SOUNDWIN

User's Manual

VOIP VOICE GATEWAY

– Version: 2.33 –

Table of Contents

PREFACES

0.1 About This Manual

This manual is designed to assist users in using VoIP Gateway and Call Manager. 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 trade marks or copyrights of their respective companies, and are used only for identification or explanation 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 S series gateways 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

L / S series VoIP Gateway and Call Manager are the low to high VoIP total Solutions. This document describes the usage of Voice gateway and Call Manager.

1.1 Overview

- VoIP Gateway which is a device that allows one to connect a normal PSTN telephone to the Internet in order to make or place telephone calls.
- VoIP Gateway device may work in conjunction with a computer, such as an IP-sharing / Router, or it may be a stand-alone device that communicates with a service provider over the Internet.
- VoIP Gateway provides a direct analog interface for computer modems, fax machines, analog telephones, and other devices that require an analog port.
- 2/4/8 port Series VoIP Gateway can build in a simple H.323 Gatekeeper or SIP Proxy Server.
- VoIP Gateway also support standard Internet services, such as IP-Sharing, NAT, Virtual Server, DDNS, QOS, Port Filter, IP Filter function.

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	FXO	Foreign Exchange Office
FXS	Foreign Exchange Station	GMT	Greenwich Mean Time
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 Introduction

This VoIP Gateway provides a total solution for integrating voice-data network and PSTN.
The L200 and S series gateway is low to high density port gateway which support SIP / H.323 VoIP Protocol. Low model (2/4 port) can embedded H.323 Gatekeeper or SIP Proxy Server (Option). **The L200 and S series** gateway allows 2 ~ 24 lines (model option) analog voice and fax communication over a traditional data communications/data networking digital Internet. There are 6 model compare table follow.

Model Compare Table

Model	FXO Port	FXS Port	LAN Port	WAN Port	LCD Display	RS-232 port	SIP	H.323	H.323 Gatekeeper / SIP proxy Server (Software embedded)
S2400 Series (24 analog lines)									
S2400	0	24	1	1	√	√	√	√	
S2412	12	12	1	1	√	√	√	√	
S2424	24	0	1	1	√	√	√	√	
S1600 Series (16 analog lines)									
S1600	0	16	1	1	√	√	√	√	
S1608	8	8	1	1	√	√	√	√	
S1616	16	0	1	1	√	√	√	√	
SB800 Series (8 analog lines)									
SB 800	0	8	1	1	√	√	√	√	
SB 804	4	4	1	1	√	√	√	√	
SB 808	8	0	1	1	√	√	√	√	
S800 Series(8 analog lines Gateways /Embedded H.323 Gatekeepers/Embedded Sip Proxy Serves)									
S800	0	8	1	1		√	√	√	SK800/SVR800
S802	2	6	1	1		√	√	√	SK802/SVR802
S804	4	4	1	1		√	√	√	SK804/SVR804
S808	8	0	1	1		√	√	√	SK808/SVR808
S400 Series (4 analog lines Gateways/embedded Gatekeepers/Sip Proxy Servers)									
S400	0	4	4	1			√	√	SK400/SVR400
S401	1	3	4	1			√	√	SK401/SVR401
S402	2	2	4	1			√	√	SK402/SVR402
S404	4	0	4	1			√	√	SK404/SVR404

S200 Series (2 analog lines gateways/ Embedded H.323 Gatekeepers/ Embedded Sip Proxy Servers)									
S200	0	2	4	1			√	√	*SK200/SVR200*
S201	1	1	4	1			√	√	*SK201/SVR201*
S202	2	0	4	1			√	√	*SK202/SVR202*
L200 Series (2 analog lines Gateways/embedded Gatekeepers/Sip Proxy Servers)									
L200	0	2	1	1			√	√	*LK200/LVR200*
L201	1	1	1	1			√	√	*LK201/LVR201*
L202(*)	2	0	1	1			√	√	*LK202/LVR202(*)*
Call Manager									
C400	4	0	4	1			√	√	

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

1.4 Front Panel LED Indicators & Rear Panels

1.4.1 Gateway & Embedded Sip proxy server & Embedded H.323 Gatekeeper Outlook

S200/S400 Series & C400 Call Manager:

S200/S400 VoIP Gateway
SK200 /SK400 VoIP Gateway building in H.323 Gatekeeper Software
SVR200/SVR400 VoIP Gateway building in SIP Proxy Server Software
C400 Call Manager

S800 Series & C800 Call Manager:

-S800 VoIP Gateway
-SK800 VoIP Gateway building H.323 Gatekeeper Software
-SVR800 VoIP Gateway building SIP Proxy Server Software
-C800 Call Manager

S1600 / S2400 Series:

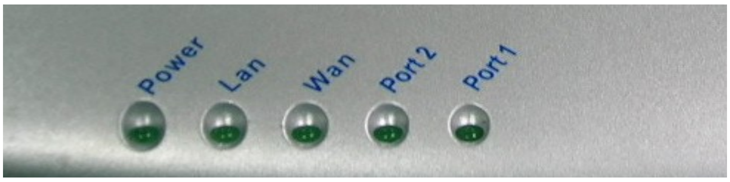
SSB800/S1600/S400 VoIP Gateway

L200 Series:

-L200 Series Gateway
-LK200 Series Gateway building in H.323 Gatekeeper Software
-LVR200 Series Gateway building in SIP Proxy Server Software

1.4.2 Front Panel LED and Container Descriptions

L200(GW/GK/SVR) Series



LED	State	Description
1. POWER	On	ATA is power ON
	Off	ATA is power Off
2. WAN port	On	ATA network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
3. LAN port	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
4. FXS	Off	Telephone Set is On-Hook
	Flashing	Ring Indication
	On	Telephone Set is Off-Hook
5. FXO	Off	Line is On-hook
	On	Line is In-Use

S200/S400(GW/GK/SVR) Series & C400 Call Manager

LED	State	Description
1. POWER	On	GW is power ON
	Off	GW is power Off
2. WAN port	On	GW network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
3. LAN port	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
4. FXS(Port)	Off	Telephone Set is On-Hook
	Flashing	Ring Indication
	On	Telephone Set is Off-Hook
5. FXO(Port)	Off	Line is not enabled
	On	Line is busy

NOTE: System initialization will turn some LEDs ON for a few sec.
When System Boot/Reboot , the Port LEDs will flash in turn for a few sec.

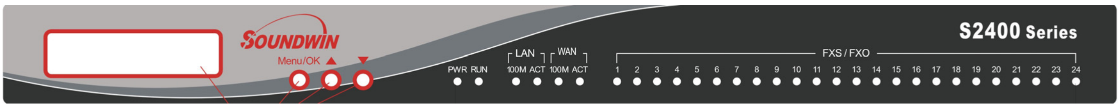
S800 (GW/GK/SVR) Series & C800 Call Manager

LED	State	Description
1. POWER	On	GW is power ON
	Off	GW is power Off
2. RUN port	On	GW connection established
	Flashing	Data traffic on cable network
	Off	Waiting for GW connection
3. WAN port	**100M**	
	On	GW network connection 100MB network
	Off	GW network connection 10MB network
	ACT	
	ON	GW network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
4. LAN port	**100M**	
	On	GW LAN connection 100MB network
	Off	GW LAN connection 10MB network

	ACT	
	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
5. FXS(Port)	Off	Telephone Set is On-Hook
	Flashing	Ring Indication
	On	Telephone Set is Off-Hook
6. FXO(Port)	Off	Line is not enabled
	On	Line is busy
7. RES Button	Push	Push Button until 5 second
		Set to Factory Default
8. RS-232		Console port connect to PC

NOTE: System initialization will turn some LEDs ON for a few sec.
When System Boot/Reboot , the Port LEDs will flash in turn for a few sec.

SB800/S1600/S2400 Series Gateway



LED	State	Description
1. POWER	On	GW is power ON
	Off	GW is power Off
2. RUN port	On	GW connection established
	Flashing	Data traffic on cable network
	Off	Waiting for GW connection
3. WAN port	**100M**	
	On	GW network connection 100MB network
	Off	GW network connection 10MB network
	ACT	
	ON	GW network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
4. LAN port	**100M**	
	On	GW LAN connection 100MB network
	Off	GW LAN connection 10MB network
	ACT	
	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
5. FXS(Port)	Off	Telephone Set is On-Hook

	Flashing	Ring Indication
	On	Telephone Set is Off-Hook

----

6. FXO(Port)	Off	Line is not enabled
	On	Line is busy

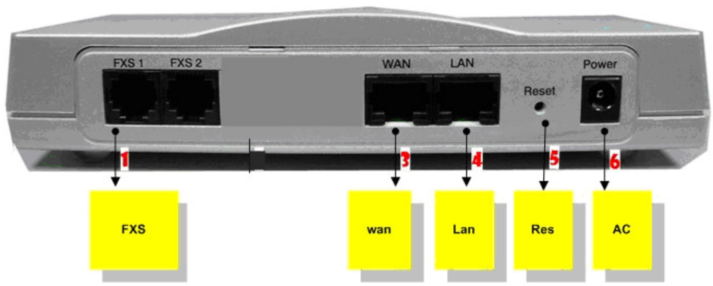
----

6. LCD Panel	Off	System is Shutdown
	On	System is Up

NOTE: System initialization will turn some LEDs ON for a few sec.
When System Boot/Reboot , the Port LEDs will flash in turn for a few sec.

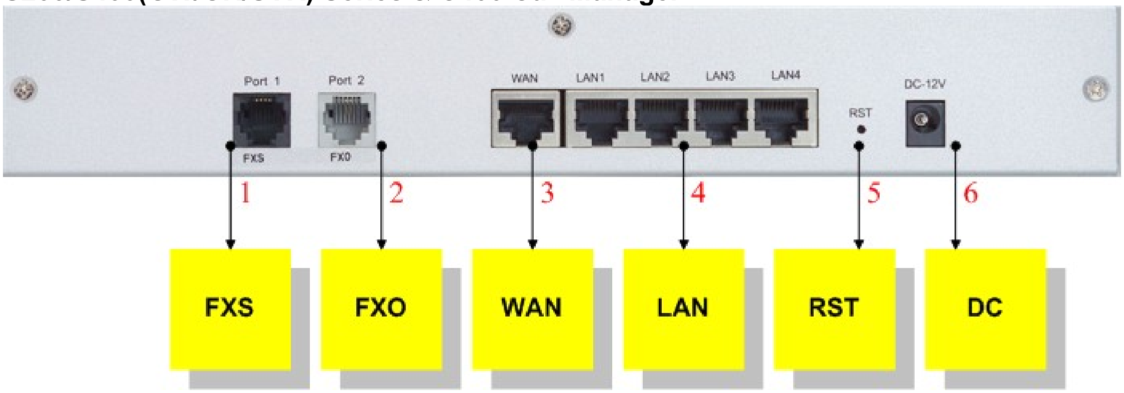
1.4.3 Rear Panel Descriptions

L200(GW/GK/SVR) Series



Item	Port	Description
1	FXS(Foreign Exchange Station)	FXS port can be connected to analog telephone sets or Trunk Line of PBX.
2	FXO(Foreign Exchange Office)	Can be Connected to PBX or CO line with RJ-11 analog line. FXO port can be connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
3	WAN(Wide Area Network)	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.
4	LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allows your PC or Switch/Hub to be connected to the ATA through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
5	RES(Reset button)	Push this button until 3 seconds, and ATA will be set to factory default configuration.
6	AC power(DC in 12V)	A power supply cable is inserted

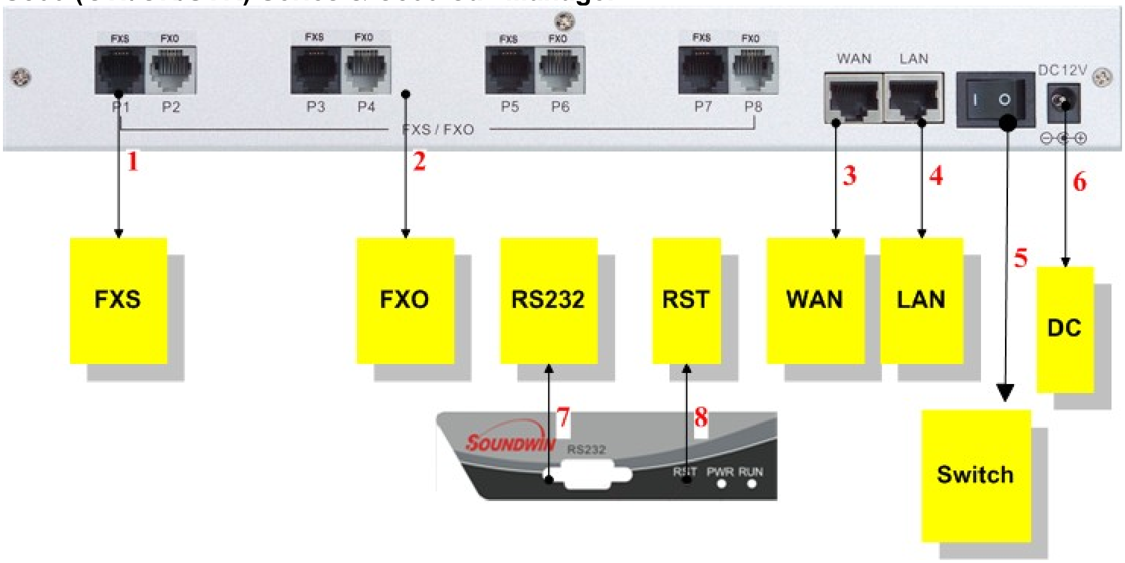
S200/S400(GW/GK/SVR) Series & C400 Call Manager



Item	Port	Description
1	FXS(Foreign Exchange Station)	Connect to Phone with RJ-11 (Black) analog line. FXS port was connected to your telephone sets, FAX, or Trunk Line of PBX.
2	FXO(Foreign Exchange Office)	Connect to PBX or CO line with RJ-11(Write) analog line. FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
3	WAN(Wide Area Network)	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.
4	LAN(Local Area Network)	Connect to PC with Ethernet cable. 4 ports allow your PC or Switch/Hub to be connected to the GW through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
5	RES(Reset button)	The reset button, when pressed, resets the cable voice gateway without the need to unplug the power cord. Push this button until 5 seconds, and GW will be set to factory default.
6	AC power(DC in 12V)	A power supply cable is inserted. The supplied power adapter converts 110V or 220V AC to DC as required for this device.

*there is no FXO port in Call Manager

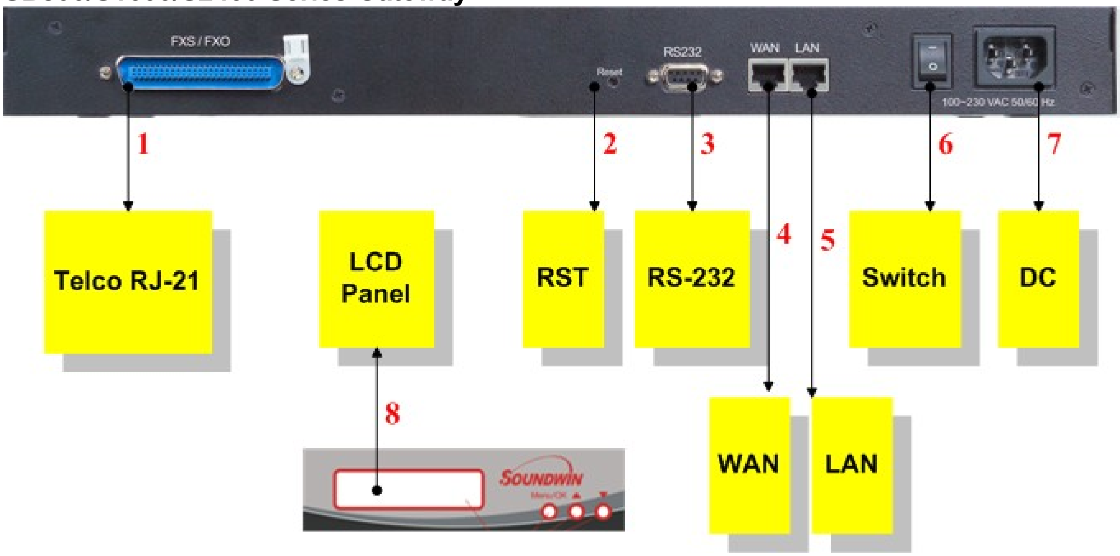
S800 (GW/GK/SVR) Series & C800 Call Manager



Item	Port	Description
1	FXS(Foreign Exchange Station)	Connect to Phone with RJ-11 (Black) analog line. FXS port was connected to your telephone sets, FAX, or Trunk Line of PBX.
2	FXO(Foreign Exchange Office)	Connect to PBX or CO line with RJ-11(Write) analog line. FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
3	WAN(Wide Area Network)	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.
4	LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allow your PC or Switch/Hub to be connected to the GW through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
5.	Switch power	Power Switch, turn on/off the GW power supplied. [I] is turn on the power, and [o] is turn off the power.
6	AC power(DC in 12V)	A power supply cable is inserted. The supplied power adapter converts 110V or 220V AC to DC as required for this device.
7	RS-232	RS-232 console port connect to PC, Use Pc com port to connect RS-232 console , setting GW configure.
8	RES(Reset button)	The reset button, when pressed, resets the cable voice gateway without the need to unplug the power cord. Push this button until 5 seconds, and GW will be set to factory default.

*there is no FXO port in Call Manager

SB800/S1600/S2400 Series Gateway



Item	Port	Description
1	Standard Telco 50 PIN Connector (RJ-21)	It is a 50 pins RJ-21 connector for connecting to telephone patch pane
2	RES(Reset button)	The reset button, when pressed, resets the cable voice gateway without the need to unplug the power cord. Push this button until 5 seconds, and GW will be set to factory default.
3	RS-232	RS-232 console port connect to PC, Use Pc com port to connect RS-232 console , setting GW configure.
4	WAN(Wide Area Network)	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.
5.	LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allow your PC or Switch/Hub to be connected to the GW through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
6	Switch power	Power Switch, turn on/off the GW power supplied. [I] is turn on the power, and [o] is turn off the power.
7	AC power(DC in 12V)	A power supply cable is inserted. The supplied power adapter converts 110V or 220V AC to DC as required for this device.
8	LCD Panel	Setting GW and view GW status, use [up/down] button to select Menu, and [menu] button return to main and click select.

1.5 Features and Specifications

The **L200 and S** Series Gateways provide many built-in server and software features to provide a convenient comprehensive solution for your VoIP network

1.5.1 Gateway Features

VoIP Key Features

- Both support SIP and H.323 protocols: SIP Registration and Digest Authentication and H.323 Gatekeeper Registration.
- Single Number / Account for multiple ports.
- Caller ID Delivery and Detection: FXS support DTMF&FSK Caller ID generation; FXO supports DTMF&FSK Caller ID detection. **(Optional)**
- Smart VoIP call Dialing Book: VoIP call Book could provide any application VoIP call to any type destination (Domain name / IP address, PSTN or PBX) or hunting number setting.
- AC termination Impedance : 600/900 OHM and complex impedance
- Answer Supervision for Polarity Reversal Detection and Voice detection
- NAT traversal: This feature allow gateway to operate behind any NAT/Firewall device. There is no need to change any configuration of NAT/Firewall like setting virtual server.
- Smart-QoS: This feature provides good voice quality when user place a VoIP call and access internet at the same time. The gateway will automatically start to reserve bandwidth for voice traffic when VoIP call proceeds.
- Call Hunting Facility: This function helps gateway to use the lines effectively. This facility automatically transfers your incoming call to a free line. Subscribers need not indicate numerous numbers of each port of gateway.
- Voice channels status display: This function display each port status like as on-hook, off-hook, calling number callee's number, talk duration, codec.
- Pulse Dial support: Support pulse dialing generation and detection. **(Optional)**
- Flash Detection and Generation Program: FXO support Flash Generation and FXS support Flash Detection.
- CDR: Use Syslog Server to receive CDR information that gateway send by UDP.
- Modules Card Extension: Extension FXO/FXO Modules card upgrade gateway port max to 24 port.**(SB800/S1600/S2400)**
- Embedded H.323 Gatekeeper (GK) / SIP Proxy Server (SVR): For 2/4/8 port gateway can embedded GK/SVR function. A simple H.323 GK / SIP proxy with gateway at the same device. Support standard Registered and call Sever Function.**(Option)**

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 Support(PLAR)

Configuration & Management

- Web-based Graphical User Interface
- RS232 for configuration(S800/S1600/S2400)
- Remote management over the IP Network
- FTP firmware upgrade
- Backup and Restore Configuration file
- Front LCD Panel for System Status and Management(SB800/S1600/S2400)
- Syslog support

1.5.2 Embedded H.323 Gatekeeper Features

- 200 H.3232 endpoints scale: SK400 providing 250 H.323 endpoints to register.
- Register Security Policy: SK400 providing security setting on your H.323 VoIP network. This provides protection for VoIP calls and insures proper endpoint identification.
- Pre-Granted Endpoints: letting other gateways or H.323 endpoints which were not register to this embedded gatekeeper. And the registered VoIP Gateways can make an Off-Net call to these pre-granted endpoints..
- Real time Call Detail Record and Post Call Detail Record Report : Support Real Time CDR to monitor VoIP calls, including caller ip, called ip , call date , call duration and other information. Also providing a CDR report to look up VoIP call record.
- Top 20 list: SK400 Gatekeeper can lists top 20 calls by call duration, caller number, calling number, caller IP or callee IP address.
- Syslog Client: Providing CDR information to Syslog Server.

1.5.3 Embedded Sip Proxy Features

- 200 SIP endpoints scale: SVR200 / SVR400 /SVR800 series providing 200 SIP endpoints to register.
- Trunk Line Setting for Off-Net Call: SVR400 / SVR 200 /SVR 800 series providing trunk interface for Off-Net call by ITSP. (Optional)
- Register Security Policy: SVR200 / SVR400 / SVR800 series providing MD5 authentication setting.
- Real time Call Detail Record and Post Call Detail Record Report : Support Real Time CDR to monitor VoIP calls, including caller ip, called ip , call date , call duration and other information. Also providing a CDR report to look up VoIP call record.
- Top 20 list: SVR200 / SVR 400 / SVR800 series can lists top 20 calls by call duration, caller number, calling number, caller IP or called IP address.
- Syslog client: Send CDR information to Syslog server.

1.5.4 Gateway & Gatekeeper & Sip proxy server Specifications

S200/S400/S800 Series Gateway & Gatekeeper & Sip proxy server & Call Manager

Telephony Specification:	
Voice Codec:	G.711(A-law / μ -law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps).
FAX support :	T.30 / T.38.
Echo Cancellation:	G.165/G168(Version:2000).
FXO Caller ID detection :	DTMF and FSK **(Optional).**
FXO hang up detection / anti-seized :	Tone Learning Automatically / Manual Tone Learning **(Optional).**
Answer supervision:	Support Battery Reverse Detection and Voice Detection.
FXO answer delay time:	Support delay 0 – 8000 ms to answer.
Adjustable AC Termination Impedance :	600 / 900 OHM and complex Impedance.
Failsafe Mechanism (FXS relay to FXO) :	Power failed by pass support / Internet Failed by pass (**Optional**)).
Creative Metering:	12K Hz and 16K Hz Metering **(Customized)**
IP Specification:	
Protocol:	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.
LAN :	Support Virtual Server, DHCP Server.
WAN:	Support PPPoE client, DHCP client, Fix IP Address, DDNS client.
Network Address Translation:	Providing build-in NAT router function.
Smart QoS:	Guarantee the voice bandwidth
TOS:	IP TOS (IP Precedence) / DiffServ
General Specification	
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 :	IU-440 x 250 x 45 mm**(SB800/S1600/S2400)** 260 x 130 x 35 mm**(S800/GK/SVR)** 260 x 130 x 35 mm**(S200/S400/GK/SVR)**
Weight:	5200 g**(SB800/S1600/S2400)** 1500g (Aluminum)**(S800/GK/SVR)** 00g (Aluminum)**(S200/S400/GK/SVR)**
Others:	Standard 50 pin RJ-21 Telco connectors **(SB800/S1600/S2400)**

SB800/S1600/S2400 Series gateway can be extension. Combination different modules card, you can change/upgrade your gateway FXS/FXO ports. Max Up to 24 ports.

Modules Card Table (SB800/S1600/S2400 Only)

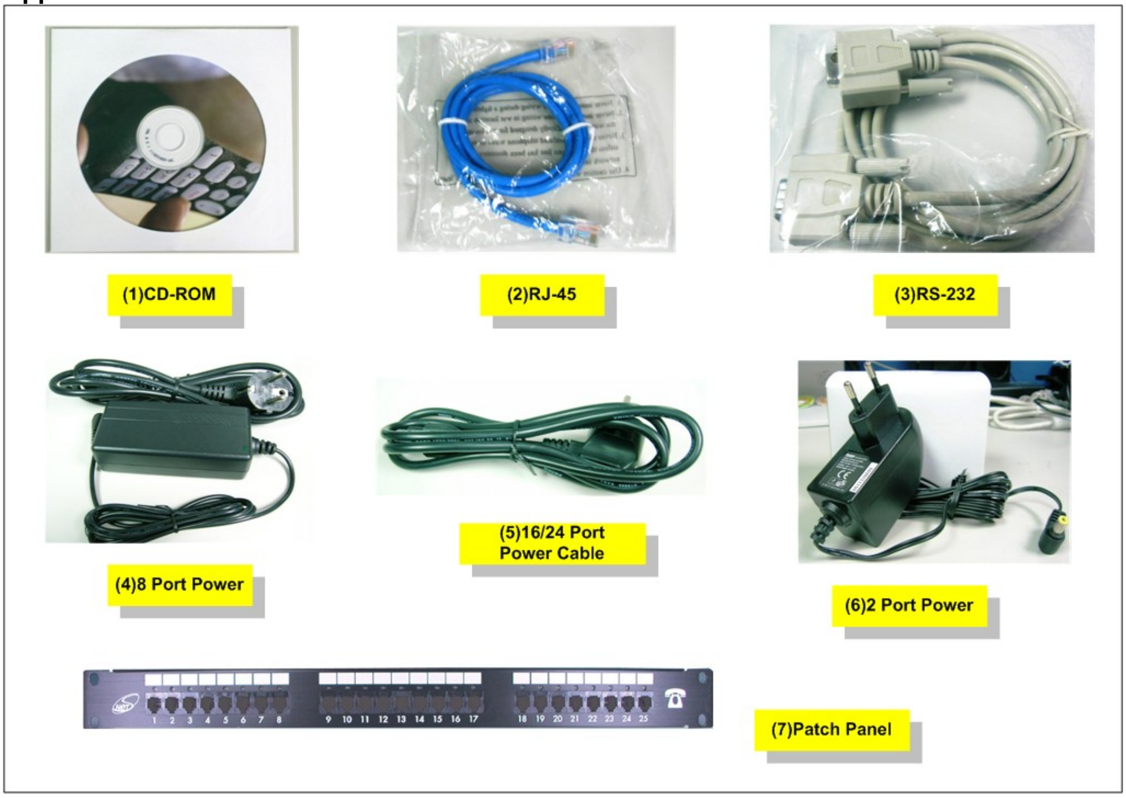
Mother Board	Description and Function
SB800	Mother Board with 8 FXS Interface
SB804	Mother Board with 4 FXS + 4 FXO Interface
SB808	Mother Board with 8 FXO Interface
Modules Card	**Description and Function**
SM800	8 FXS Interface Module
SM804	4 FXS + 4 FXO Interface Module
SM808	8 FXO Interface Module

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:



Item	Appurtenances	Description
1	CD-ROM	CD Include in all product user manual and datasheet.
2	RJ-45 cable	Internet cable RJ-45 connect to NIC/Gateway/Router
3	RS-232 cable	RS - 232 Console port connect to PC COM port.
4	Power supply & cable(8)	Power Supply,input:100-240V output:+12V (Europe/UK/US)
5	Power cable(16/24)	Power Supply cable.
6	Power supply & cable(2/4)	Power Supply,input:100-240V output:+12V (Europe/UK/US)
7	25 port Telephone Patch Panel and Cable (Optional).	For 16/24 port telephone interface Patch Panel .

