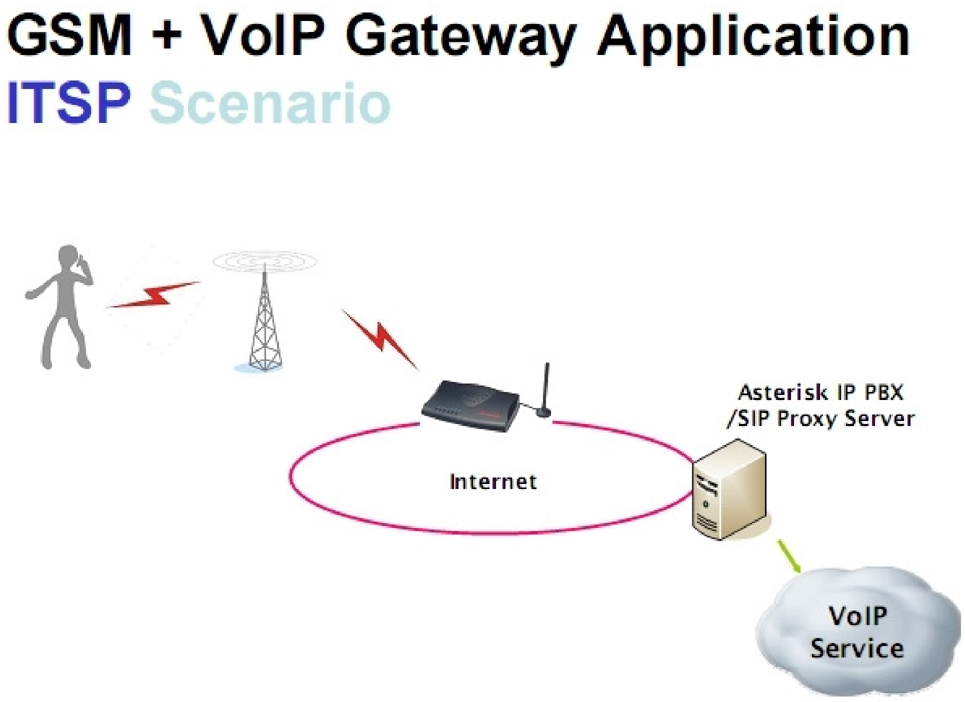
1. choose “Dialing plan” and Setting Outgoing Dial plan.
2. Setting dial plan just like picture for demo.”10x” the “x” mean wild card , it can be one of “0~9” number. And length “3~3”, when you input 3 number and the call will be made. Destination is the Gateway_1 IP address.

Step_3:make call each other

1.When you setting 2 gateway done, you can make call by each other. On gateway_1, just call “200”,and the gateway_2 Port_1 will ringing, then be made a call. And gateway_2 call “100” , the gateway_1 will ringing, then be made a ca