2.1.1 L200 /S200/S400 Series Gateway & embedded Gatekeeper/ Sip Proxy Server

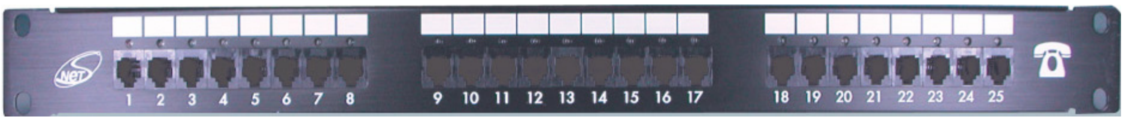
The 2/4 Port packet contents:	
GW,GK,SVR(S200/ S400 Series) RJ-45 AC Power Adapter CD-Rom(User manual)	X1 X1 X1 X1

2.1.2 S800 Series Gateway & embedded Gatekeeper/ Sip Proxy Server

The 8 Port packet contents:	
GW,GK,SVR(S800 Series) RJ-45 RS-232 AC Power Adapter CD-Rom(User manual)	X1 X1 X1 X1 X1

2.1.3 SB800/S1600/S2400 Series High Density Gateway

Patch Panel:



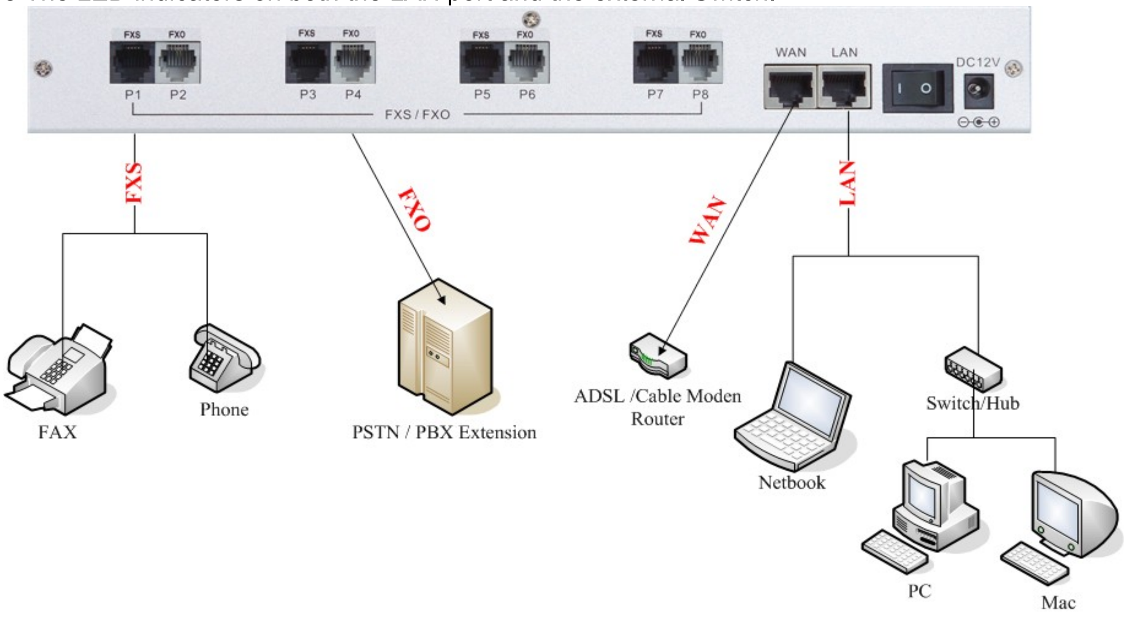
The 8/16/24 Port packet contents:	
Gateway (SB800/S1600/S2400 Series)	X1
RJ-45	X1
RS-232	X1
AC Power Cable	X1
CD-Rom(User manual)	X1
Patch Panel(Option)	

2.1.4 C400 Call Manager

Call Manager contents:	
Call Manager(C400/C800)	X1
RJ-45	X1
RS-232(only C800)	(X1)
AC Power Adapter	X1
CD-Rom(User manual)	X1
Modular Duplex Jack Extension Cord	X4 (C400) X8 (C800)

2.2 Installation

Install Gateway
1 Connect the 12V DC IN to the power outlet with power adaptor.
2 Connect FXO to PSTN / PBX Extension Line.
3 Connect FXS to a telephone jack with the RJ-11 analog cable (Phone / PBX Trunk Line.)
Connecting to a PC:
1 Connect the Ethernet cable (with RJ-45 connector) to any LAN port.
2 Connect the other end of the Ethernet cable to your PC’s installed network interface card (NIC).
3 The LED indicators at both the Ethernet port and the NIC should be ON.
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.
3 The LED indicators on both the LAN port and the external Switch.

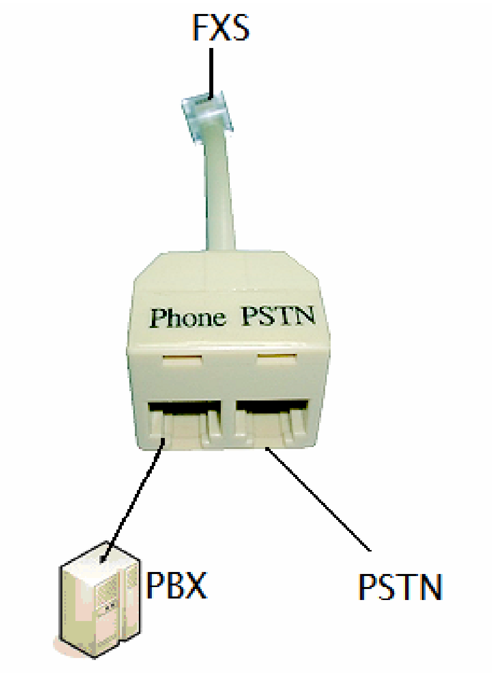


Port	Description
FXS(Foreign Exchange Station)	FXS port can be connected to analog telephone sets or Trunk Line of PBX.
FXO(Foreign Exchange Office)	Can be Connected to PBX or CO line with RJ-11 analog line. FXO port can be connected to the extension port of a PBX or directly connected to a PSTN line of carrier.
WAN(Wide Area Network)	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.
LAN(Local Area Network)	Connect to PC with Ethernet cable. 1 port allows your PC or Switch/Hub to be connected to the GW through a networking cable with RJ-45 connectors used on 10BaseT and 100BaseTX networks.
RES(Reset button)	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

Call Manager ONLY

Connect the cord to FXS port
Connect the PSTN line to PSTN port
Connect the PBX to phone port



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

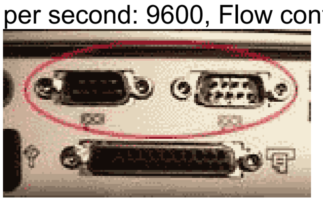
There are 3 way to setting gateway - **[Web User Interface] [Telnet] [Console]** (Some Series modules have RS-232 console port like S800/SB800/S1600/S2400).

2.3.1 Factory Default setting

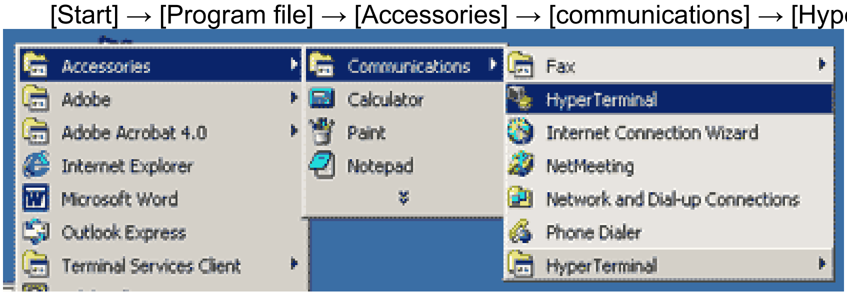
- **WAN** Port IP address : 192.168.1.1
- **LAN** Port IP address : 222.222.222.1
- LAN DHCP Server enable IP range: 222.222.222.51 ~ 222.222.222.100
- **VoIP Number(S200 Series) Port_1~Port_2 number:100,200**
- **VoIP Number(S400 Series) Port_1~Port_4 number:100,200,300,400**
- **VoIP Number(S800 Series) Port_1~Port_8 number:100,200,300,400,500,600,700,800**
- **VoIP Number(SB800 Series) Port_1 ~ Port 8 number : 101~ 108**
- **VoIP Number(S1600 Series) Port_1 ~ Port 16 number: 101~ 116**
- **VoIP Number(S2400 Series) Port_1 ~ Port 24 number: 101~ 124**
- VoIP default setting was **H.323** signal protocol, **Direct Mode**, **Fast-Start** and **G.723** codec.
- Default login authentication **username : admin, password : admin**

2.3.2 Console

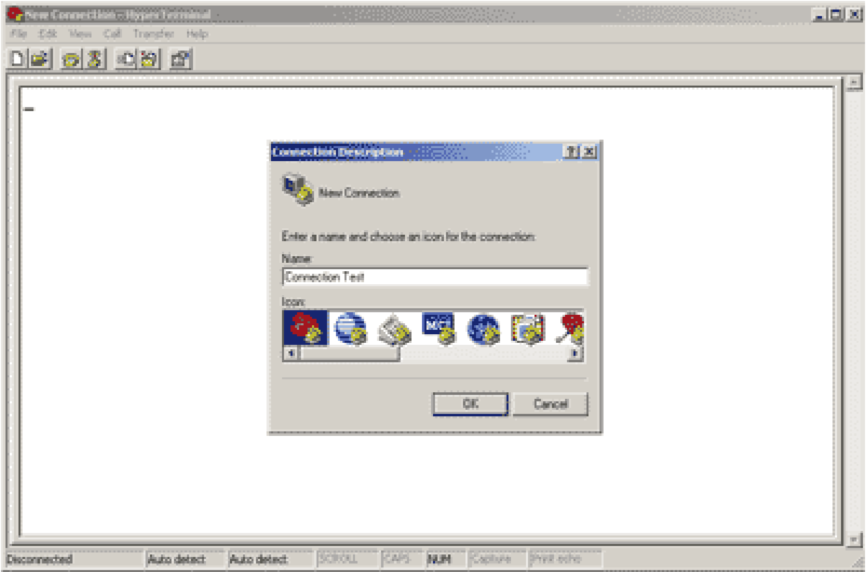
RS-232 port (DB-9pin **male** connector), Configure the COM Port Properties as following: Bits per second: 9600, Flow control: None



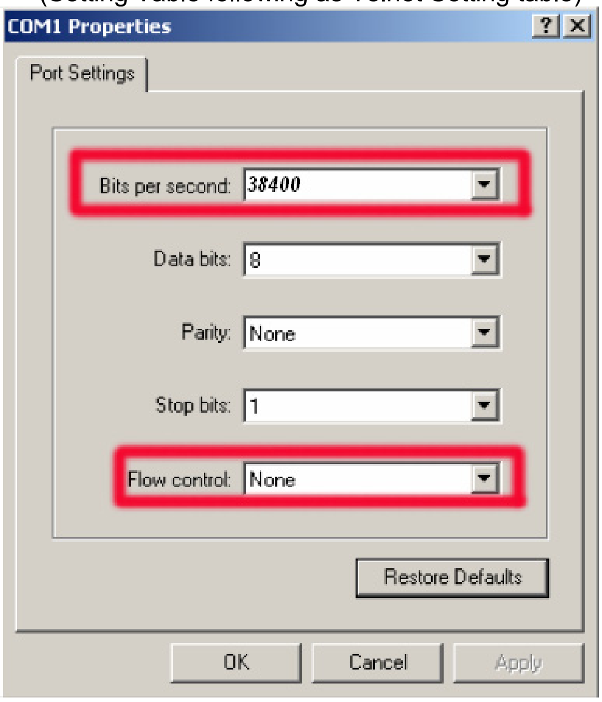
1. Connect Gateway RS-232 port to PC COM Port.
2. Power on gateway.
3. Open Terminal Program (ie. Windows XP Hyper Terminal)
   [Start] → [Program file] → [Accessories] → [communications] → [Hyper Terminal]



4. Create New connection. Select “Com” port that connect PC to gateway



5. Make connection(Bits Pre second:**38400** Flow contact: **None**)
6. Input “Enter” and Show Welcome display.
7. Login, input the Password to login.(Password as the same as Access, default is admin)
8. Setting Gateway Configure like telnet mode
   (Setting Table following as Telnet Setting table)

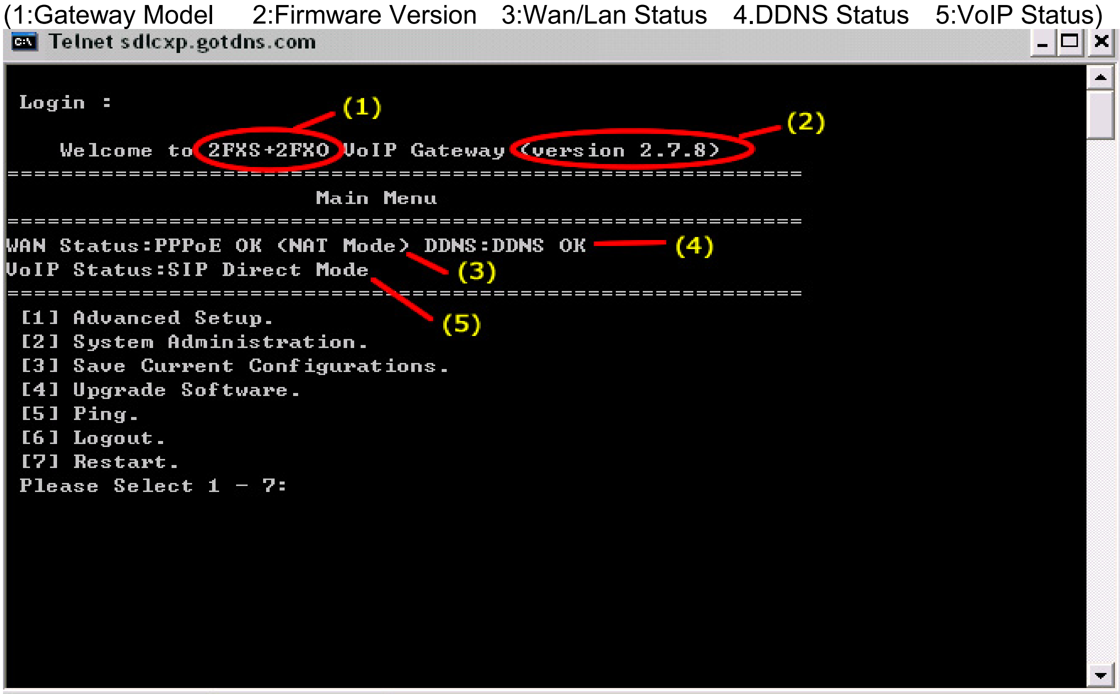


2.3.3 Telnet

Connect WAN/LAN 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/Lan)
2. Remote Gateway by Telnet. If telnet successful, you will see Login display.
(For Example: telnet 222.222.222.1)

3. Input Password (Gateway Access password, Default: admin), If login successful, you will enter the welcome display.
(1:Gateway Model 2:Firmware Version 3:Wan/Lan Status 4.DDNS Status 5:VoIP Status)



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] Restart.
[1]Advanced Setup	1.WAN Setting 2.LAN Setting 3.Virtual Server 4.Dynamic DNS 5.Network Management 6.VoIP Basic 7.Dialing Plan 8.VoIP Advance Setting 9.Hot Line Setting a.Port Status
[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.LAN Setting	1.Change to Bridge Mode 2.Change LAN IP Address 3.Disable DHCP Server 4.Change Start IP Address 5.Change End IP Address 6.Change DNS Server IP 7.Change Lease Time

[1]Advanced Setup ……3.Virtual Server	1.Add Virtual Server 2.Delete Virtual Server
[1]Advanced Setup ……4.Dynamic DNS	1.Change DDNS username 2.Change DDNS password 3.Change DDNS domain name 4.Change DNS server IP
[1]Advanced Setup ……5.Network Management	1.Change web server port 2.Change telnet server port
[1]Advanced Setup ……6.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 ……7.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 ……8.VoIP Advance Setting	**(1)Sip Advance** 1.Set DTMF Relay Mode 2.Change FAX Mode 3.Change RFC2833 Payload(96-127) **(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 7.FXO Parameters Setting 1.Change FXO Impedance 2.Change Line In Volume 3.Change Line Out Volume 4.Change FXO Tx Gain 5.Change FXO Rx Gain 6.Flash Duration 7 DTMF Tone Power 8.FXO Transmit Hybrid 9. Answer Supervision Setting a. Change Ringer Voltage Threshold b. Enable Line Silence Disconnect c. Change FXO Answer delay time d. FXO Ringer Voltage Filter Setting

	(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 ……9.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

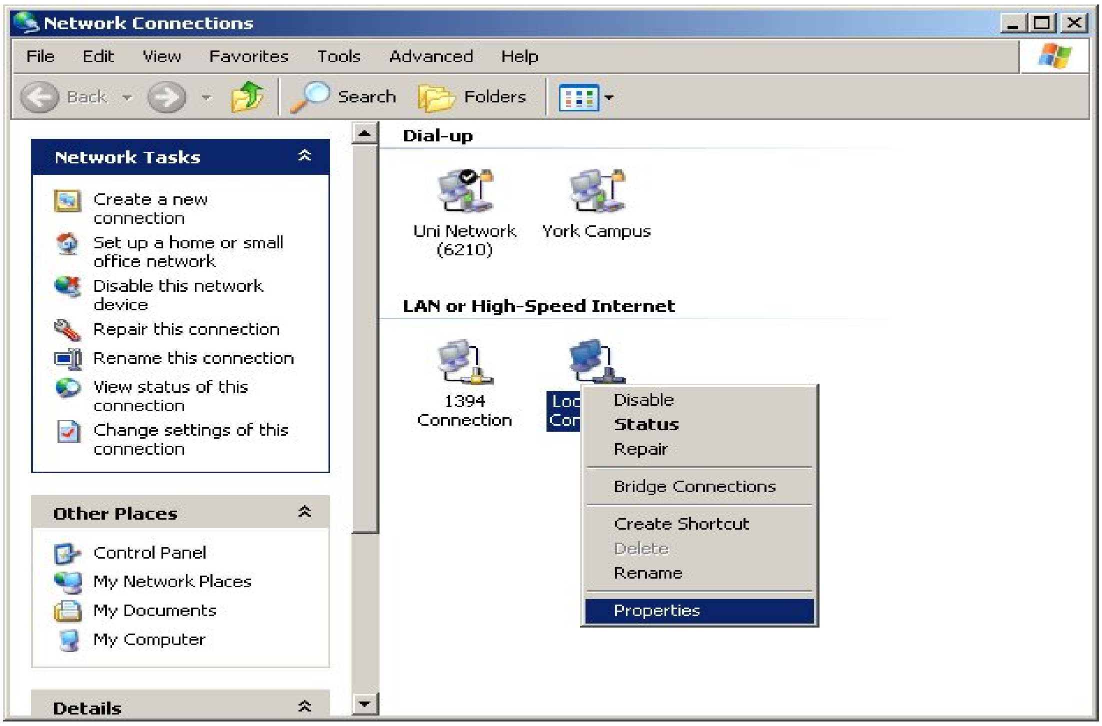
The gateway has a built-in HTTP(Web) server for configuration. Before you use the gateway to access the Internet, you should set up a management PC to log into the router for further configuration. The management PC may be configured with a fixed or dynamically assigned IP address. For a fixed IP address, use an IP address from a 192.168.1.0/24 network, such as 192.168.1.10.

For a dynamic IP address, you need to set the PC as a DHCP client, and then restart or renew the network settings. The DHCP server of router is enabled by default so the PC will then be assigned an IP address and related settings by the router. The following examples are for a MicrosoftTM Windows 2000/XP machine set to use a dynamic IP address.

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.

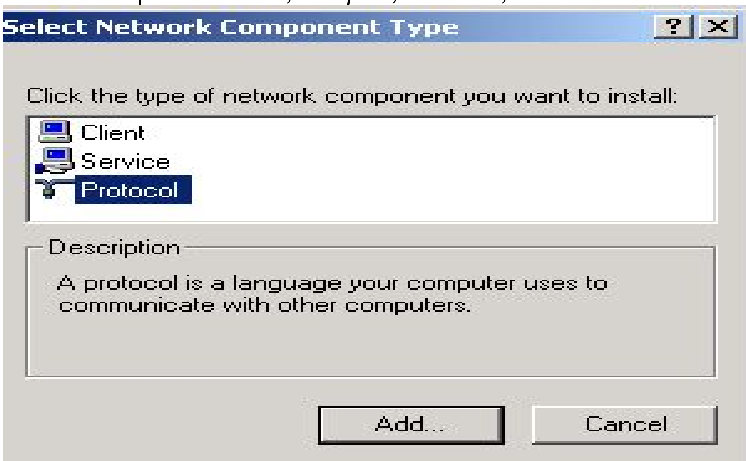
1. On the desktop, Please enter start -> control panel -> network setting.” Click Properties. The Network screen will open.



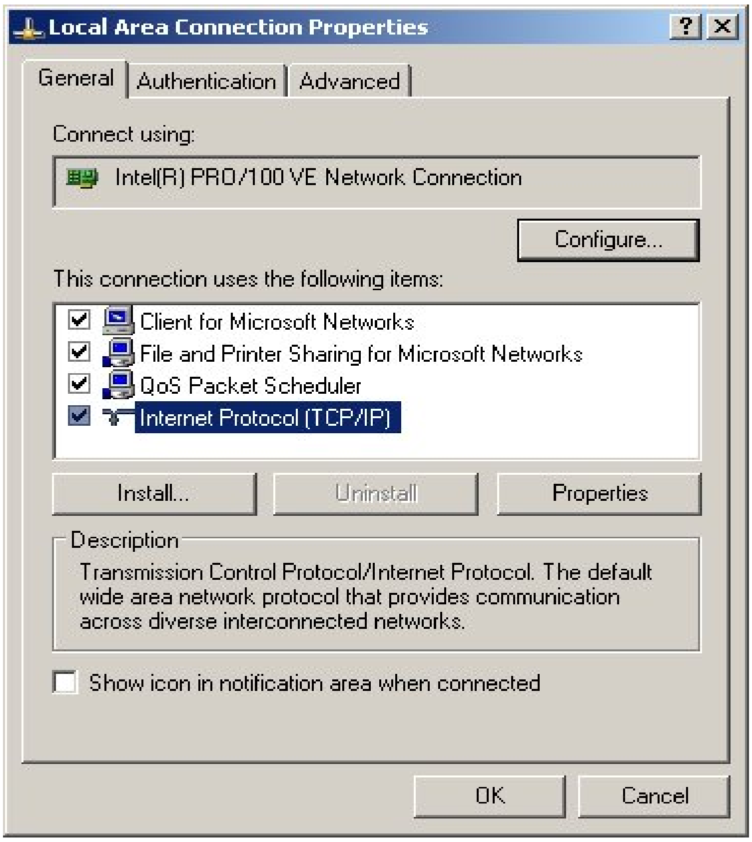
(Your particular system will be different from the screen shown here.)

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, click Obtain an IP address automatically. As the DHCP (Dynamic Host Configuration Protocol) server built into the router is enabled by default, your computer will get an IP address, subnet mask, and other related IP network settings from the router.

3. On the DNS Configuration tab, click Disable DNS”.

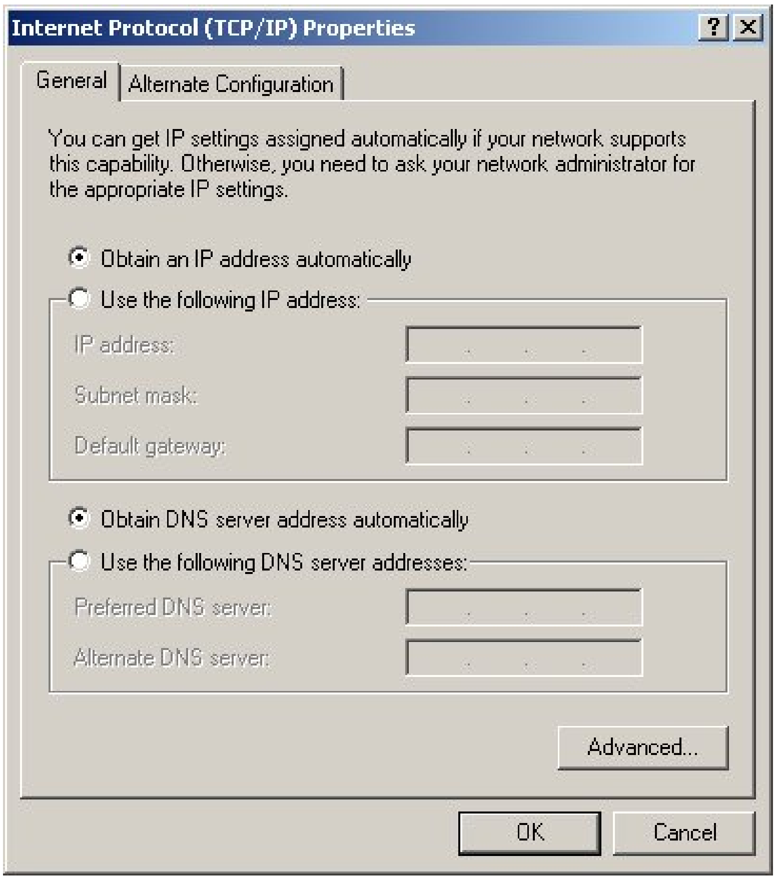
4. Click the Gateway tab.

5. Make the New gateway and Installed gateways fields blank and click OK. A dialog box will pop up asking you to restart the PC. Click Yes”.

Checking TCP/IP settings

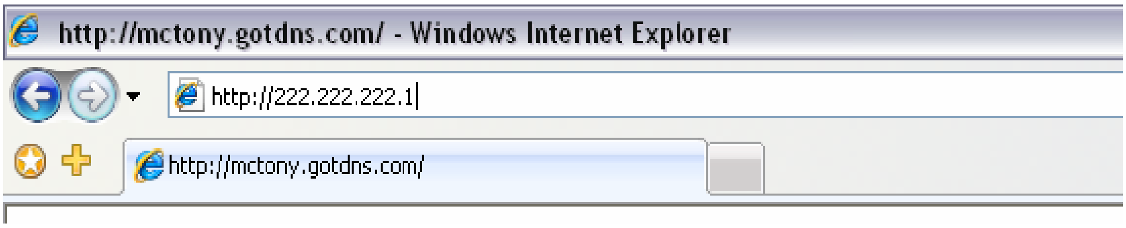
1. After completing the previous steps, click Start -> Run -> and type ipconfig /all. The IP Configuration window will open. If the PC does not show an IP address in the 222.222.222.51 to 222.222.222.100 range, click the ipconfig /release button to release the current configuration. Wait a few seconds 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 222.222.222.1” (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.



Connecting to the Web Configuration via a Web Browser

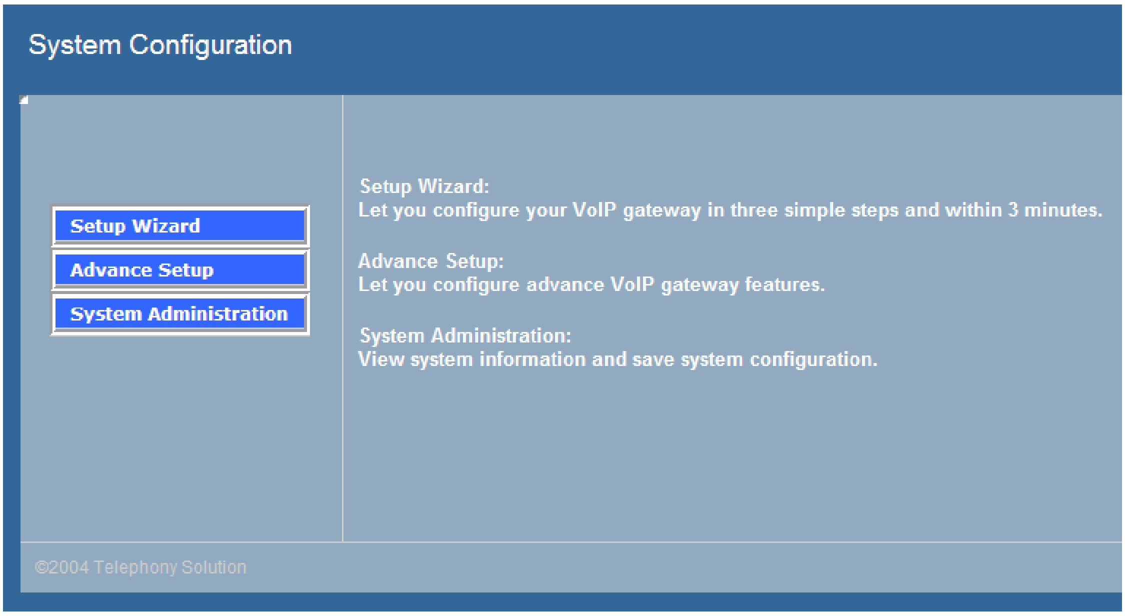
1. Launch the Web browser(IE or Firefox). Enter **http://222.222.222.1** into the browser **Address** window and press the Enter Key



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 [Setup Wizard], [Advanced Setup] and [System Information] were displayed.



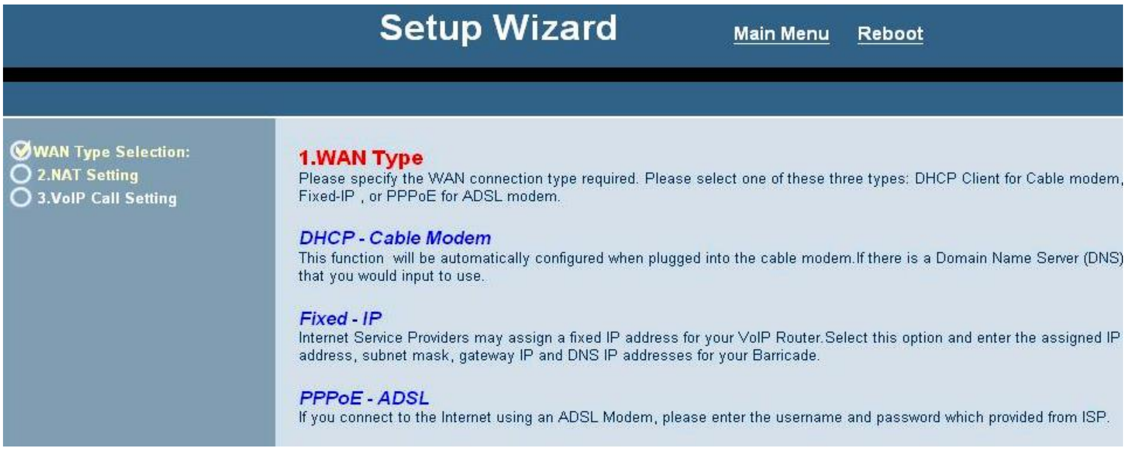
Wizard for Quick Setup

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

3.1 WAN Port Type Setup

For most users, Internet access is the primary application. The **S** Series 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 Port Type Setup**" from within the Wizard **Setup**, the following setup page will be show.

Three methods are available for Internet Access:

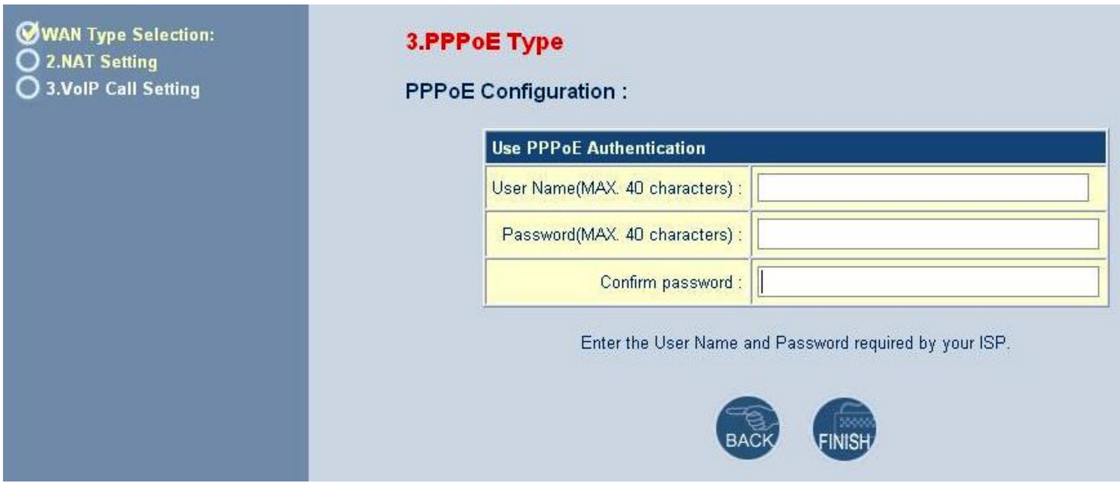


Fixed IP User: If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP.

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

ADSL Dial-Up User (PPPoE Enable)

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

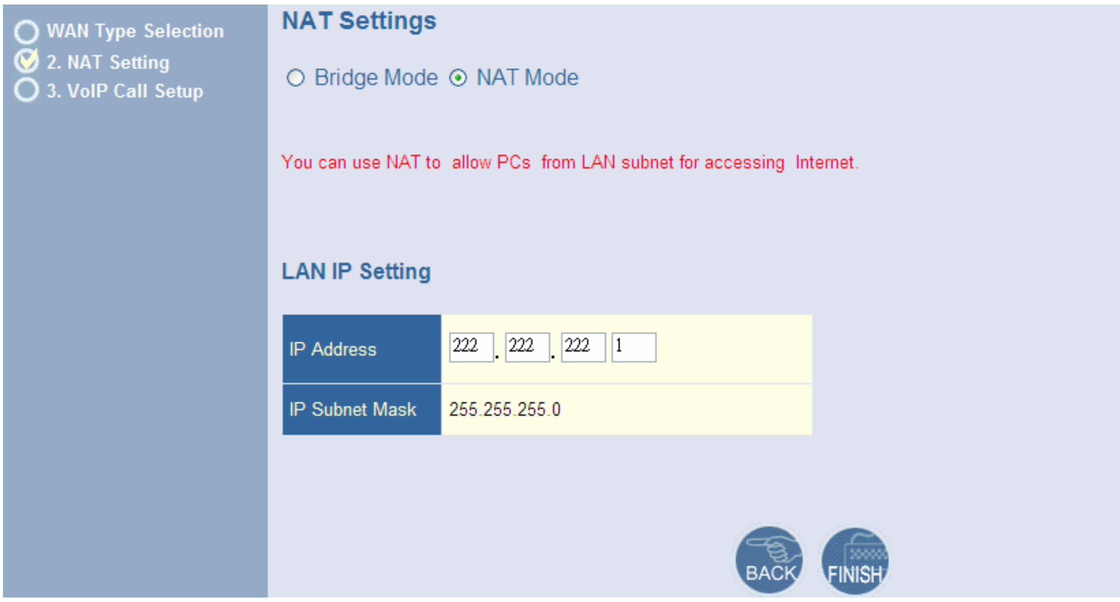


- **User Name:** Enter User Name provided by your ISP
- **Password:** Enter Password provided by your ISP.
- **Retype Password:** Enter Password to confirm again.

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.

3.2 Configuring NAT or Bridge setting:

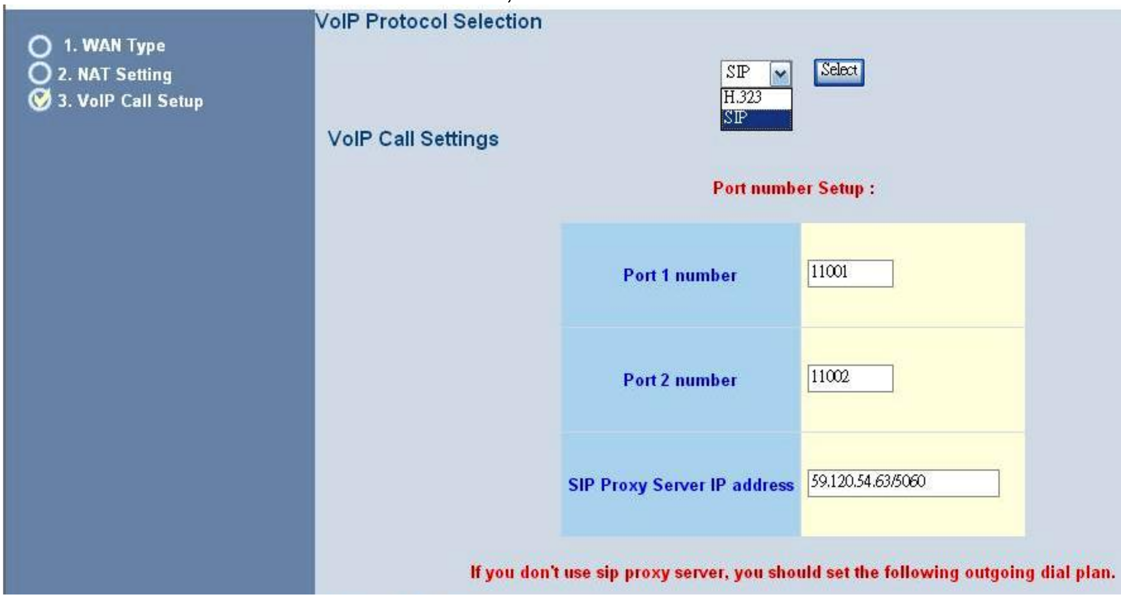


- **Bridge Mode:** Select S series Gateway as bridge.
- **NAT mode:** LAN IP Network Configuration
- **IP Address:** Private IP address for connecting to a local private network (Default: 222.222.222.51).
- **Subnet Mask:** Subnet mask for the local private network (Default: 255.255.255.0).

3.3 VoIP Call Protocol Setup

Step1 : Configure VoIP Call Signal Protocols :

- User could select H.323 or SIP Protocol, and click "select"



Step2 : configure the numbering with phone/line ports.

- **Phone Number (FXS):** The representation number is the phone number of the telephone that is connected to Phone port.

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

Step3: Let GW Register to Gatekeeper(GK)/SIP Proxy Server
(If user does not have GK/SIP Proxy Server, Please go to Step 4: Outgoing Dialing Plan)

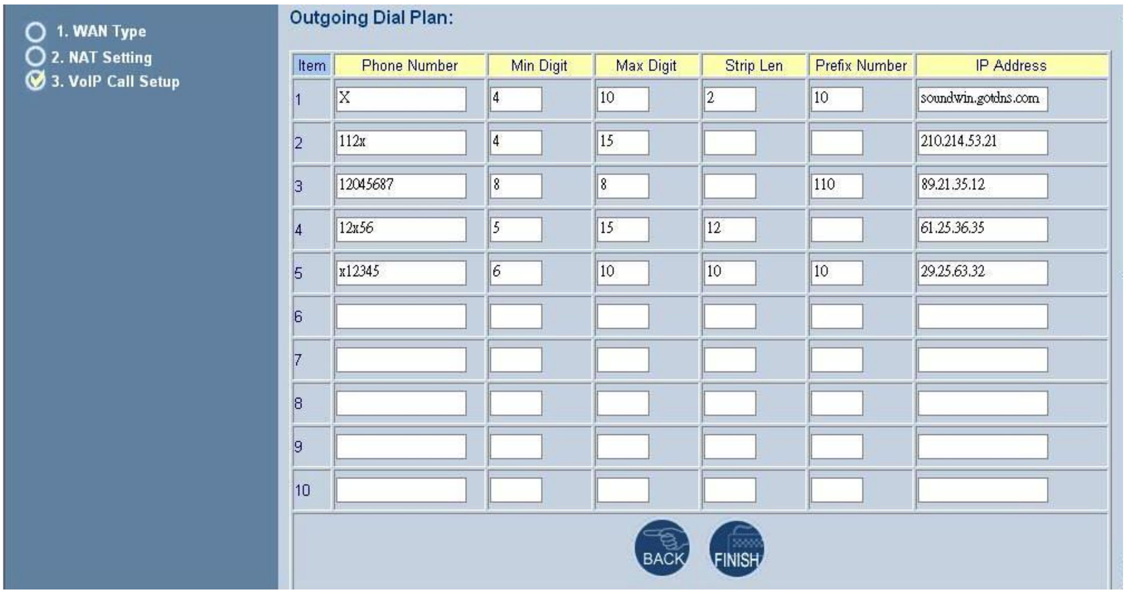
- **Gatekeeper IP address:** There is a gatekeeper address fields. If this gateway does not want to register to any gatekeeper, just set value 0.0.0.0 to the primary gatekeeper address.

- **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.

Step 4: Outgoing Dialing Plan

The purpose of "Outgoing Direct Call" setting is to let user create a proprietary dialing plan when this Gateway is not registered to any H.323 Gatekeeper or any SIP Proxy Server. This setting can also assign some dialing plan to local ports (including prefix strip, prefix addition).Through this setting, user can directly map a number to a specific gateway (IP address).

In the "Outgoing Dial Plan" settings.



Item	Phone Number	Min Digit	Max Digit	Strip Len	Prefix Number	IP Address
1	X	4	10	2	10	soundwin.gotdns.com
2	112x	4	15			210.214.53.21
3	12045687	8	8		110	89.21.35.12
4	12x56	5	15	12		61.25.36.35
5	x12345	6	10	10	10	29.25.63.32
6						
7						
8						
9						
10						

- “Leading Number” is the leading digits of the dialing number.
- “Min Length” and “Max Length” is the min/max allowed length you can dial.
- “Strip Length” is the number of digits that will be stripped from beginning of the dialed number.
- “Prefix Number” is the digits that will be added to the beginning of the dialed number.
- “Destination” is the IP address of the destination Gateway that owns this phone number.

Step 5: Finishing the Wizard Setup

After completing the Wizard Setup, please click “Finish” bottom. The VoIP Gateway will save the configuration and rebooting Gateway automatically. After 20 Seconds, you could re-login the Gateway.

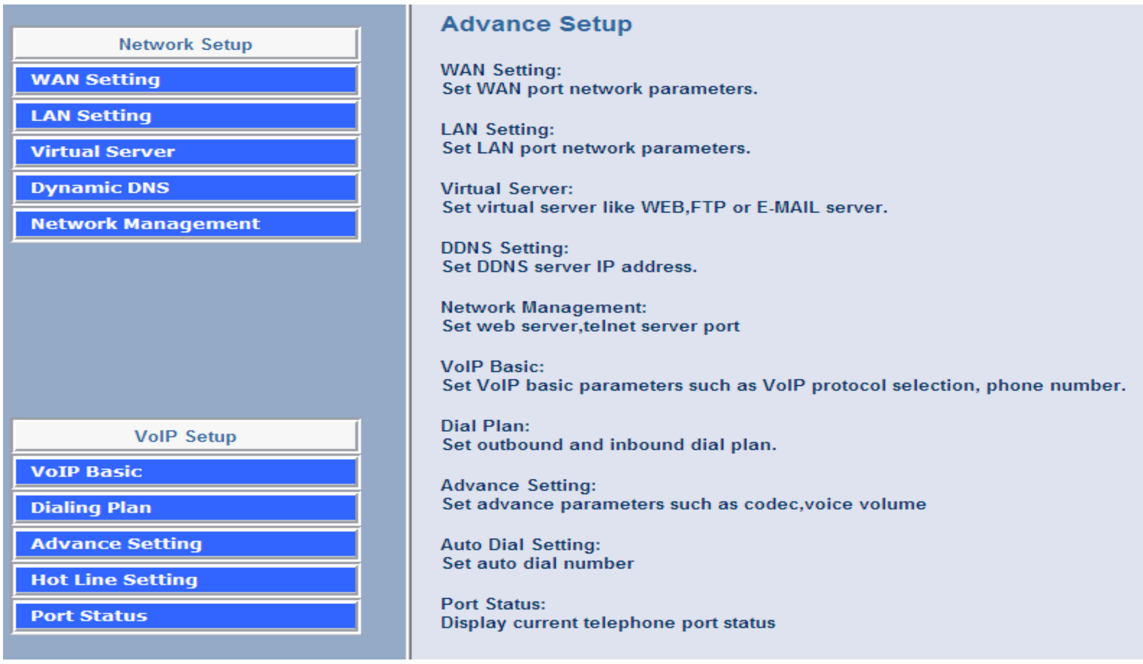
Gateway Setting

Gateway setting include in some of advance setting, SIP Proxy Server (SVR) setting, Gatekeeper (GK) setting. There are many detail explain follow. Setting in “advance Setting”.

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

- **SET Wizard**
- **Advance Setup**
- **System Administrator**

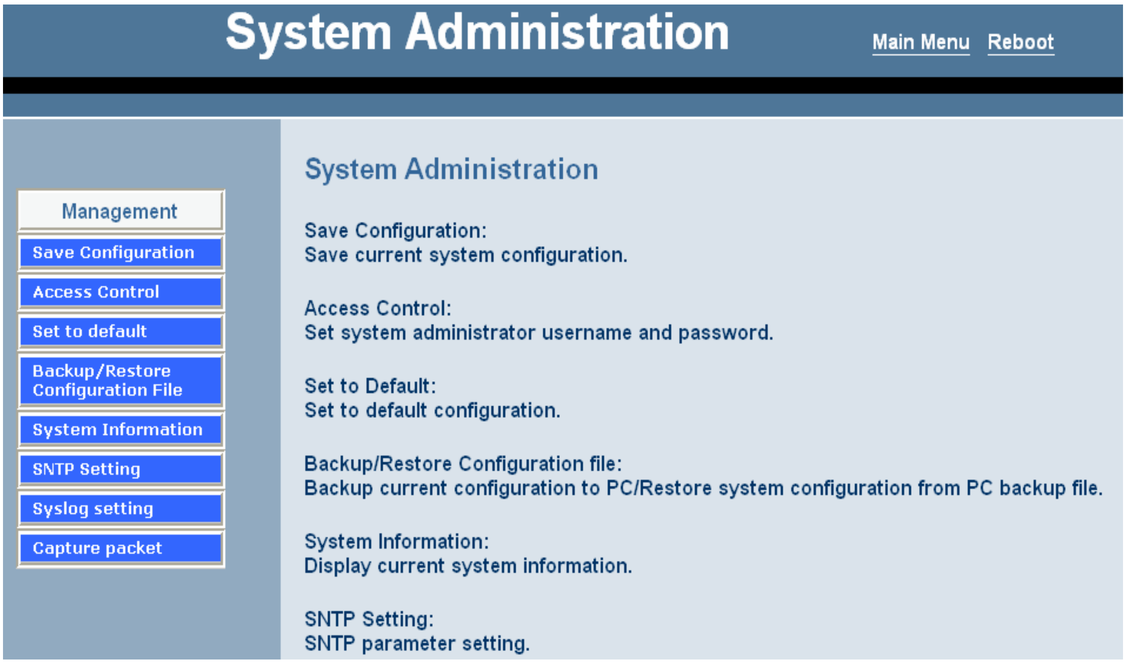
Advanced Configuration:



Advanced Setting Label

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

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.
Backup/Restore Configuration	User can backup the configuration file of Gateway to PC or Restore the configuration file from PC.
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 down load from http://www.ethereal.com/)

4.1 Network Configuration

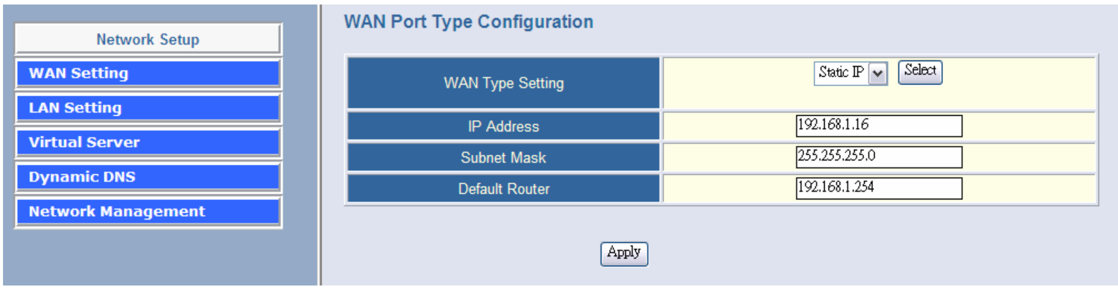
4.1.1 WAN Port Type Setup

For most users,Internet access is the primary application. The **S** Series 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:

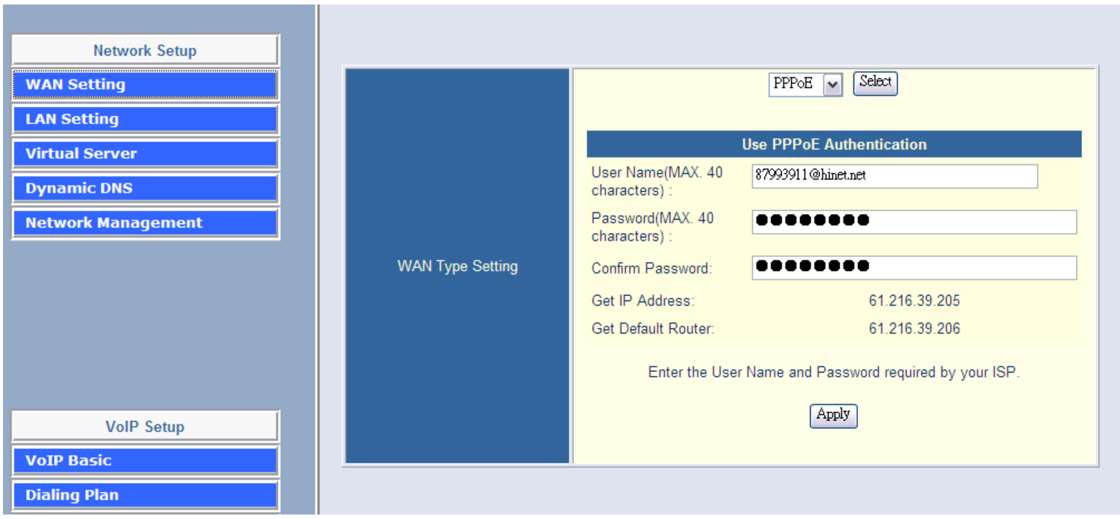
You are a leased line user with a fixed IP address; fill out the following items with the information provided by your ISP.