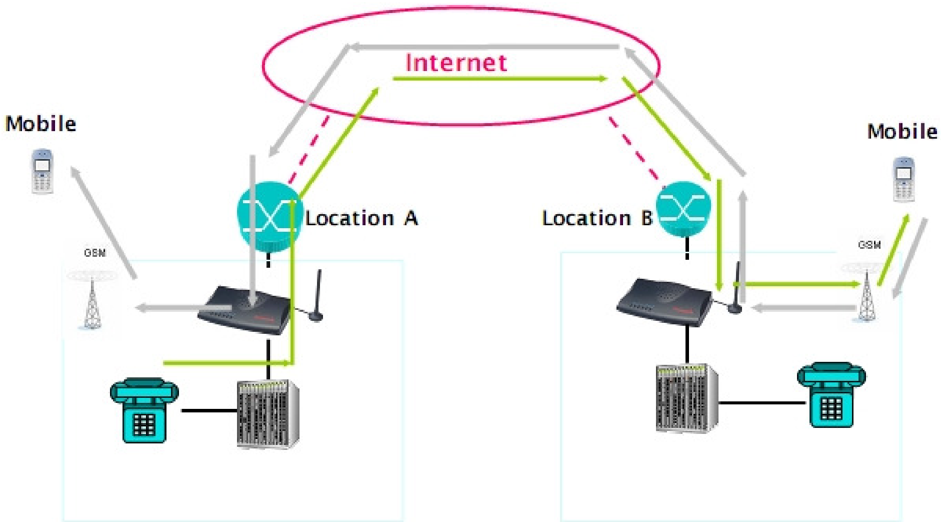
D Application

065-9066



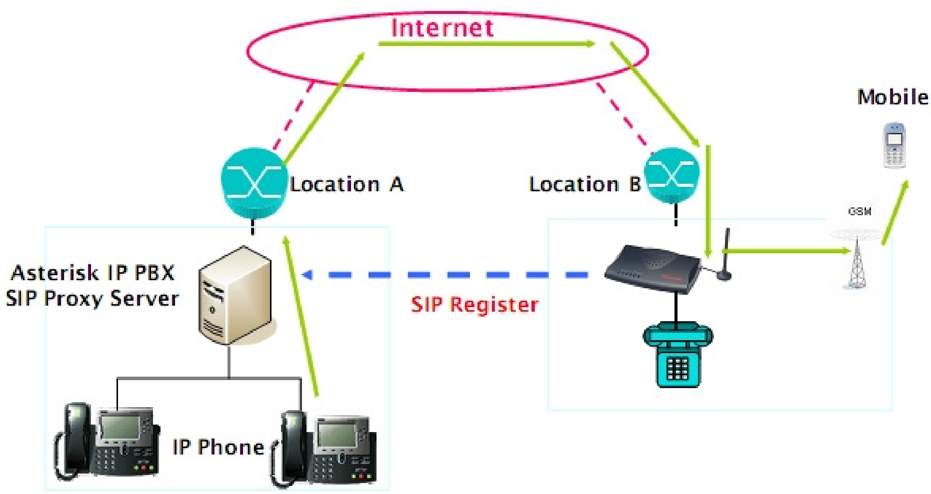
GSM + VoIP Gateway Application
Enterprise Scenario

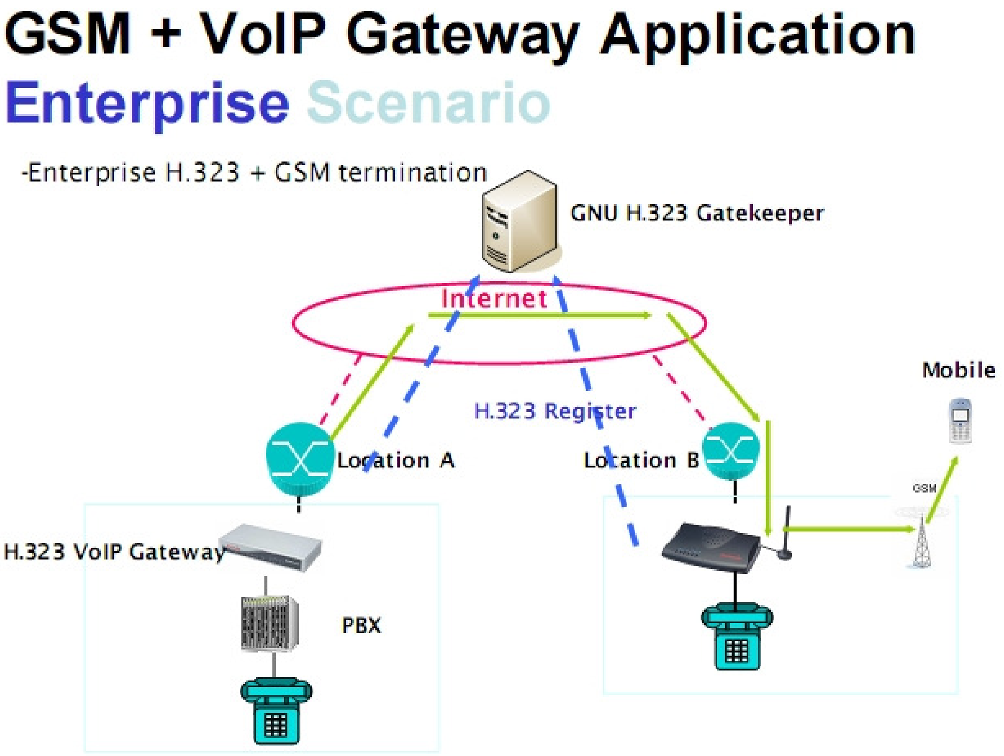
- Enterprise Peer-to-Peer GSM termination



GSM + VoIP Gateway Application
Enterprise Scenario

- Enterprise SIP + GSM termination




GSM + VoIP Gateway Application
Enterprise Scenario
-Enterprise H.323 + GSM termination
GNU H.323 Gatekeeper
Internet
Mobile
H.323 Register
Location A
Location B
GSM
H.323 VoIP Gateway
PBX