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

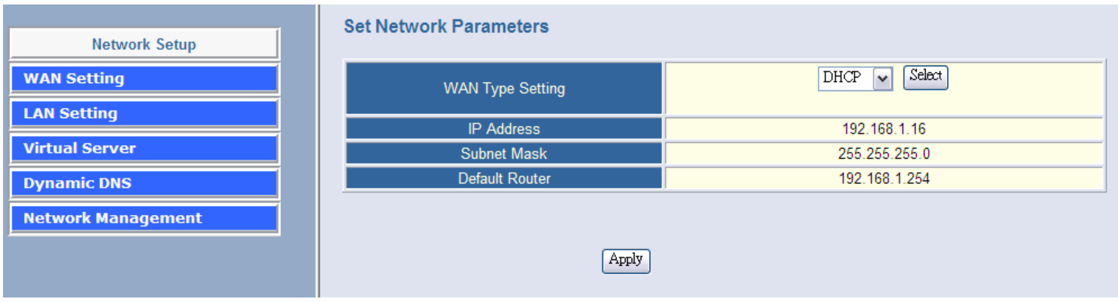
PPPoE for ADSL

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

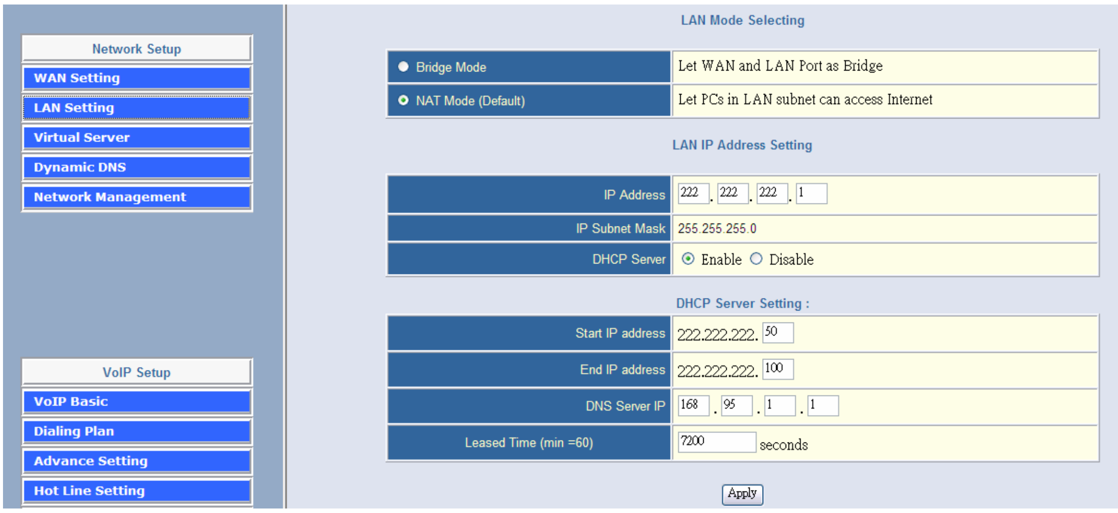


- **IP Address:** If you are connected to the Internet through a Cable modem line then a dynamic IP address will be assigned.

(Note : WAN port display the IP address, Subnet Mask and Default gateway IP address if DHCP client is successful)

4.1.2 Configuring LAN IP Address and DHCP Server

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



Bridge Mode:

Select this Gateway as Bridge. Let gateway Lan port like Switch/HUB. (WAN Port and LAN Port use the same IP address)

NAT Mode:

Each of the VoIP Gateway has two Ethernet interfaces, one is for connecting to local network users, and the other is for connecting to an external broadband device (i.e. DSL modem/router or Cable modem). The LAN port is connected to the local Ethernet network. WAN is connected to the external broadband device. The LAN IP address/subnet mask is for private users or NAT users, and the WAN IP address/subnet mask is for public users.

LAN IP Network Configuration:

- **IP Address:** Private IP address for connecting to a local private network (Default: 222.222.222.1).
- **Subnet Mask:** Subnet mask for the local private network (Default: 255.255.255.0).

DHCP Server Configuration:

DHCP stands for Dynamic Host Configuration Protocol. It can automatically dispatch related IP settings to any local user configured as a DHCP client. The DHCP server supports up to 253 users (PCs) on

Yes: Enables the DHCP server. **No:** Disables the DHCP server.

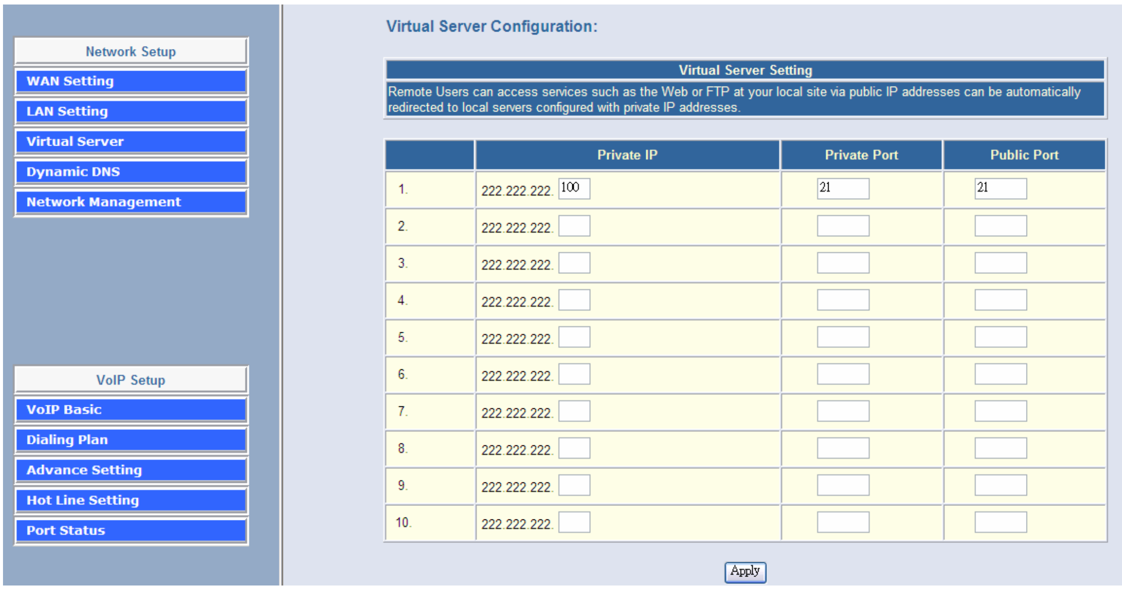
- **Start IP Address:** Sets the start IP address of the IP address pool.
- **End IP Address:** Sets the end of IP address in the IP address pool.
- **DNS Server IP Address:** DNS stands for Domain Name System. Every Internet host. must have a unique IP address, also they may have a human friendly, easy to remember name such as www.yahoo.com. The DNS server converts the human friendly name into it's equivalent IP address. (Default: None)
- **Primary IP Address:** Sets the IP address of the primary DNS server.
- **Secondary IP Address:** Sets the IP address of the secondary DNS server.

4.1.3 Virtual Server Setup

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

How to set a Virtual Server

The following example shows how an internal FTP server is exposed to the public domain. The internal FTP server is running on the local host addressed as 222.222.222.100.



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

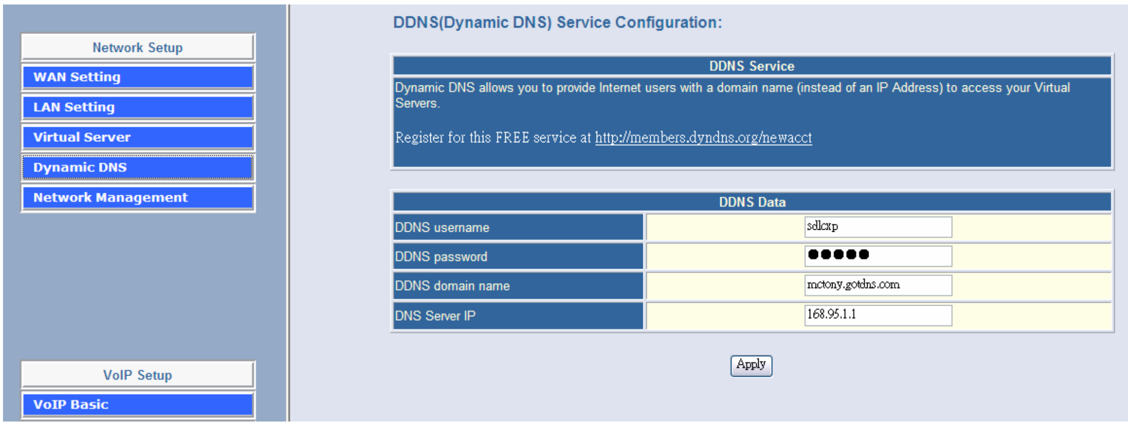
4.1.4 Dynamic DNS

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

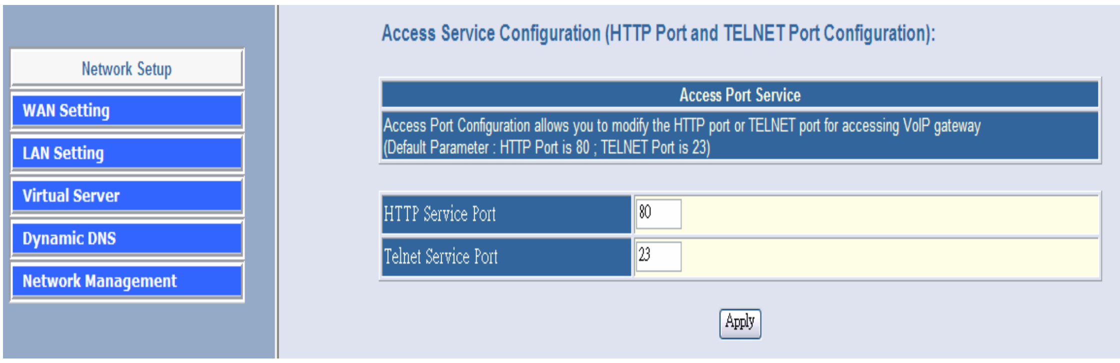
(Attention, if you use static IP address, you can't set DDNS in gateway. Use DDNS and Static IP at the same time, the dyndns will stop your DDNS service. Dyndns support DDNS service is Free, one account can create 5 different DDNS Domain Name)



- **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.

4.1.5 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)



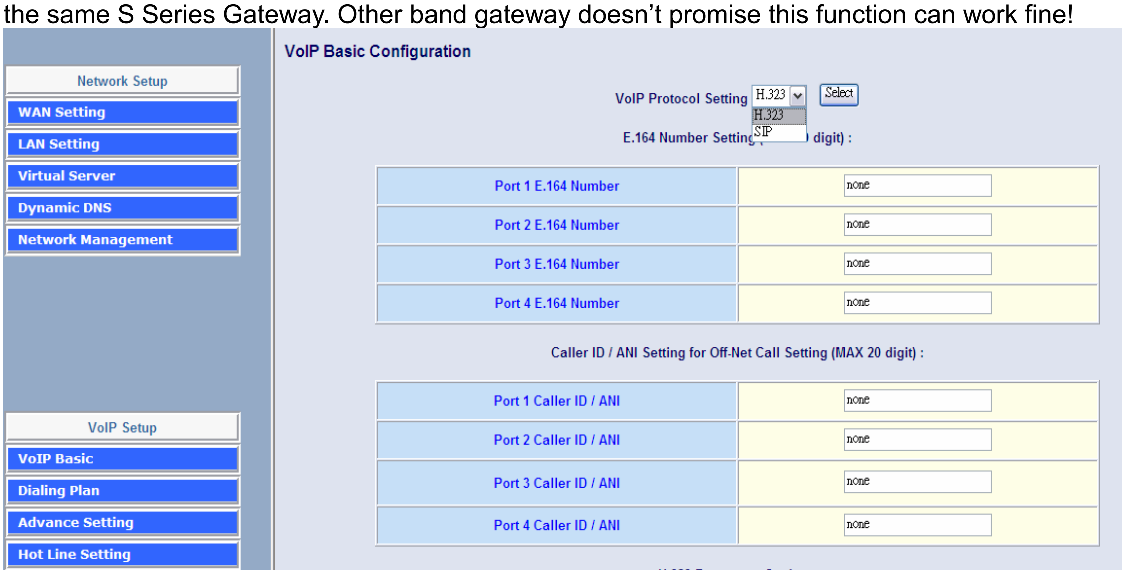
- **Http Server Port:** Input you want to change Web access port (Default is 80)
- **Telnet Server Port:** Input you want to change telnet access port (Default is 23)

4.2 VoIP Setup

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).

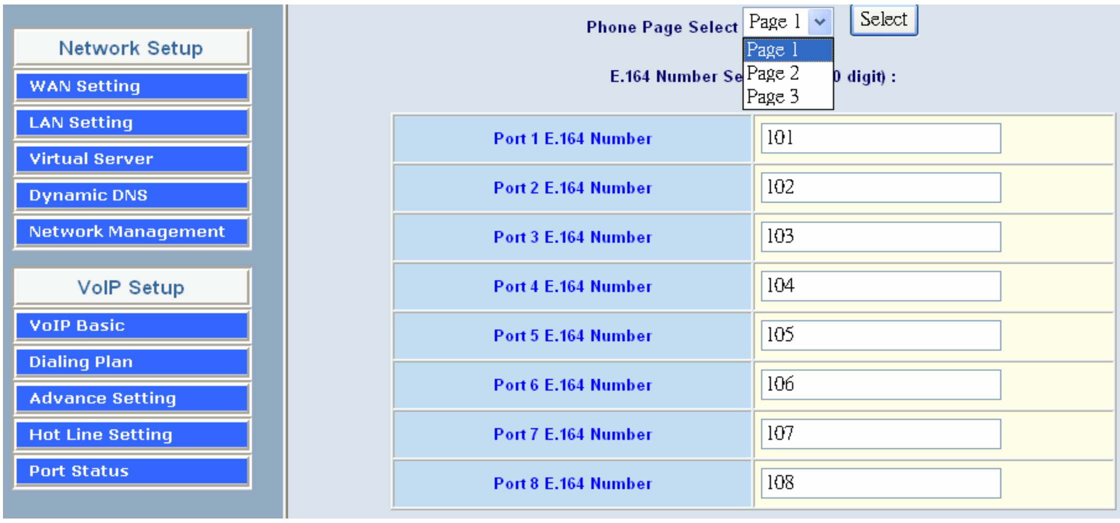
4.2.1 H.323 Setup

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. Select H.323 protocol, if you use S1600/S2400 Series, you can select different page to setting port number or others.

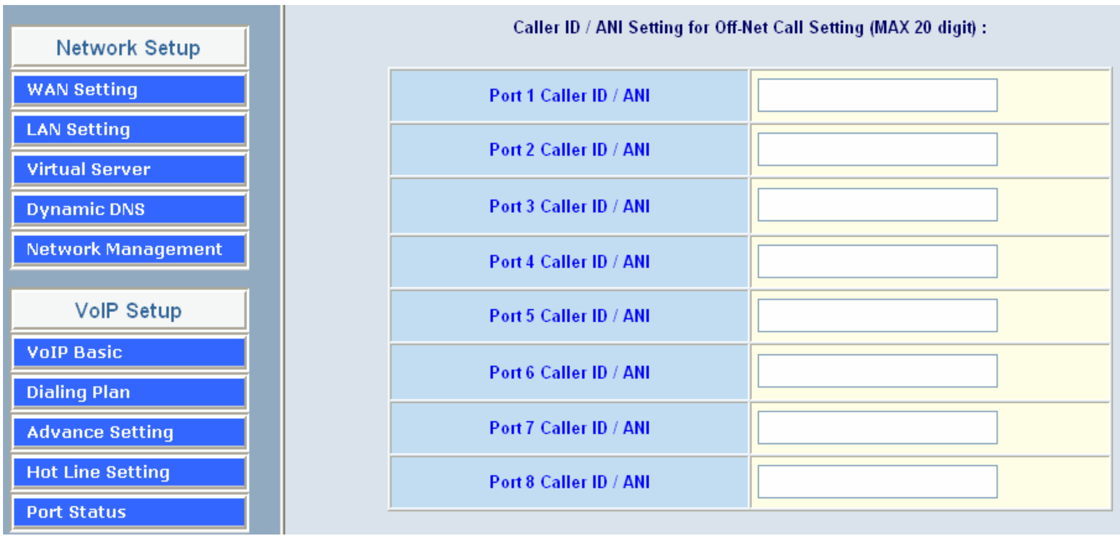
2. Configure the numbering with FXS / FXO ports.
(Depending on model: S1600 have 16 voice channels for setting, S2400 have 24 voice channels for setting)



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

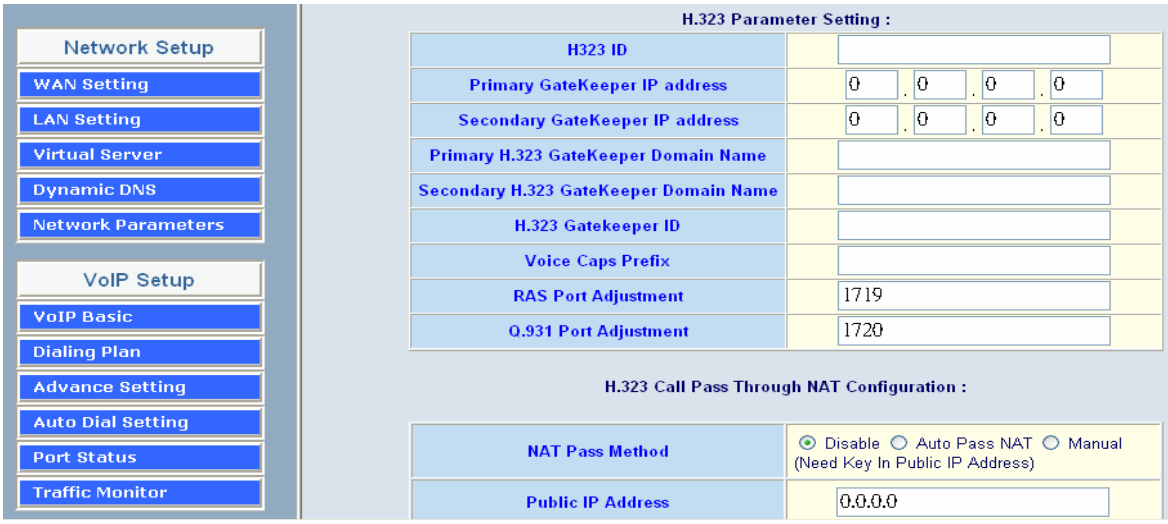
(Port number is in comparison with gateway port number. White Port socket is “FXO” port, Black Port socket is “FXS” port.)

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



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

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

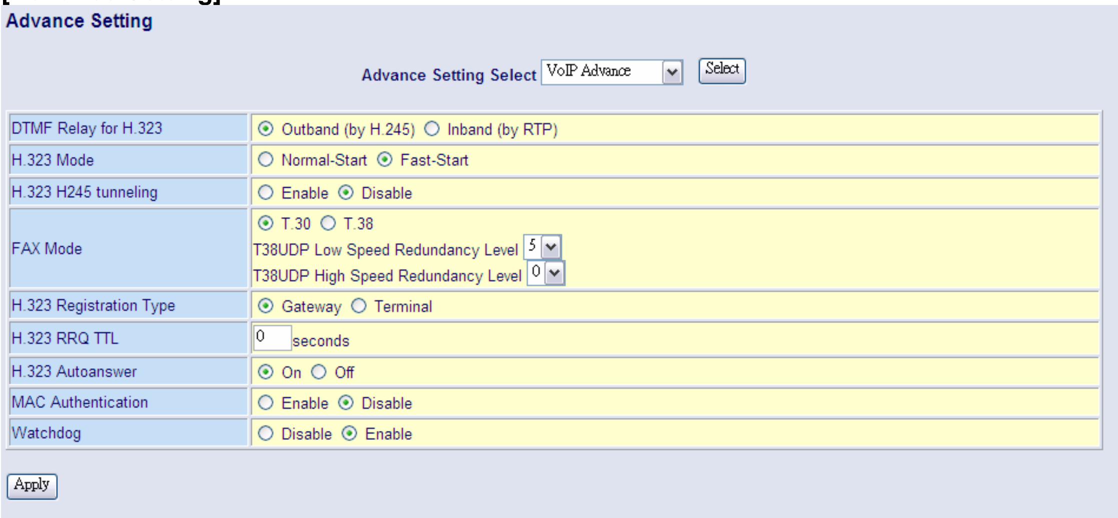


H.323 Parameters Label	
H.323 ID	Sets the unique name of this Gateway, that is communicated as part of H.323 messaging..
Primary Gatekeeper IP Address	There are two gatekeeper address fields, one is primary, the other secondary. If this gateway does not want to register to any gatekeeper, just set value 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.

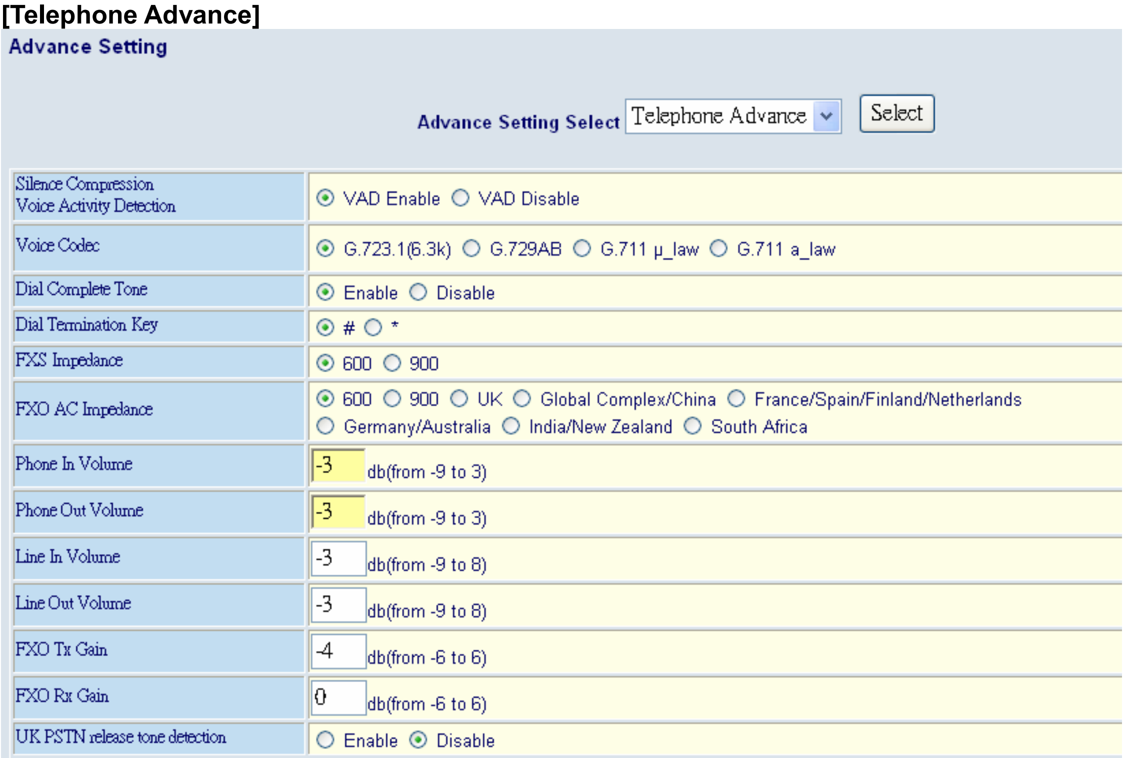
[Advance Setting]

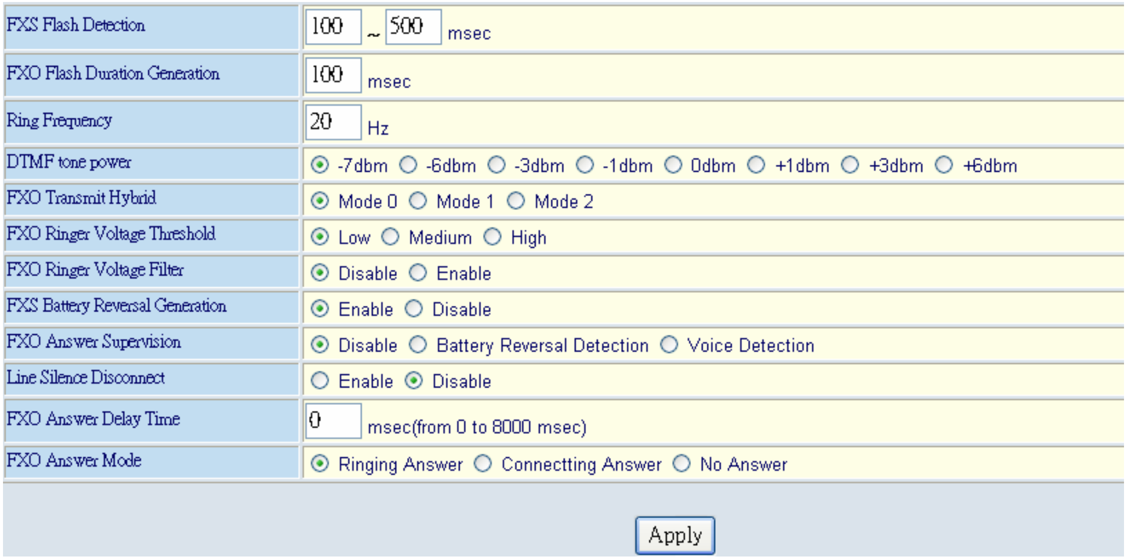


Item	Description
DTMF Relay for H.323:	After the VoIP call is connected, when you dial a digit, this digit is sent to the other side by DTMF tone. There are two methods of sending the DTMF tone. The first is "in band", that is, sending the DTMF tone in the voice packet. The other is "out band", that 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).
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)
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]

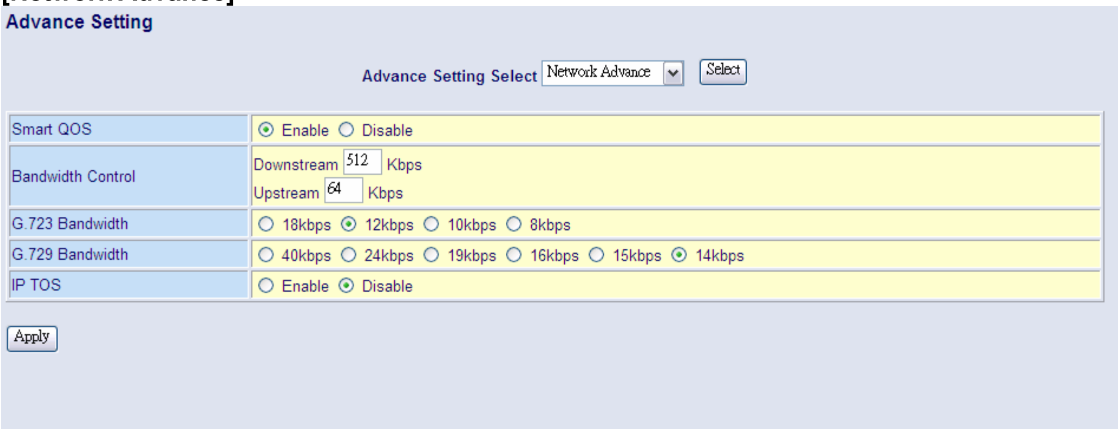




Item	Description
Silence Compression: (VAD)	If this function is enabled, when silence is occurred for a period of time, no data will be sent across the network during this period in order to save bandwidth. (If you use Asterisk, please disable Silence Compression, it maybe make you call disconnect.)
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.
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)
FXO Tx/Rx Gain:	You can adjust the FXO Tx/Rx Gain , range from -6db to 6db. (If you adjust too bigger, maybe generation some ECHO or noise)
UK PSTN release tone detection:	When you use the Gateway to UK, you can Enable this selection to detection release tone
FXO/FXS Flash detection and generation duration :	Flash Detection: Let you change flash detection (milliseconds) of Gateway when phone generate flash to FXS. Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.
Ring Frequency:	You can configure how long the Ring Frequency do you want to use.
DTMF tone power:	Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db.
FXO Transmit Hybrid:	ECHO cancellation adjust, 3 mode setting to solve ECHO problem, default is Mode 1. If you have echo problem, try to select

	other Option.
FXO Ringer Voltage Threshold:	Sometime you use FXO connect to PSTN/PBX can't be pick up the call. PSTN/PBX can't detection FXO voltage, you can adjust this function. Default is low.(Low : 16.5 Vrms Medium: 24 Vrms High: 49.5 Vrms)
FXO Ringer Voltage filter:	Some vendor's PBX generates the leakage voltage from extension port. That will mislead the FXO become Off-hook status. This function was set to avoiding a leakage voltage signal is detected as ring coming.
FXS Battery Reversal Generation:	Some Payphone need Battery Reversal to count call money. Gateway FXS can generation that Battery Reversal. If you don't use this function , don't enable it.
FXO Answer Supervision	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. SIP case: Sending the 200 OK connect signal to caller when detecting polarity reverse from PSTN Line. Voice Detection: when the FXO connect PSTN/PBX, gateway will detection the PSTN/PBX send voice , it will send Q.931 connect or SIP 200 ok.
Line Silence Disconnect:	When FXO call have silence, the call will disconnect automatically. {Default is Disable}
FXO Answer Delay Time	Some PSTN/PBA when call incoming can't pick up first time. It need delay few second time. You can set FXO pick up call delay time.(default is 0)
FXO Answer Mode	FXO Answer Mode Concept: When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 1. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line. 2. Connecting Answer Mode: **Case A:** "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line. **Case B:** "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.) **Case C:** "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user. (Note: This case can avoid the Local PSTN charge when the FXS port still ring.) 3. Non Answer Mode: FXO will NOT answer the call in any time. (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

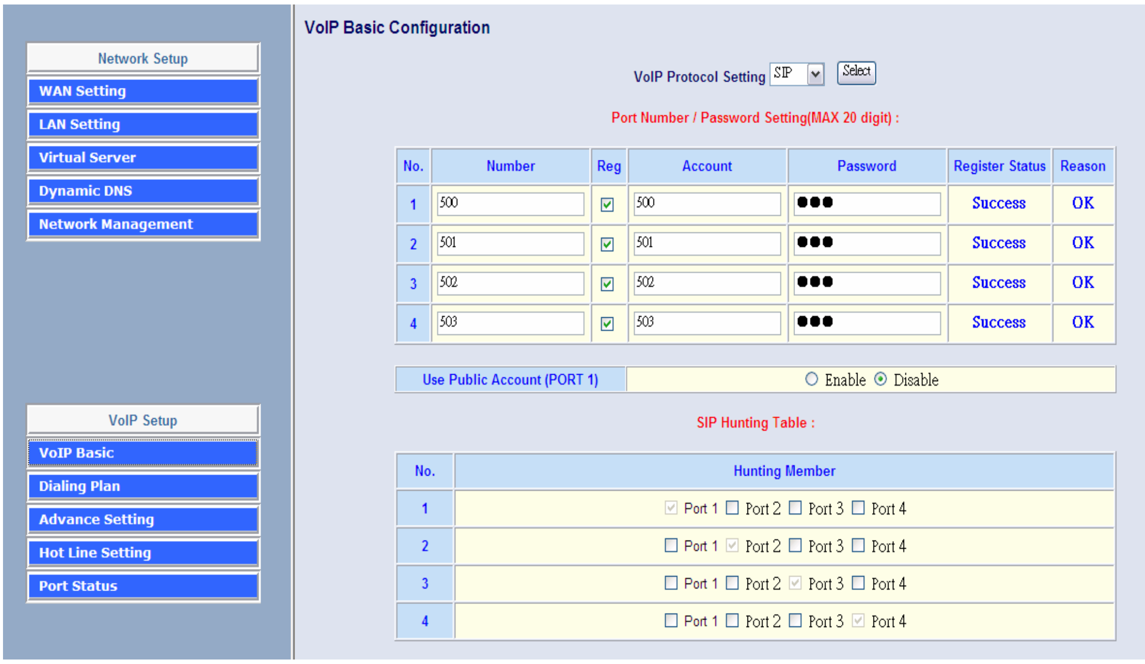
[Network Advance]



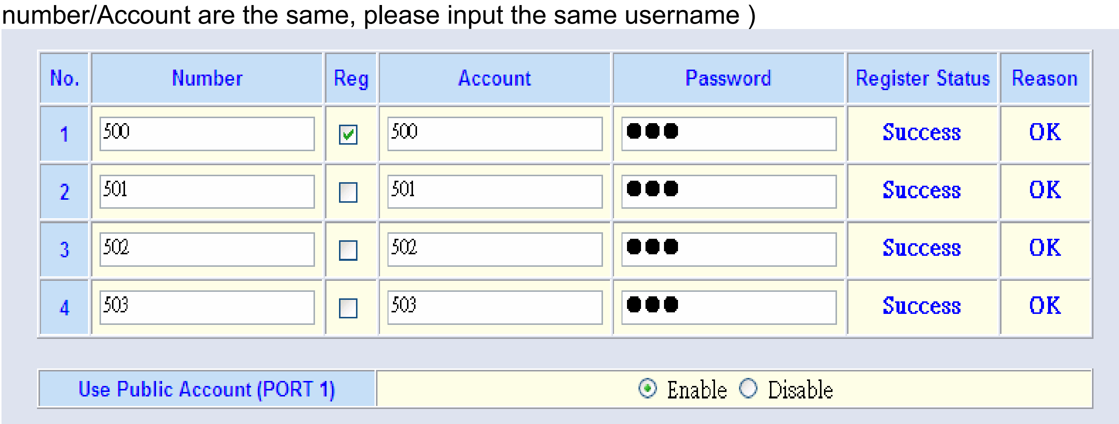
Item	Description
Smart-QoS:	If this function is enabled, when VoIP call is occurred, the other data will be automatically reduced traffic which across the internet in order to guarantee the voice bandwidth.
Bandwidth control:	You can configure your bandwidth what the Max byte of download and upload of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

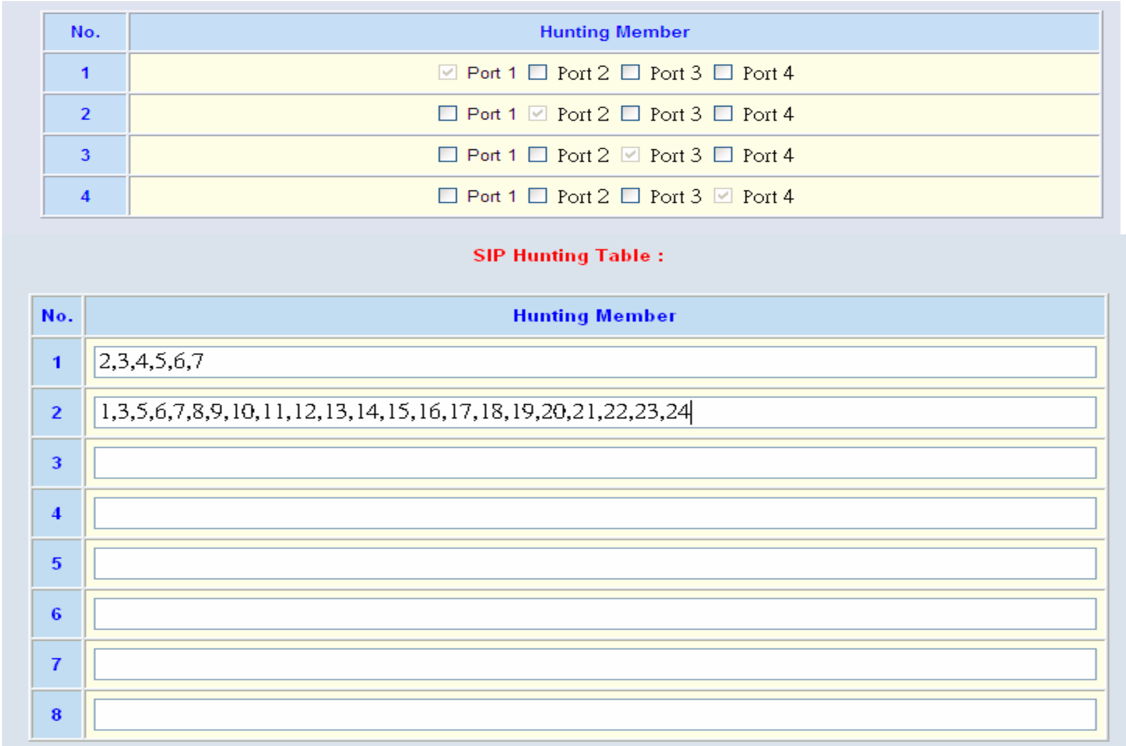
Gateway SIP support SIP(RFC3261), SDP(RFC2327), RFC2833, STUN(RFC3489), Symmetric RTP, outbound proxy, ENUM(RFC2916),and RTP/RTCP.SIP NAT pass through Function can support 80% NAT/Firewall that you don’t setting DMZ/Virtual server in router or Firewall.



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)

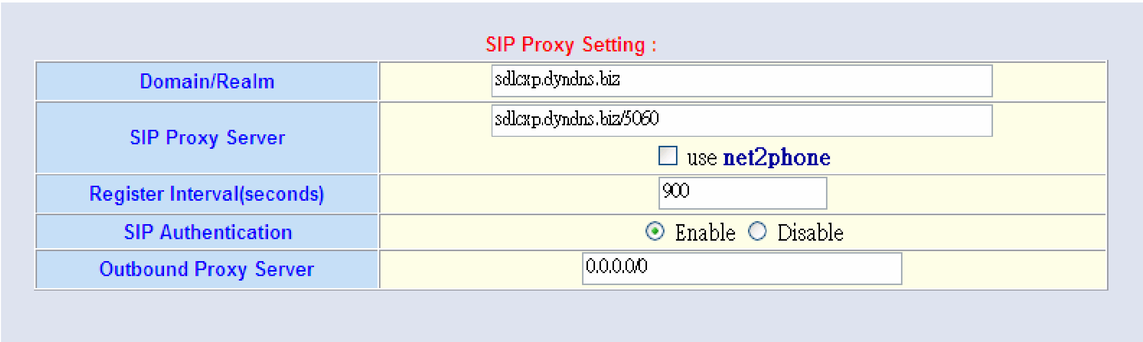


- **Number:** Input SIP Number(Username), if your server support account and number (different),input the number, else number/account are the same username.
- **Reg:** let your sip account register SIP Server, click this option.
- **Account:** Input SIP account(Username), if your server support account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



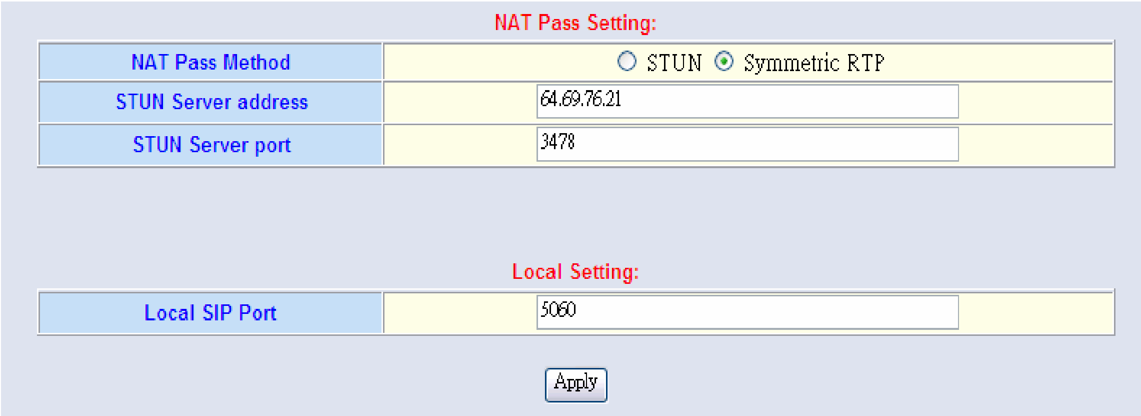
- **SIP Hunting Table:** This allows gateway can answer SIP call from internet by Hunting. For example: Port 2-7 are hunting for the Port 1 SIP account. When port 1 are on call, the other one SIP call from internet will ring port 2.S200/400/800 Series hunting table is use click box.

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.



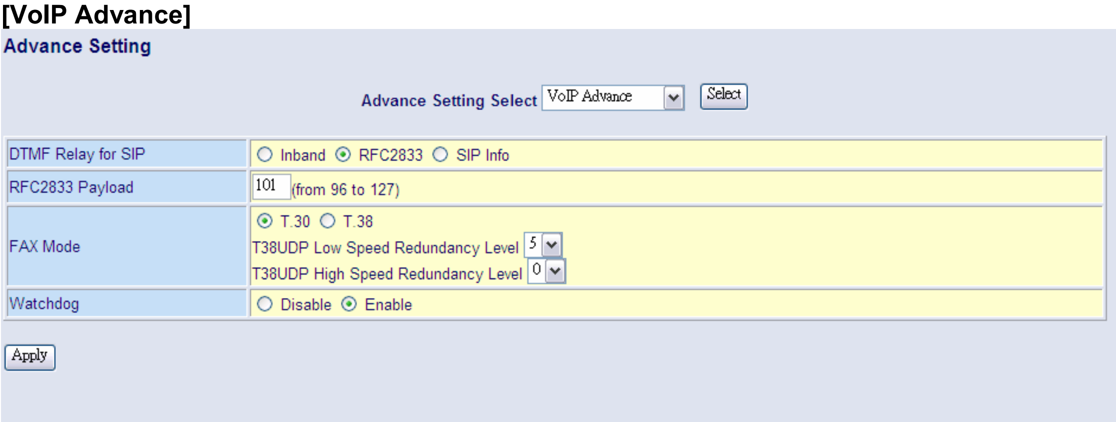
- **Symmetric RTP:** default use Nat pass function.
- **STUN Client:** setting your STUN server information, default STUN server is FWD STUN server.
- **Outbound Proxy Support:** Setting your Outbound Proxy server information.

- **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.

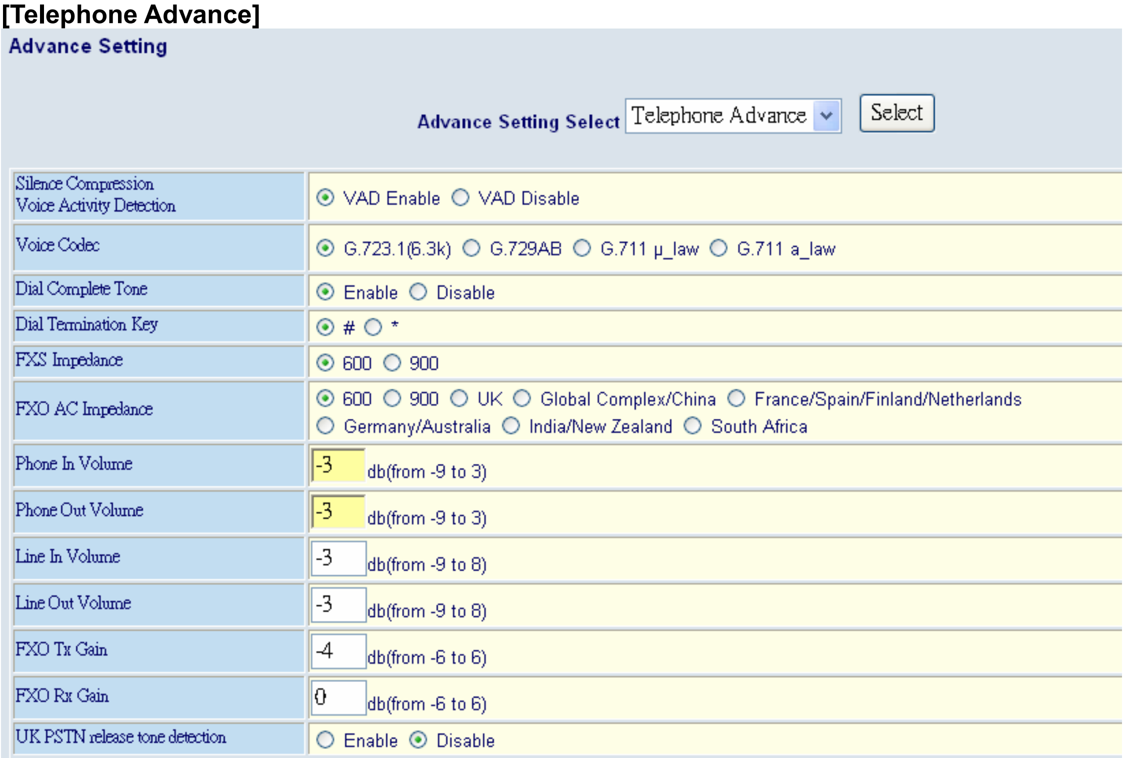
[VoIP Advance]



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.

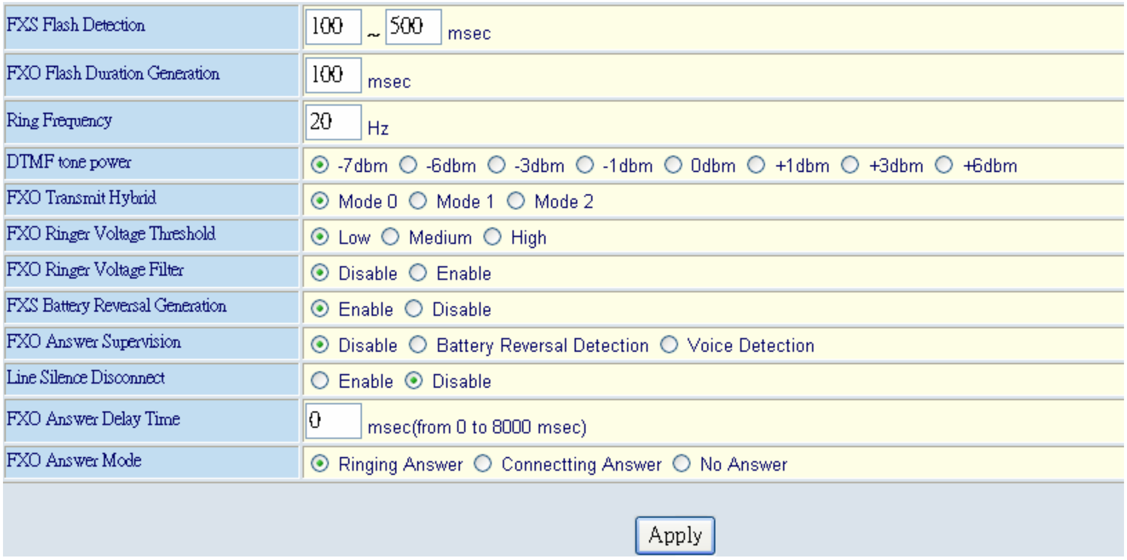
[Telephone Advance]



Advance Setting

Advance Setting Select: Telephone Advance [Select]

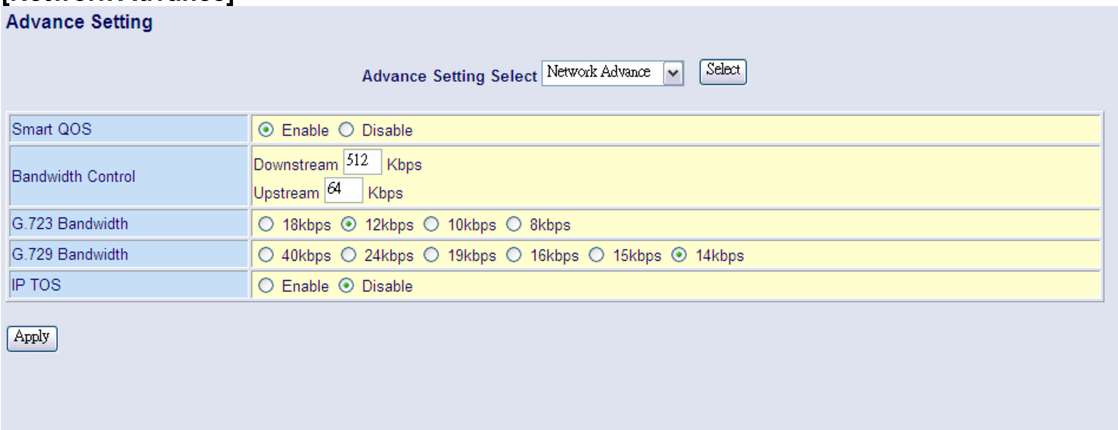
Silence Compression Voice Activity Detection	◉ VAD Enable ○ VAD Disable
Voice Codec	◉ G.723.1(6.3k) ○ G.729AB ○ G.711 µ_law ○ G.711 a_law
Dial Complete Tone	◉ Enable ○ Disable
Dial Termination Key	◉ # ○ *
FXS Impedance	◉ 600 ○ 900
FXO AC Impedance	◉ 600 ○ 900 ○ UK ○ Global Complex/China ○ France/Spain/Finland/Netherlands ○ Germany/Australia ○ India/New Zealand ○ South Africa
Phone In Volume	-3 db(from -9 to 3)
Phone Out Volume	-3 db(from -9 to 3)
Line In Volume	-3 db(from -9 to 8)
Line Out Volume	-3 db(from -9 to 8)
FXO Tx Gain	-4 db(from -6 to 6)
FXO Rx Gain	0 db(from -6 to 6)
UK PSTN release tone detection	○ Enable ◉ Disable



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.
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)
FXO Tx/Rx Gain:	You can adjust the FXO Tx/Rx Gain , range from -6db to 6db. (If you adjust too bigger, maybe generation some ECHO or noise)
UK PSTN release tone detection:	When you use the Gateway to UK, you can Enable this selection to detection release tone
FXS/FXO Flash detection and generation duration :	Flash Detection: Let you change flash detection (milliseconds) of Gateway when phone generate flash to FXS. Flash Generation: Let you change flash generation time (milliseconds) for PBX detection.
Ring Frequency:	You can configure how long the Ring Frequency do you want to use.
DTMF tone power:	Sometimes you input DTMF, but no request. You can adjust this function, range from -6db to +6db.

FXO Transmit Hybrid:	ECHO cancellation adjust, 3 mode setting to solve ECHO problem, default is Mode 1. If you have echo problem, try to select other Option.
FXO Ringer Voltage Threshold:	Sometime you use FXO connect to PSTN/PBX can't be pick up the call. PSTN/PBX can't detection FXO voltage, you can adjust this function. Default is low.(Low : 16.5 Vrms Medium: 24 Vrms High: 49.5 Vrms)
FXO Ringer Voltage filter:	Some vendor's PBX generates the leakage voltage from extension port. That will mislead the FXO become Off-hook status. This function was set to avoiding a leakage voltage signal is detected as ring coming.
FXS Battery Reversal Generation:	Some Payphone need Battery Reversal to count call money. Gateway FXS can generation that Battery Reversal. If you don't use this function, don't enable it.
FXO Answer Supervision:	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. SIP case: Sending the 200 OK connect signal to caller when detecting polarity reverse from PSTN Line. Voice Detection: when the FXO connect PSTN/PBX, gateway will detection the PSTN/PBX send voice , it will send Q.931 connect or SIP 200 ok.
Line Silence Disconnect:	When FXO call have silence, the call will disconnect automatically. {Default is Disable}
FXO Answer Delay Time	Some PSTN/PBA when call incoming can't pick up first time. It need delay few second time. You can set FXO pick up call delay time.(default is 0)
FXO Answer Mode	FXO Answer Mode Concept: When user calls the PSTN line which was connected with the FXO port, there are three answer mode for user to configure. 4. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line. 5. Connecting Answer Mode: **Case A:** "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line. **Case B:** "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device. In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call. (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.) **Case C:** "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user. (Note: This case can avoid the Local PSTN charge when the FXS port still ring.) 6. Non Answer Mode: FXO will NOT answer the call in any time. (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

[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

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
		~					ADD

DELETE Outbound Dial Plan From To

- "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

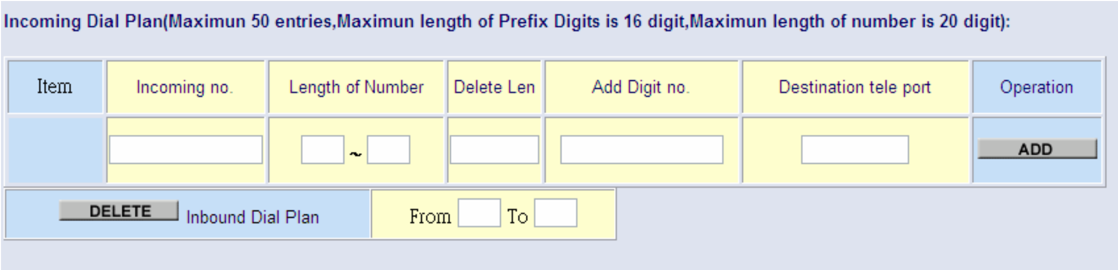
Example2: Speed Dial

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

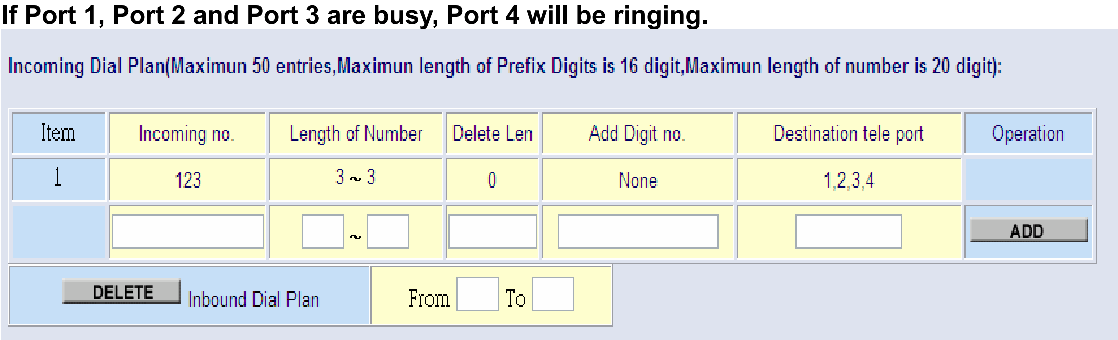
DELETE Outbound Dial Plan From To

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



- “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.

Example1: Hunting for FXS Port
Port 1: FXS
Port 2: FXS
Port 3: FXS
Port 4: FXS
H.323 (SIP) number “123”call incoming, Port 1 will be ringing.
If Port 1 is busy, Port will be ringing.
If Port 1 and 2 are busy, Port 3 will be ringing.
If Port 1, Port 2 and Port 3 are busy, Port 4 will be ringing.



Example2: Hunting for FXO Port
Port 1: FXO was connected to PSTN.
Port 2: FXO was connected to PSTN.
Port 3: FXO was connected to PSTN.
Port 4: FXO was connected to PSTN.
H.323 (SIP) number “123”call incoming, Port 1 will be off-hook and hear the Dial Tone from PSTN.
If Port 1 is busy, Port will be will be off-hook and hear the Dial Tone from PSTN.
If Port 1 and 2 are busy, Port 3 will be off-hook and hear the Dial Tone from PSTN.
If Port 1, Port 2 and Port 3 are busy, Port 4 will be off-hook and hear the Dial Tone from PSTN.

Incoming Dial Plan(Maximun 50 entries,Maximun length of Prefix Digits is 16 digit,Maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Len	Add Digit no.	Destination tele port	Operation
1	123	3 ~ 3	0	None	1,2,3,4	
		~				ADD
DELETE Inbound Dial Plan	From To					

Example3: Termination Call to FXO for One-Shoot Call
Port 1: FXO was connected to PSTN (area code is 81xxxxxxxx).
H.323 (SIP) leading number "081x"incoming, and delete the first one digit "0", and call to PSTN number.

Incoming Dial Plan(Maximun 50 entries,Maximun length of Prefix Digits is 16 digit,Maximun length of number is 20 digit):

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

Example4: Termination Call to FXO
Port 1: FXS
Port 1: FXO was connected to PSTN (area code is 92xxxxxxxx).
Port 1 FXS call to "092x" to PSTN, FXO port will delete the first one digit "0" and call to PSTN number.

Incoming Dial Plan(Maximun 50 entries,Maximun length of Prefix Digits is 16 digit,Maximun length of number is 20 digit):

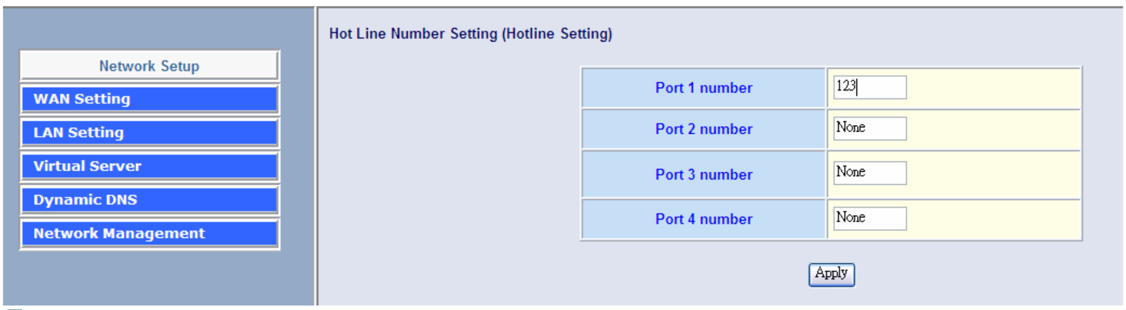
Item	Incoming no.	Length of Number	Delete Len	Add Digit no.	Destination tele port	Operation
1	092x	4 ~ 20	1	None	2	
		~				ADD
DELETE Inbound Dial Plan	From To					

([x]: mean wild card, 0~9. Speed dial: When you dial over the number, input "#" key can speed dial call, or you key in number as the length as the "length of number")

4.2.4 Other VoIP Setting

Hot Line:

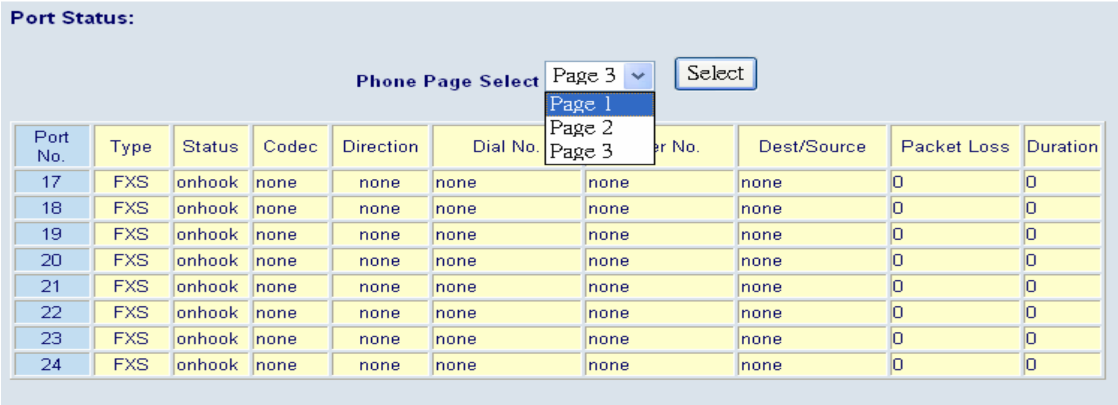
You can setting hot line. when the call incoming the hot line port, it will call hot line number automatically. The hot line call the number via VoIP, so you setting the hot line number must VoIP number. Usually, you want to FXO port call transfer to FSX, you only setting the FXO hot line to FXS number.



- **Port number:** Input FXS/FXO 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. (SB800/S1600/S2400 Series can change page to view all ports)



Port Status:

Phone Page Select Page 3 Select

Page 1
Page 2
Page 3

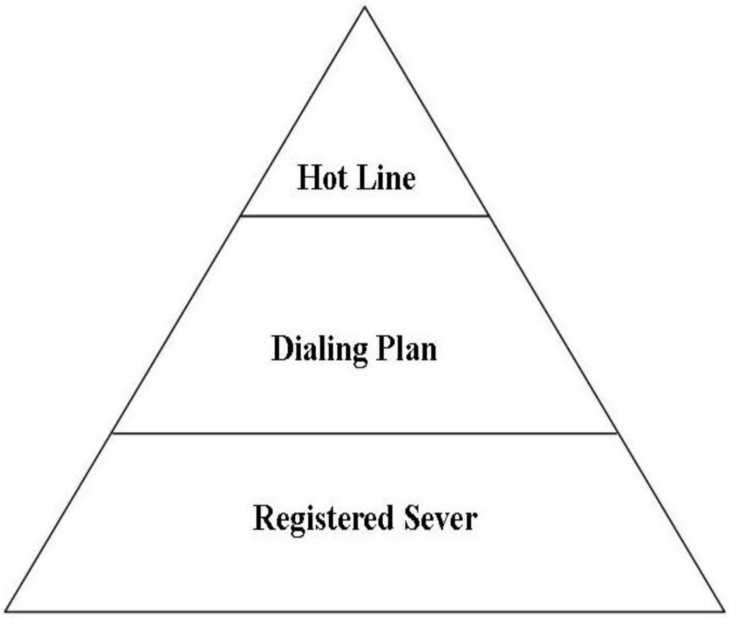
Port No.	Type	Status	Codec	Direction	Dial No.	r No.	Dest/Source	Packet Loss	Duration
17	FXS	onhook	none	none	none	none	none	0	0
18	FXS	onhook	none	none	none	none	none	0	0
19	FXS	onhook	none	none	none	none	none	0	0
20	FXS	onhook	none	none	none	none	none	0	0
21	FXS	onhook	none	none	none	none	none	0	0
22	FXS	onhook	none	none	none	none	none	0	0
23	FXS	onhook	none	none	none	none	none	0	0
24	FXS	onhook	none	none	none	none	none	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:

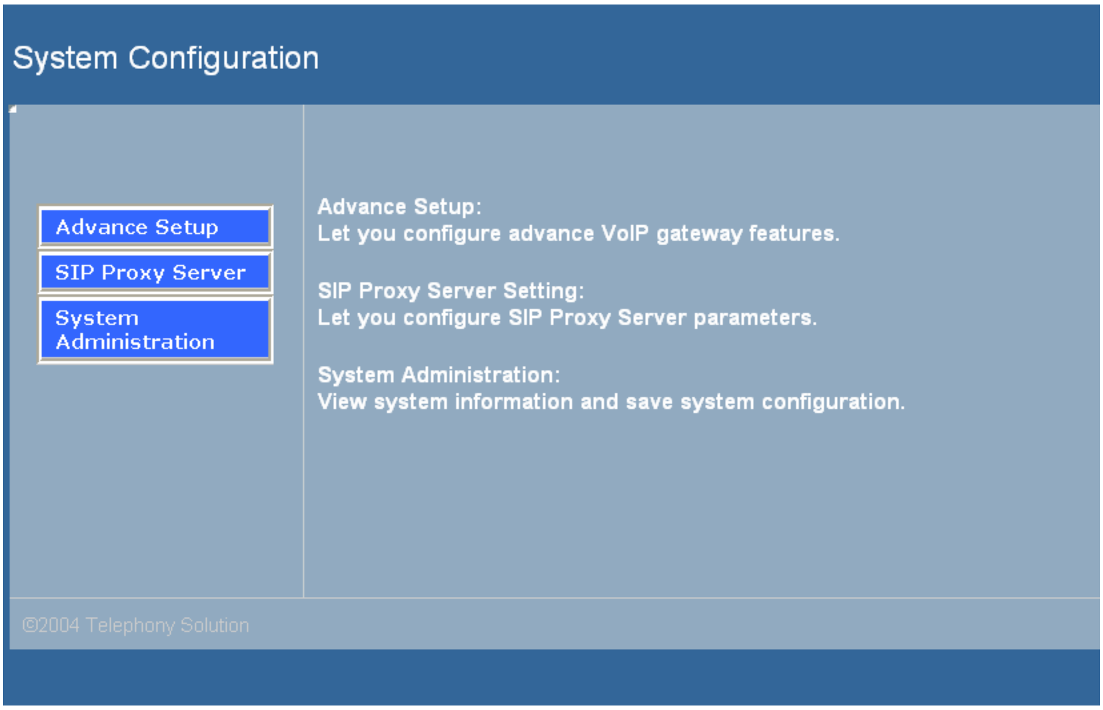
Gateway have a rule for call Priority, up to down is 1)Hot Line 2)Dialing plan 3)Registered server(SIP Proxy / H.323 Gatekeeper).When a VoIP call made, Gateway will process by Hot Line first, then it will check the dialing plan table, last fine Server(SIP/H.323).

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…..

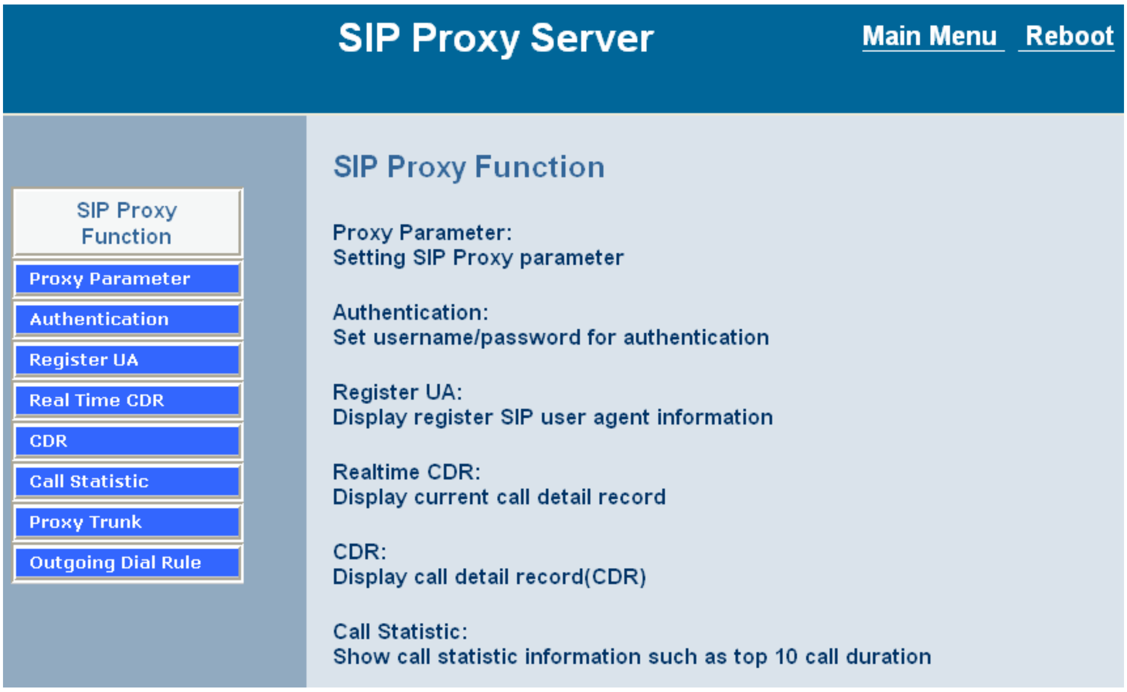


4.3 SIP Proxy Server (SVR) Setup

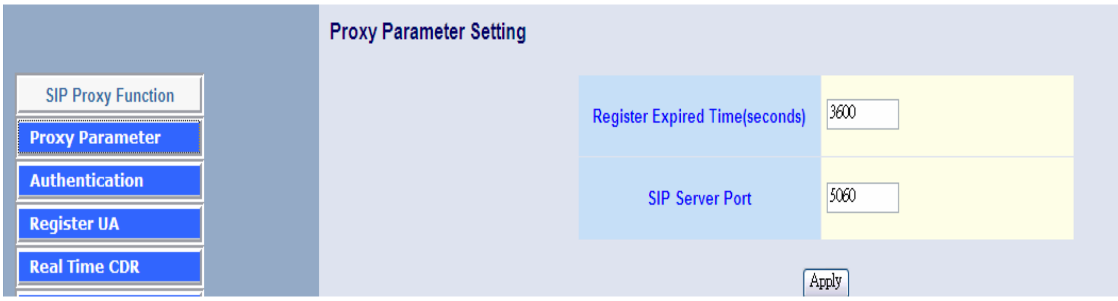
The SVR200/SVR400/SVR800 Series, a Hybrid SIP Proxy Server, registers and authenticates users, and routes calls between user agents. With SVR400/SVR200/SVR800 Series SIP Server, you can use SIP Agents, SIP soft phones, and SIP Gateways for VoIP communications. The SVR400/SVR200/SVR800 Series also provide Trunk Route capability to call Least Cost Route to ITSP or PSTN line. Otherwise SVR400/SVR200/SVR800 Series were implemented telephone interface connected with PSTN, PBX or telephone set for SIP Communication.



1. Select SIP Proxy Server option and setting Proxy server.

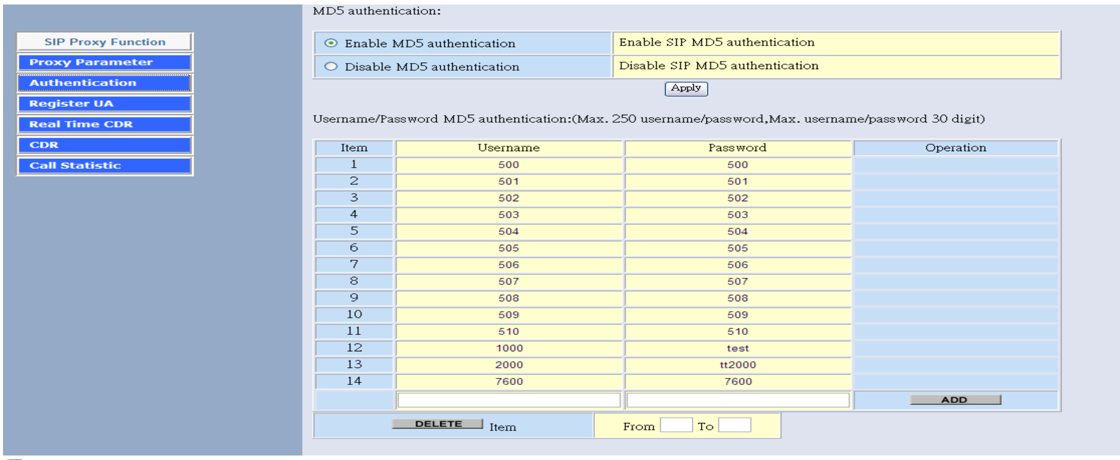


2. Setting Proxy Parameter



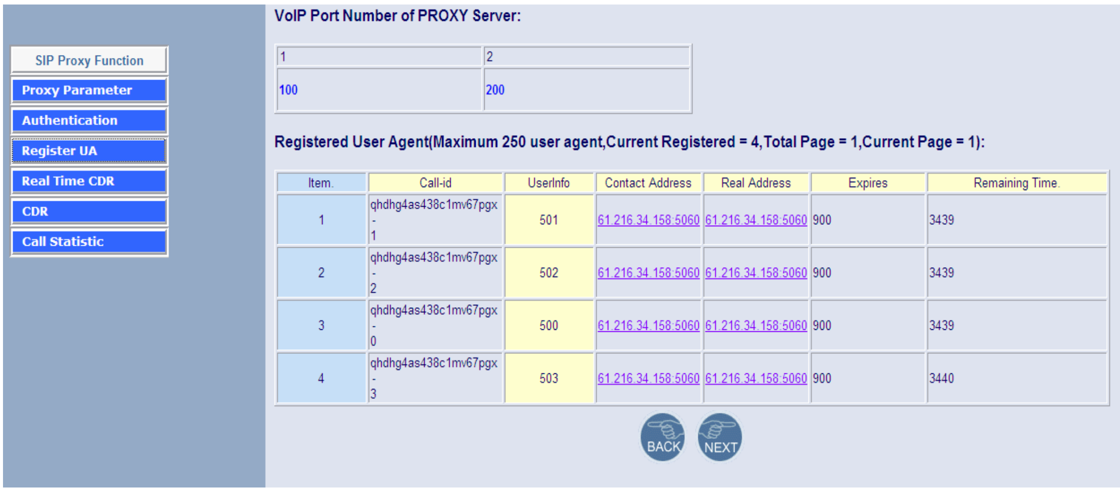
- **Register Expired Time(seconds):** This field sets how long an entry remains registered with the SIP register server. The register server can use a different time period. The Gateway sends another registration request after half of this configured time period has expired.
- **SIP Server Port:** Setting the Proxy service Port, default port is 5060.

3. Setting Authentication, if you want to use MD5 Authentication , add user and password in table.

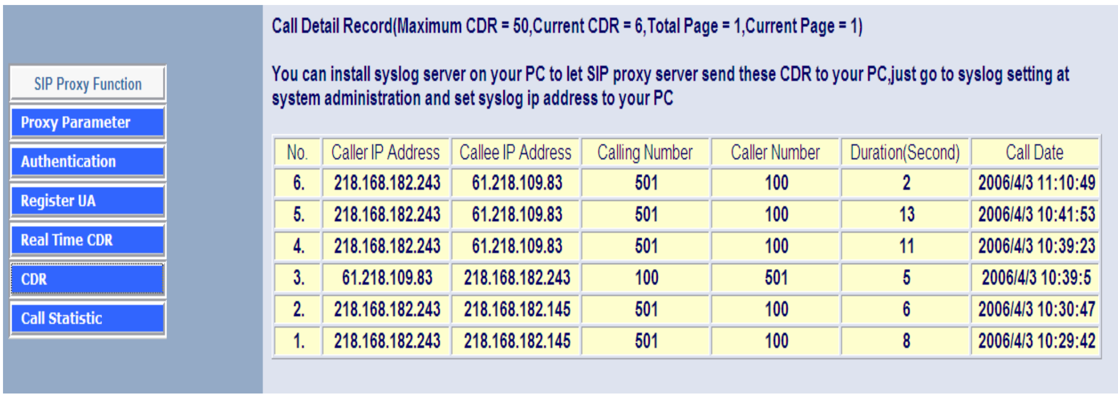


- **Enable/Disable MD5 Authentication:** If you want to UA registered SVR have Authentication, Enable this option.
- **Add Username/Password:** Add Username / Password for Authentication.
  (Max 250 username / password and Max username/password is 30 digit)

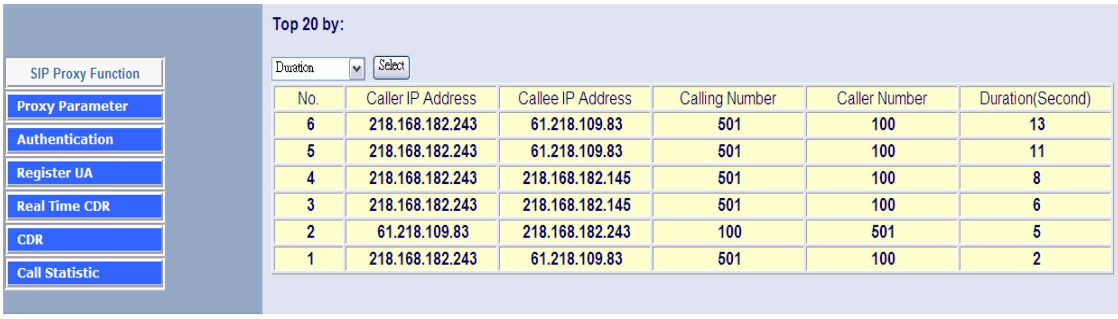
4. If you want to view gateway registered information, select "Register UA" option.



5. If you want to view CDR information (Real Time), select "CDR (Real Time)" option.



6. If you want to know what is the TOP (Max 20), select "call statistic" option



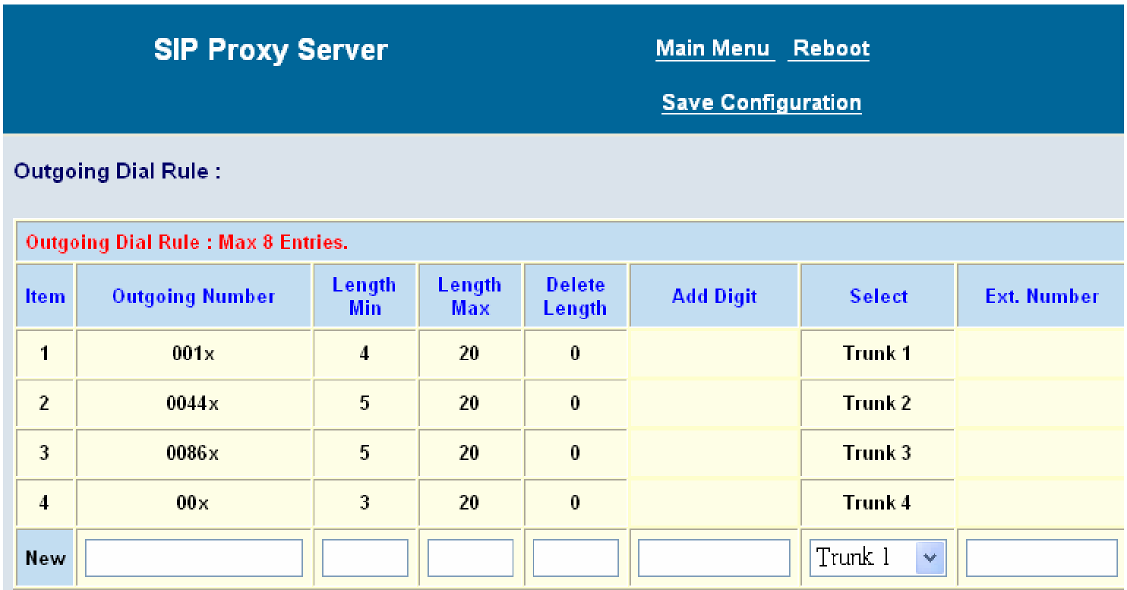
No.	Caller IP Address	Callee IP Address	Calling Number	Caller Number	Duration(Second)
6	218.168.182.243	61.218.109.83	501	100	13
5	218.168.182.243	61.218.109.83	501	100	11
4	218.168.182.243	218.168.182.145	501	100	8
3	218.168.182.243	218.168.182.145	501	100	6
2	61.218.109.83	218.168.182.243	100	501	5
1	218.168.182.243	61.218.109.83	501	100	2

7. If you want to register Multi-SIP Proxy Server, select “trunk” option.

- **ITSP Trunk:** There are 4 trunk lines for user t configure. Each trunk have two types:
  A. ITSP registering: This SVR can register to another one ITSP Proxy Server for call international calls to save the call cost.

Termination Trunk: Setting the Proxy service Port, default port is 5060.
This SVR can route the calls to terminate gateway for call termination.

8. If you want to set Dialing Rule for Call Routing, select "Outgoing Dial Rule" option.



Item	Outgoing Number	Length Min	Length Max	Delete Length	Add Digit	Select	Ext. Number
1	001x	4	20	0		Trunk 1	
2	0044x	5	20	0		Trunk 2	
3	0086x	5	20	0		Trunk 3	
4	00x	3	20	0		Trunk 4	
New						Trunk 1	

Outgoing Dial Rule Setting: SVR can register to another one ITSP Proxy Server for call international calls to save the call cost.

Least Cost Trunk Route: SVR200 / SVR 400 /SVR 800 series provide 4 Trunk Route Setting

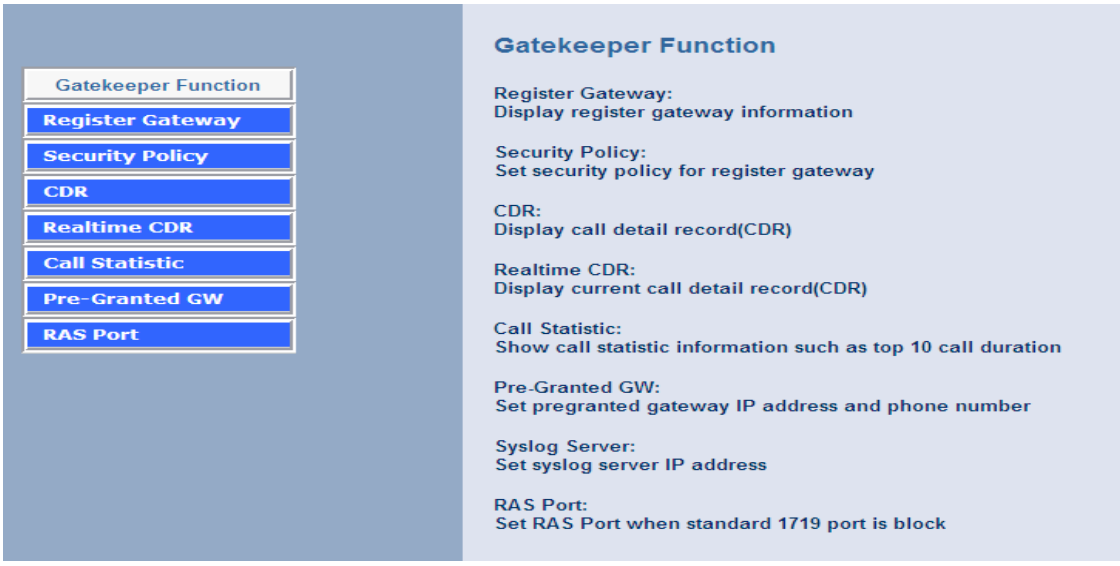
for Off-Net call to ITSP or Termination Gateway. (SVR series can subscribe to most 4 Service provider)

For example, you can send all international calls through 4 different ITSP by the least call rate. The feature can select the most effective service provider by call route rule setting.

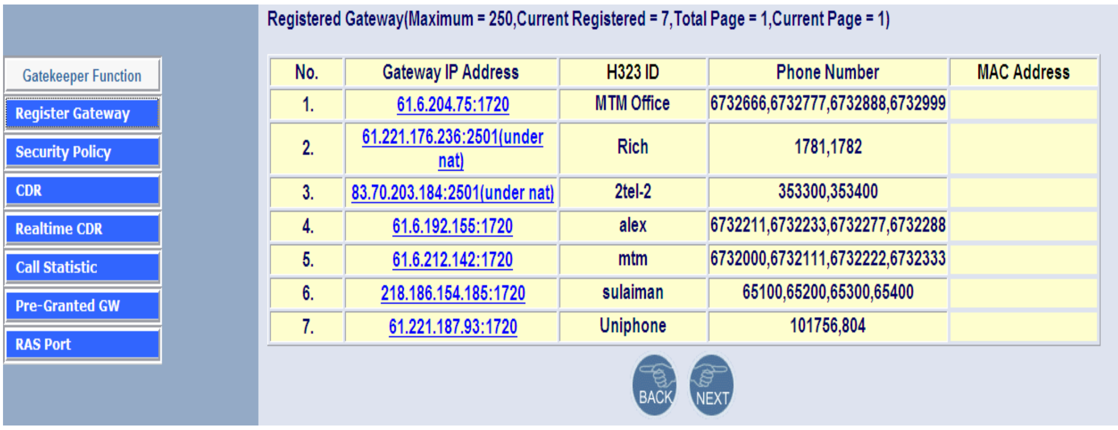
4.4 H.323 Gatekeeper (GK) Setup

2/4/8 port gateway can embedded H.323 Gatekeeper (GK). GK are both H.323 Gatekeeper and Gateway at the same box. GK support simple security, 200 UA registered, but it no group call and H.235 security Authentication.

1. Select Gatekeeper option and setting gatekeeper.

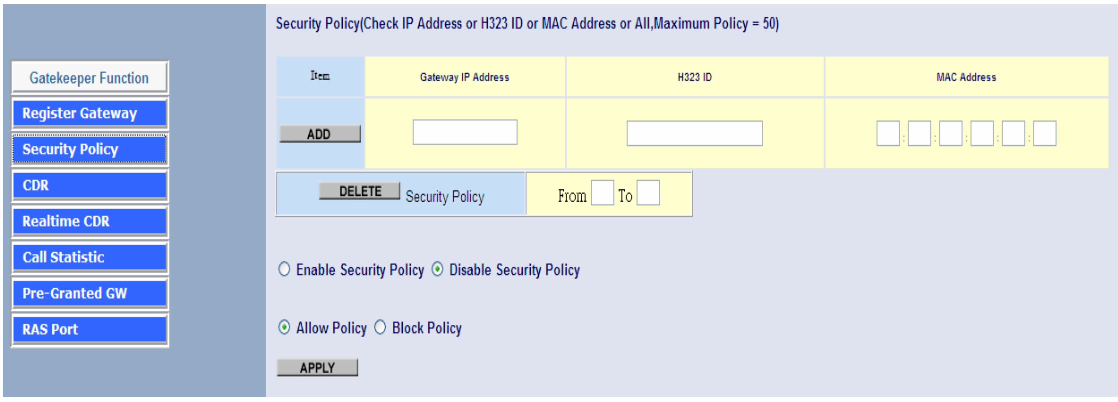


2. Register Gateway, this can view all registered Gateway information.



No.	Gateway IP Address	H323 ID	Phone Number	MAC Address
1.	61.6.204.75:1720	MTM Office	6732666,6732777,6732888,6732999	
2.	61.221.176.236:2501(under nat)	Rich	1781,1782	
3.	83.70.203.184:2501(under nat)	2tel-2	353300,353400	
4.	61.6.192.155:1720	alex	6732211,6732233,6732277,6732288	
5.	61.6.212.142:1720	mtm	6732000,6732111,6732222,6732333	
6.	218.186.154.185:1720	sulaiman	65100,65200,65300,65400	
7.	61.221.187.93:1720	Uniphone	101756,804	

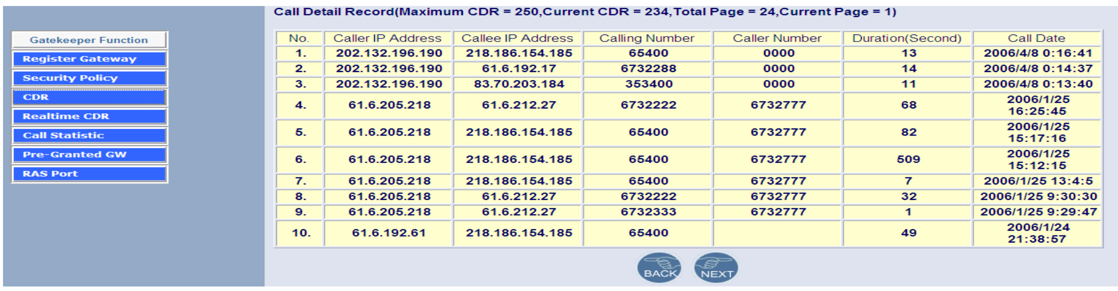
3. Setting simple security, if you want to gateway authentication before registered, you can use this function.



- **Gateway IP address:** Input you allow registered gateway IP address, if gateway not in this IP address, they can't registered.
- **H.323 ID:** Input H.323 ID for Authentication, gateway must send H.323 ID to GK check, H.323 ID just like "Password".
- **MAC address:** Input you allow MAC address to authentication.
- **Enable/Disable Security Policy:** if you enable the option, you gateway must send security information to registered GK, even if you don't set any Security Policy.
- **Allow/Block Policy:** This option allow you can registered GK with Security Policy, but you don't set any Security Policy, gateway can registered GK without security information.

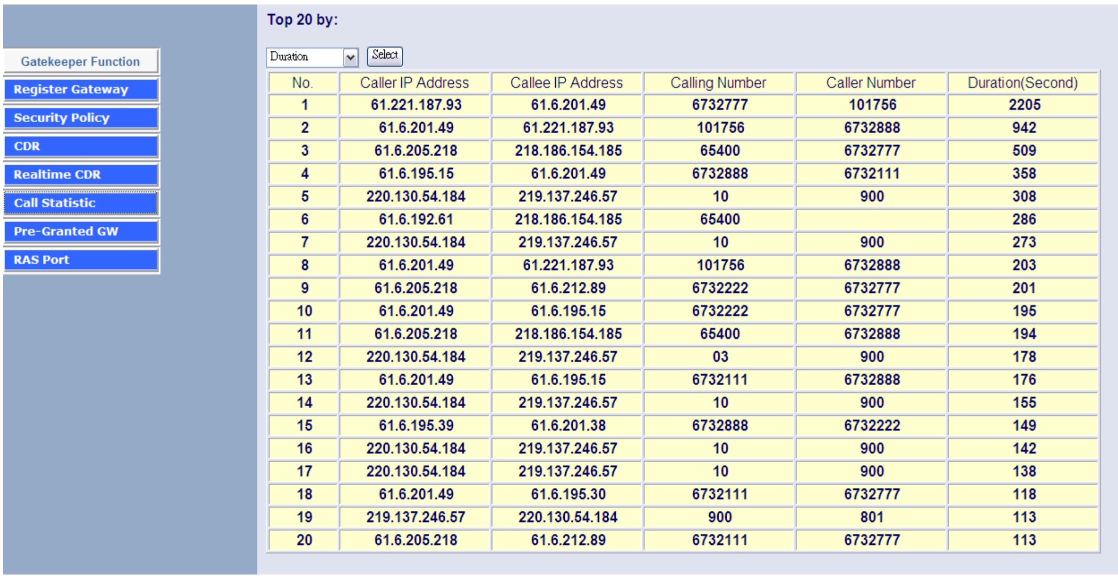
(These 3 authentications, you should 1 or all to security your gatekeeper registered. Max 50 Policy)

4. View CDR and Real time CDR, you can select CDR (Real time) option.



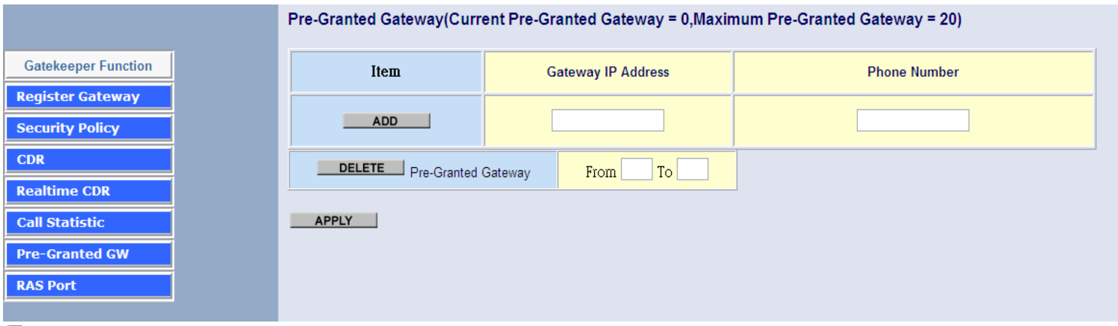
No.	Caller IP Address	Callee IP Address	Calling Number	Caller Number	Duration(Second)	Call Date
1.	202.132.196.190	218.186.154.185	65400	0000	13	2006/4/8 0:16:41
2.	202.132.196.190	61.6.192.17	6732288	0000	14	2006/4/8 0:14:37
3.	202.132.196.190	83.70.203.184	353400	0000	11	2006/4/8 0:13:40
4.	61.6.205.218	61.6.212.27	6732222	6732777	68	2006/1/25 16:25:45
5.	61.6.205.218	218.186.154.185	65400	6732777	82	2006/1/25 15:17:16
6.	61.6.205.218	218.186.154.185	65400	6732777	509	2006/1/25 15:12:15
7.	61.6.205.218	218.186.154.185	65400	6732777	7	2006/1/25 13:4:5
8.	61.6.205.218	61.6.212.27	6732222	6732777	32	2006/1/25 9:30:30
9.	61.6.205.218	61.6.212.27	6732333	6732777	1	2006/1/25 9:29:47
10.	61.6.192.61	218.186.154.185	65400		49	2006/1/24 21:38:57

5. Call statistic, you can view what is the top when you select this option.



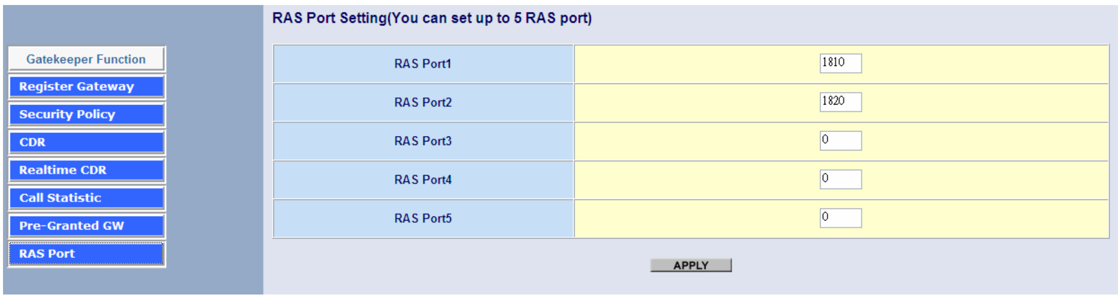
No.	Caller IP Address	Callee IP Address	Calling Number	Caller Number	Duration(Second)
1	61.221.187.93	61.6.201.49	6732777	101756	2205
2	61.6.201.49	61.221.187.93	101756	6732888	942
3	61.6.205.218	218.186.154.185	65400	6732777	509
4	61.6.195.15	61.6.201.49	6732888	6732111	358
5	220.130.54.184	219.137.246.57	10	900	308
6	61.6.192.61	218.186.154.185	65400		286
7	220.130.54.184	219.137.246.57	10	900	273
8	61.6.201.49	61.221.187.93	101756	6732888	203
9	61.6.205.218	61.6.212.89	6732222	6732777	201
10	61.6.201.49	61.6.195.15	6732222	6732777	195
11	61.6.205.218	218.186.154.185	65400	6732888	194
12	220.130.54.184	219.137.246.57	03	900	178
13	61.6.201.49	61.6.195.15	6732111	6732888	176
14	220.130.54.184	219.137.246.57	10	900	155
15	61.6.195.39	61.6.201.38	6732888	6732222	149
16	220.130.54.184	219.137.246.57	10	900	142
17	220.130.54.184	219.137.246.57	10	900	138
18	61.6.201.49	61.6.195.30	6732111	6732777	118
19	219.137.246.57	220.130.54.184	900	801	113
20	61.6.205.218	61.6.212.89	6732111	6732777	113

6. You can let GK to call other gateway without registered, setting in Pre-granted GW option. MAX setting 20 gateway.



- **Gateway IP Address:** Input you want to call without registered Gateway IP address
- **Phone number:** Input that gateway call phone number.

7. You can change your gatekeeper RAS port to gateway registered. 5 RAS port can be setting.



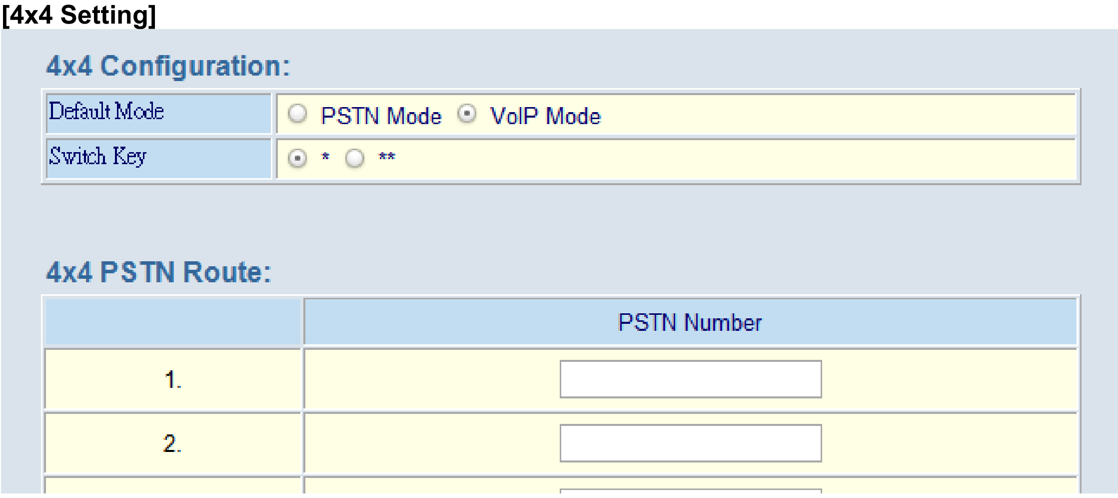
RSA Port: Setting H.323 RSA signal port , you can setting 5 different RSA port to Registered.

4.5 Call Manager

Call Manager is used to determine which mode your dial calls will go through by two methods to achieve, PSTN or VoIP. You can manually press Switch Key to switch the mode or simply set up the dial plan to let the rules judge the mode automatically.

The setting of Call Manager is under [VoIP Setup] -> [Advanced Setting] -> [4x4 Setting]

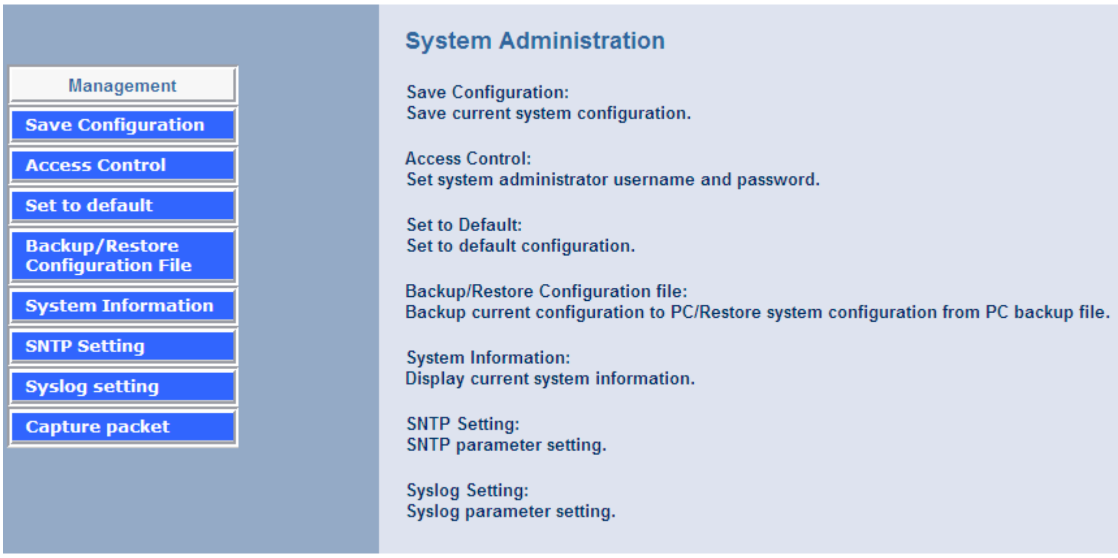
[4x4 Setting]



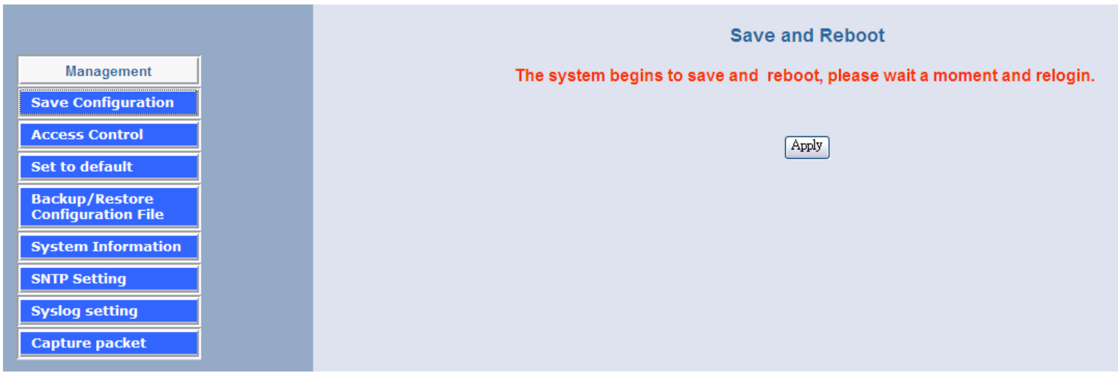
Item	Description
Default Mode	The mode which determine which way your calls will go directly
Switch Key	The method to switch two modes. To switch the mode quickly by pressing the star (*) button once or the star (*) button twice before you dial any numbers. Once you have finished the dial, you will hear a beep sound that notifies you the action is done.
PSTN Number:	The numbers which are filled in the form will go through the PSTN line unconditionally. For examples: Emergent calls, like 911 Zone Numbers, like 02x (the phone numbers start with 02)

4.6 System Administrator

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



4.6.1 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!)

4.6.2 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.

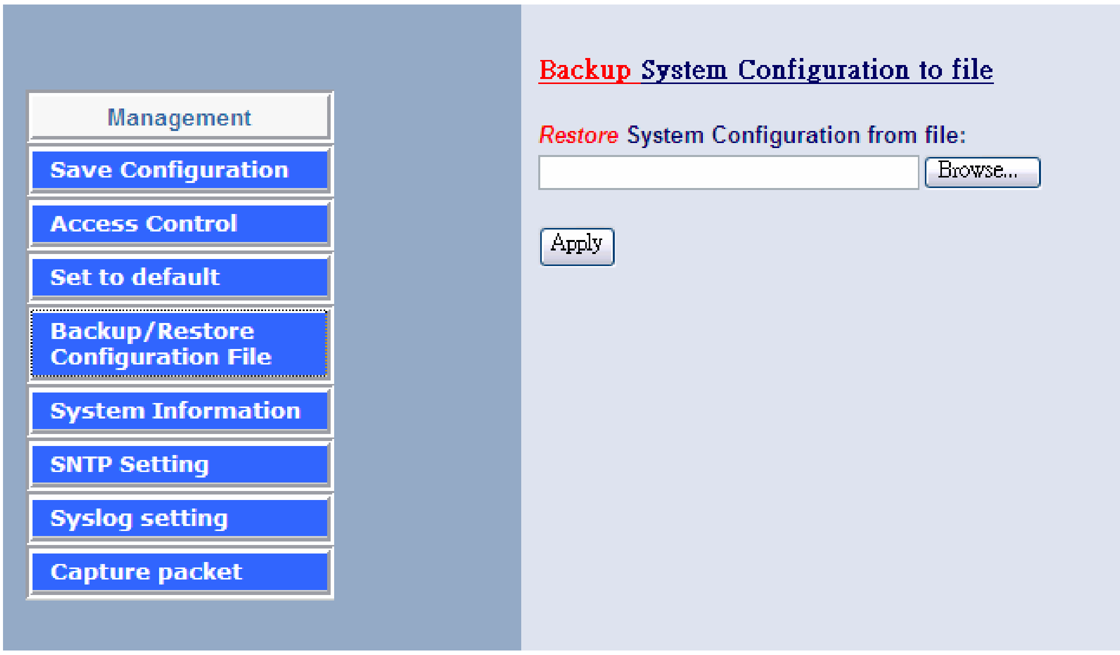
- Click **Access Control Setup**, the following screen will open.

(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)

4.6.3 Set To Default Configuration

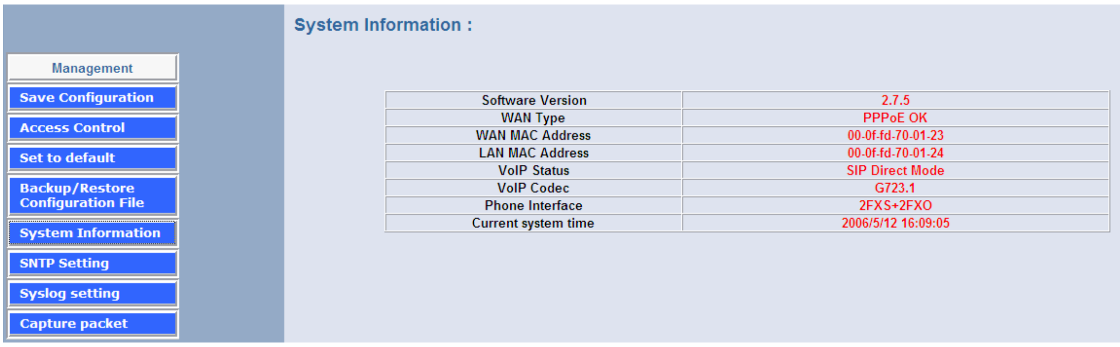
- If you want to reboot the router using **factory default configuration**, click "**Apply**" then reset the router' s settings to default values.

4.6.4 Backup/Restore Configuration to a File



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

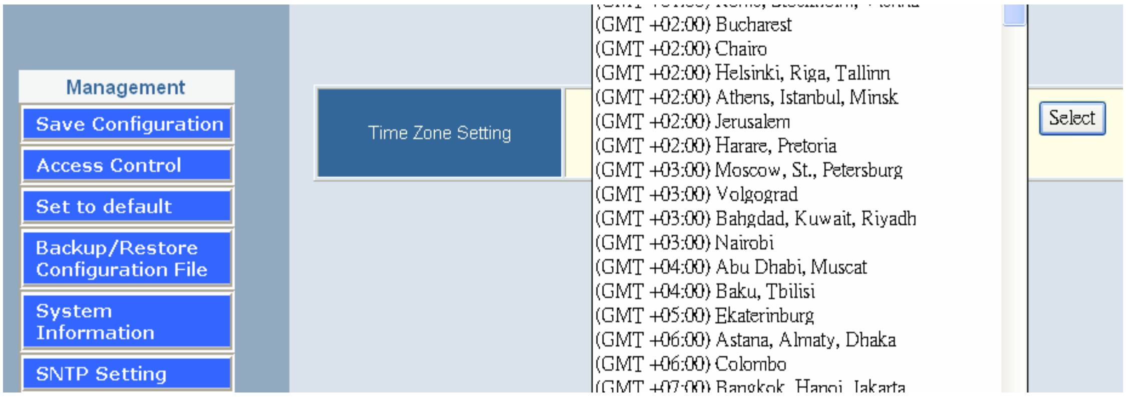
4.6.5 System Information Display Function



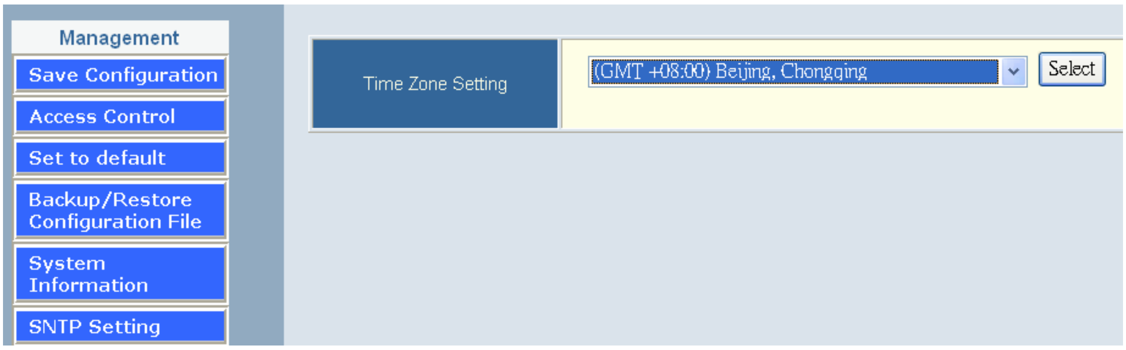
- Click System Information Display to open the Online Status page. In the example, on the following page, both PPPoE connection is up on the WAN interface, H323 Status, MAC address, Register Status, etc.…

4.6.6 SNTP Setting Function

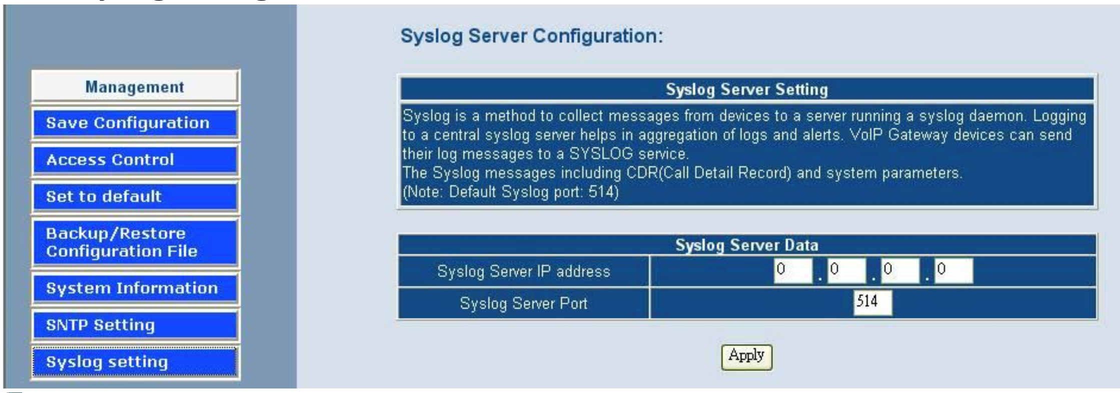
Click **SNTP Setting** to open the Online Status page. In the example, on the following page, .



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

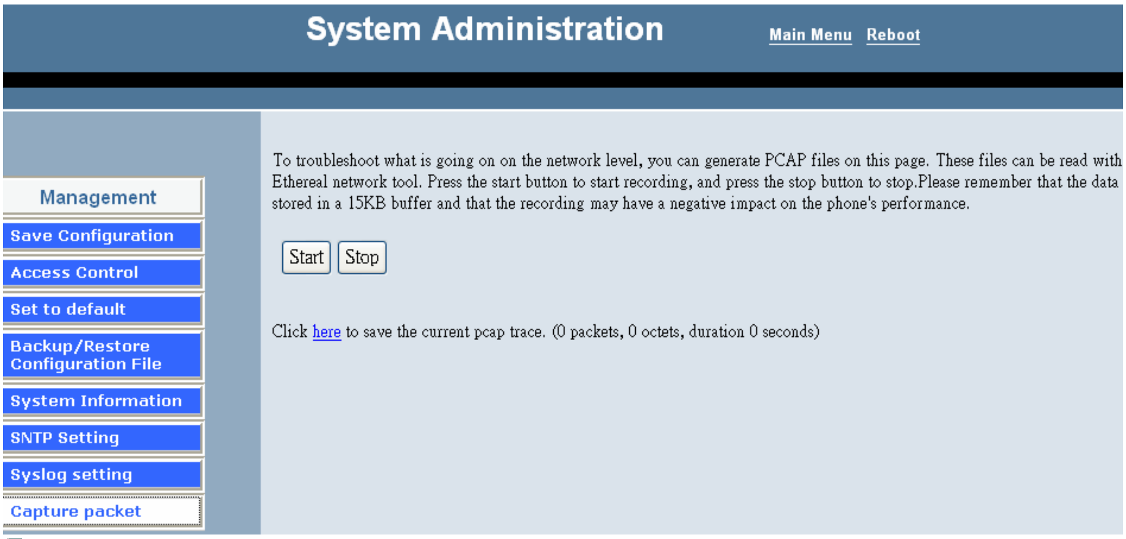


4.6.7 Syslog Setting Function



- 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!

4.6.7 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.)

4.7 Update firmware(For Gateway & GK & SVR)

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 "S400.275", the first name s400 is mean the gateway module. There are 3 module name use different firmware.S200/S400(GK/SVR) are use the Firmware first name "S400", S800(GK/SVR) use the "S800", SB800/S1600/S2400 use the "S2400". The second name is the firmware version. For example, "s400.275" is mean that firmware use S200/S400 (GK/SVR) module, version is 275.

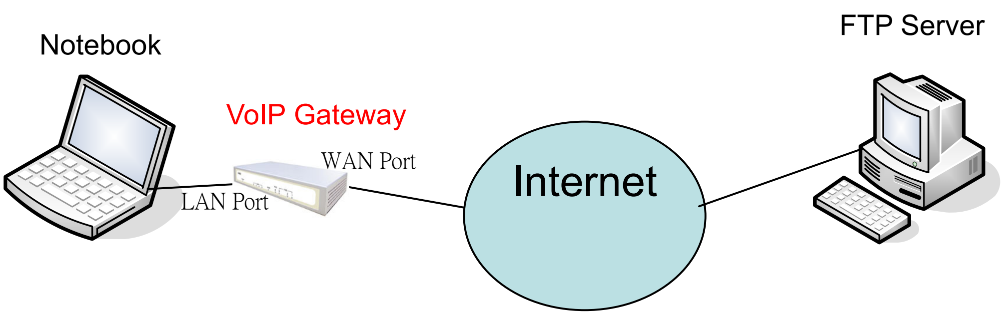
(Gateway update firmware only support use **telnet** via **FTP,** no other else upgrade function.)

FTP upgrade Requirement and Process

1. Environment Requirement

- PC with FTP Server (Server-U software, 3CDaemon,..)
- PC or Notebook witch connected to LAN port of Gateway.
- Put the image (firmware) named "s400.xxx" at the assigned folder in FTP Server.

(for example: "s400.270" 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 10.10.10.1 (Default LAN port IP address) (before version 2.6.5)
->C:> telnet 222.222.222.1 (Default LAN port IP address) (after version 2.6.6)
Please select [4] Upgrade Software

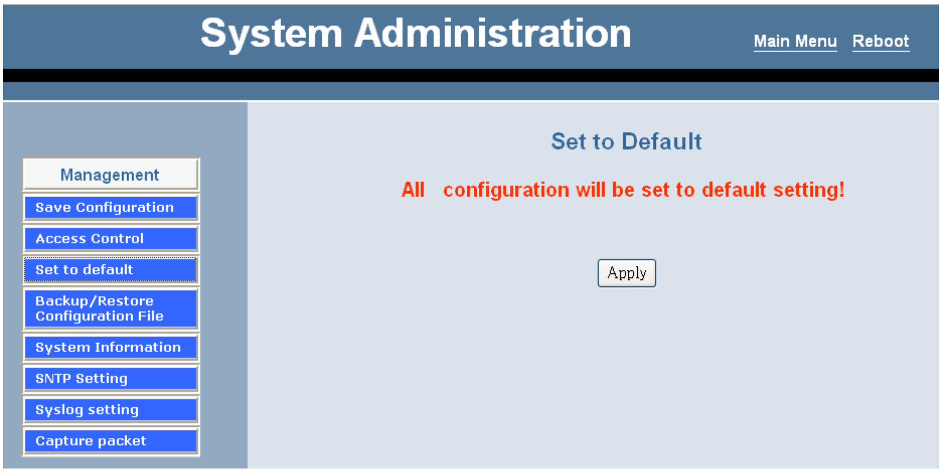
```
            Welcome to 4FXS VoIP Gateway (version 2.7.1)
=====================================================================
                              Main Menu
=====================================================================
WAN Status:Fixed IP (NAT Mode)
VoIP Status:Register OK.(H.323 GK Mode)
=====================================================================
 [1] Advanced Setup.
 [2] System Administration.
 [3] Save Current Configurations.
 [4] Upgrade Software.
 [5] Ping.
 [6] Logout.
 [7] Restart.
 Please Select 1 - 7:4

FTP Server IP = 61.218.109.83

Username : share

Password :
 Image Name = s400.271_
```

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.**



Soundwin Network, Inc. - Headquarters
10F-4, No. 295, Sec. 2, Guangfu Rd.,
Hsinchu City, Taiwan

Tel: +886-3-5733113
Fax: +886-3-5736131
E-mail: sales@soundwin.com
Web Site: http://www.soundwin.com
This manual Write by Tony chi check by gloria

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.

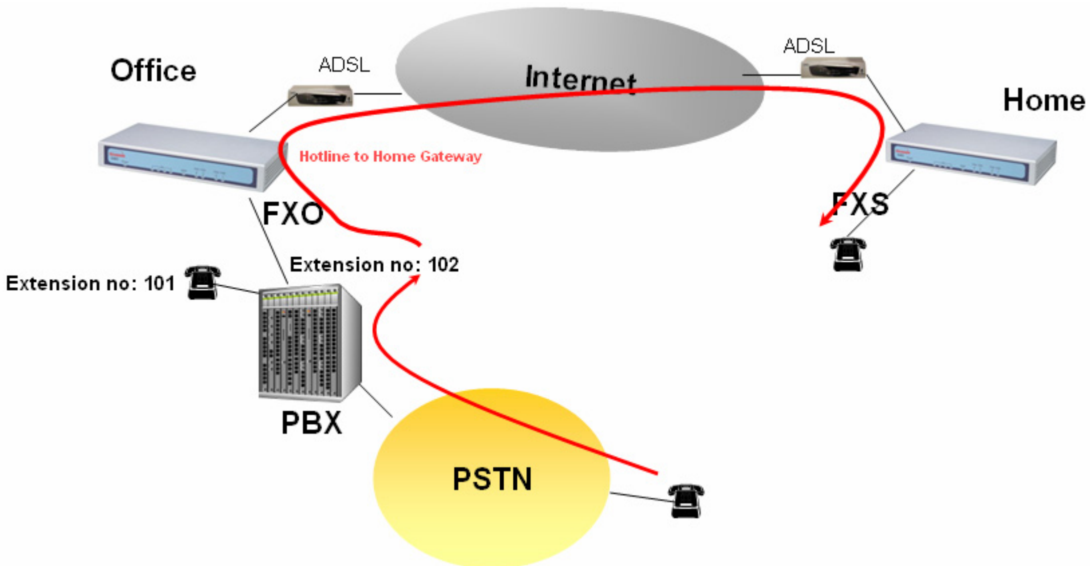
3. What is the default IP address of the router?
A: The default WAN IP address is 192.168.1.1 with subnet mask 255.255.255.0.
The default LAN IP address is 222.222.222.1 with subnet mask 255.255.255.0.

4. Why is it that I can ping to outside hosts, but not access Internet Web sites?
A: Check the DNS server settings on your PC. You should get the DNS servers settings from your ISP. If your PC is running a DHCP client, remove any DNS IP address setting. As the router will assign the DNS settings to the DHCP-client-enabled PC.

5. What is the maximum number of IP addresses that the DHCP server of the gateway can assign to local PCs?
A: The built-in DHCP server can support 253 IP addresses for local network usage.

6. 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/Lan port become the factory default(192.168.1.1/222.222.222.1). 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.

7. What is the Flash?



1. PSTN Call from PSTN to Office PBX and dial the extension 102 go to Gateway
2. Call to Gateway of Home by Hotline.
3. Home user needs call transfer to extension number 101.

4. Dial Flash and Gateway FXS detect and generate the Flash to PBX in Office.
Flash is mean on-hook and off-hook fast switch, on-hook and off-hook duration. It usually use on PBX system, in place of transfer function key.

8. 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.

9. 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.

10. 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.

11. I can't use web Interface to setting gateway. How can I do?
A: Please check you PC connect the Gateway Lan port or PC and ATA with the same Subnet. If you PC aren't at the same Subnet, you can't Login the gateway Web interface. Else you let your gateway on Public Internet(Public IP address).when you Lan and Wan can't login always, try to set factory default. (Push RST button until 5 second), and try again

12. 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.

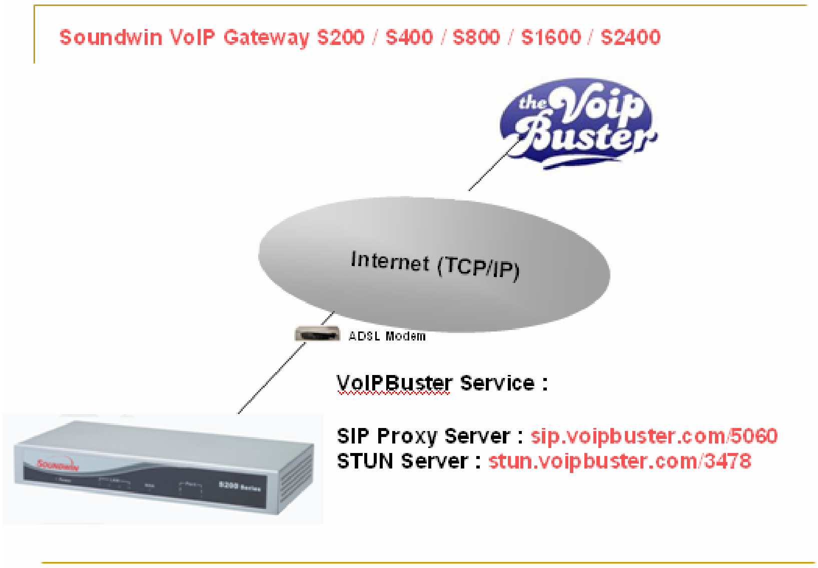
13. Can I use re-invite function when server is asterisk?
A: No, our gateway not support this function complete, the new firmware will support re-invite.

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

15. Why I can't use FAX?
A: The VoIP FAX has 2 mode, one is T.30, another is T.38. T.30 use voice date to FAX, so usually T.30 FAX by VoIP is work. There are some problem with T.38. Because T.38 is use protocol and data packet to sand FAX, different platform and device maybe have different change, that made the T.38 filed. If you can use T.38 to FAX, Please try to use T.30.

B SIP Setting VoIPBuster

VoIPBuster Service Using Soundwin VoIP Gateway



The Soundwin S200/S400/S800/SB800/S1600/S2400 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.

Basic Configuration

VoIP Protocol Setting SIP Select

Port Number / Password Setting(MAX 20 digit) :

Number	Reg	Account	Password	Register Stat
samchen0809	☑	samchen0809	••••••••	Success
soundwin035733113	☑	soundwin035733113	••••••••	Success
soundwin035733114	☑	soundwin035733114	••••••••	Success
soundwin035733115	☑	soundwin035733115	••••••••	Success

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 soundwin company phone number is +886-35733113, the dial number is 0088635733113

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

FREE CALLS! (LANDLINES ONLY)

Andorra	Georgia	New Zealand
Australia	Greece	Norway
Austria	Hong Kong	Panama
Belgium	Iceland	Peru
Bulgaria	Ireland	Portugal
Canada	Italy	Puerto Rico
Chile	Japan	Singapore
Colombia	Latvia	Slovenia
Croatia	Liechtenstein	South Korea
Cyprus	Luxembourg	Spain
Denmark	Malaysia	Taiwan
Estonia	Monaco	Thailand
Finland	Mongolia	Venezuela
France	Netherlands	

100% Free, no call setup! Click here for more info. For all other rates, click here

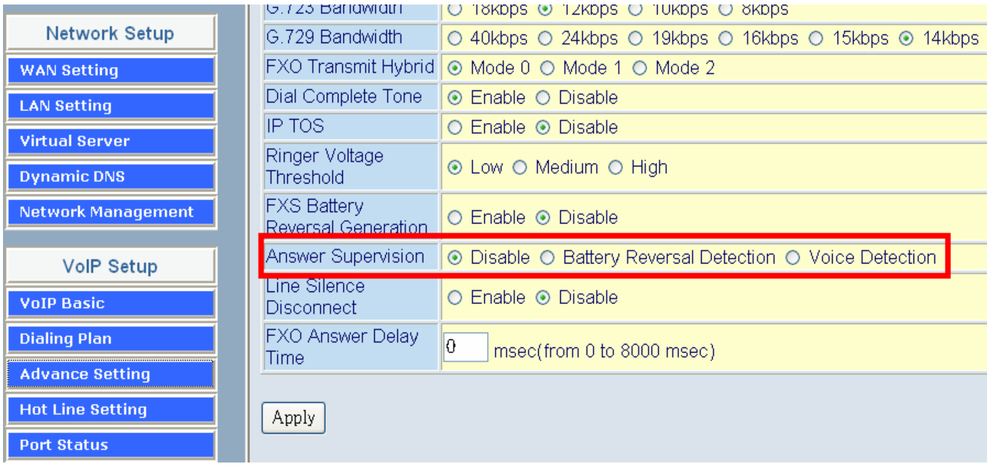
C Answer supervision

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

Gateway provides 2 Types of Answer Supervision:

1. Loop-Start Reverse Battery, Reverse battery (also called Polarity Reverse) is when the PSTN provider reverses the polarity of the battery voltage, for both answer supervision and disconnect supervision.

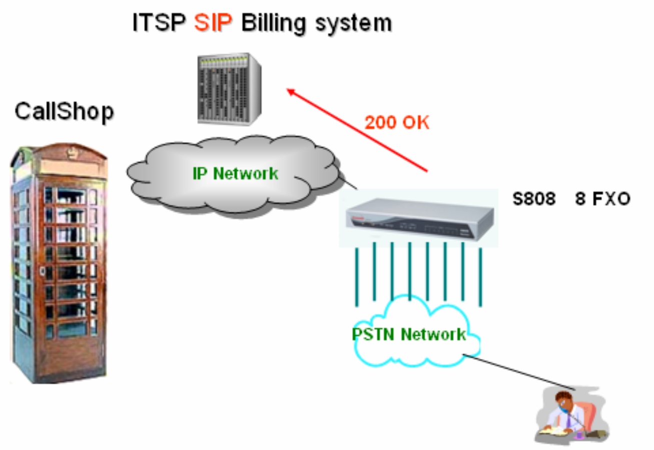
2. Voice Detection-based answer supervision is a feature where the Gateway can be configured to "listen" on the line for different tones and voice. The Gateway sends a "connect" signals out or "disconnect" signaling using internet.



SIP Application

Case 1: Loop Start Reverse Battery : PSTN Line was set Polarity Reverse:

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



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

Case 2: Voice Detection based on answer supervision:

PSTN Line was not support Polarity Reverse: The gateway can send the 200 OK SIP signals to Billing System of ITSP, after the user pick up the Phone and detect the voice.

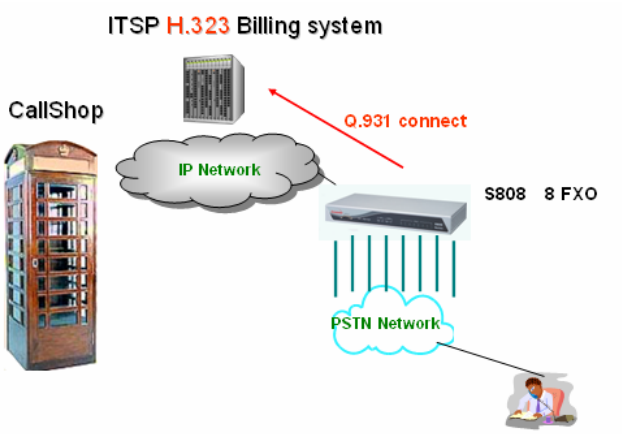
The gateway can send the 200 BYE SIP signals to Billing System of ITSP, after the user hang up the Phone and detect the hang up voice.

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

H.323 Application

Case 1: Loop Start Reverse Battery PSTN Line was set Polarity Reverse:

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



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

Case 2: Voice Detection based on answer supervision

PSTN Line was not support Polarity Reverse:

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

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

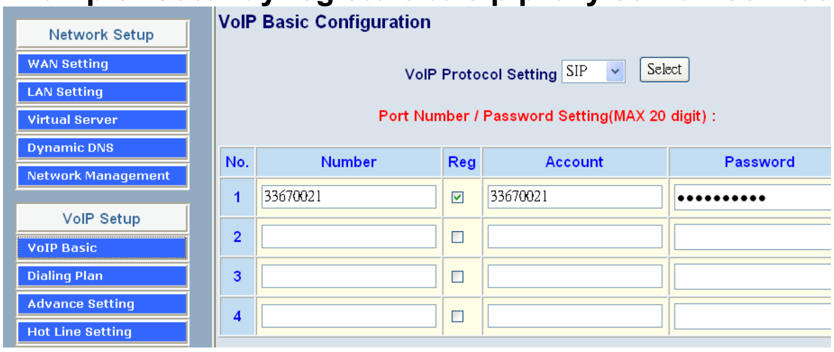
This type of answer supervision is not 100% accurate. Any voice frequency is detected as connect, including any intercept or recorded messages.

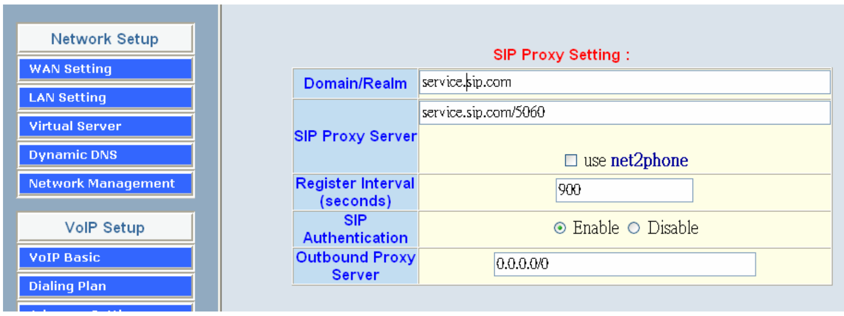
D Sip Speeds call

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

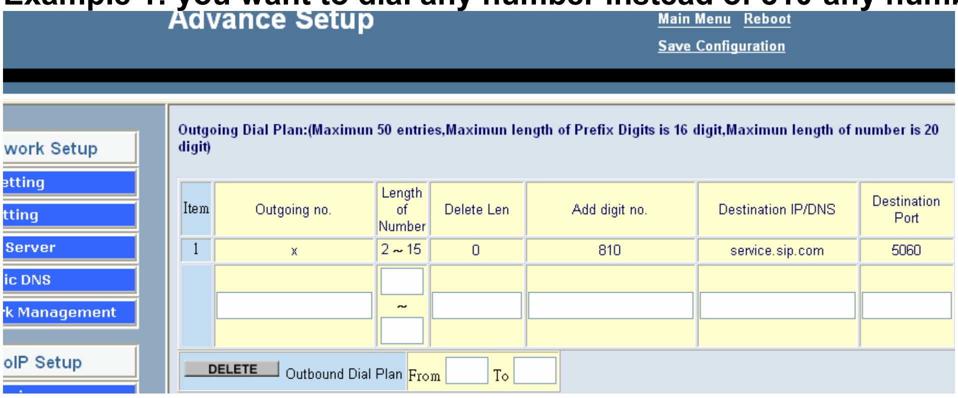
Example: Gateway registers to sip proxy server: service.sip.com





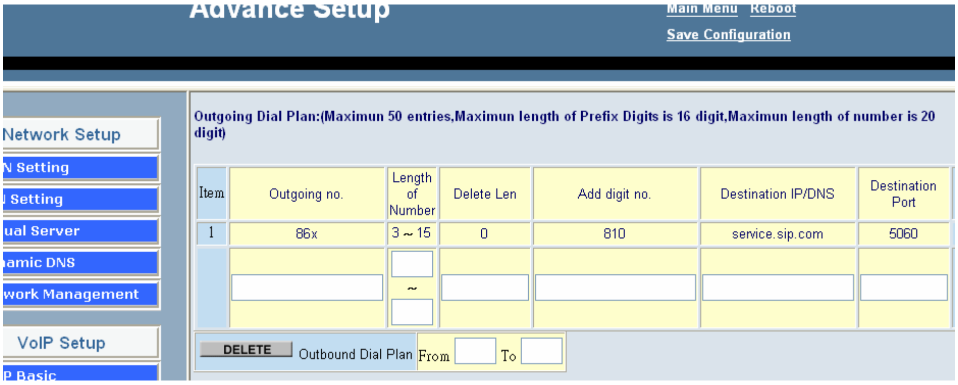
What's even better is that you can customize and manage your speed dial phone numbers in Dial Plan Setting on your gateway! Dial Plan allows you to set up to speed dial numbers that can be called with the fewer numbers.

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



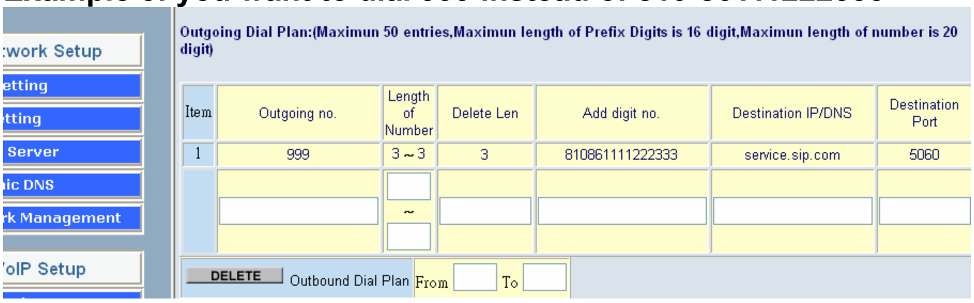
The destination IP address is the domain name of sip proxy server

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 server

E Interoperability List

- **Gatekeeper**
  - GnuGK openH323
  - Radvision ProLab GateKeeper Simulator, Version 1.0, October 2001
  - Clarent Gatekeeper
  - MediaDigm-SureKeepe
  - Lucent -i Merge GK

- **SIP Proxy Server**
  - Vovida SIP Proxy Server
  - SER (SIP Express Router)
  - Party SIP ServerV0.5.0
  - Clarent SIP server
  - Asterisk 0.5.0
  - SXi SIP server

- **Gateway:**
  - Welltek 2~8 port GW
  - ACCEL 2~8 port GW
  - Cisco 5300 and Cisco 5350 Trunk GW
  - Cisco ATA -186
  - MOSA 4 port
  - CISCO ATA-186
  - Quintum Tenor A400/A800, AS/AX Series
  - Antek 2500 series and vsp5004 series
  - Audiocodes MP-104 FXS
  - Clarent CPG-101 , CPG-22102Sc
  - D-Link 4-port gateway
  - BOSaNOVA Analog Gateways

- **Trunk Gateway:**
  - Cisco AS5300, 5350
  - Clarent BHG2500
  - AudioCodes IPM-260 Board

- **IP Phone:**
  - ACT LAN phone
  - TECOM IP PHONE
  - BCM IP Phone
  - Cisco 7900 IP Phone
  - UMEC video phone

F RJ21 (Telco 50) Cable and Patch Panel Install

1. General Description

Depend on customer's requirement, there are two kinds of accessories of FXS/FXO wiring for SB800 / S1600 / S2400 series: RJ21 (Telco 50) cable only and RJ21 cable with patch panel. The RJ21 cable only is suitable for customers who have their own MDF (Main Distribution Frame), and the cable length is about 2.7m. The RJ21 cable with patch panel is suitable for customers who need RJ11 wiring to phone or PSTN directly, and the cable length is about 1.6m.

2. RJ21 cable wiring

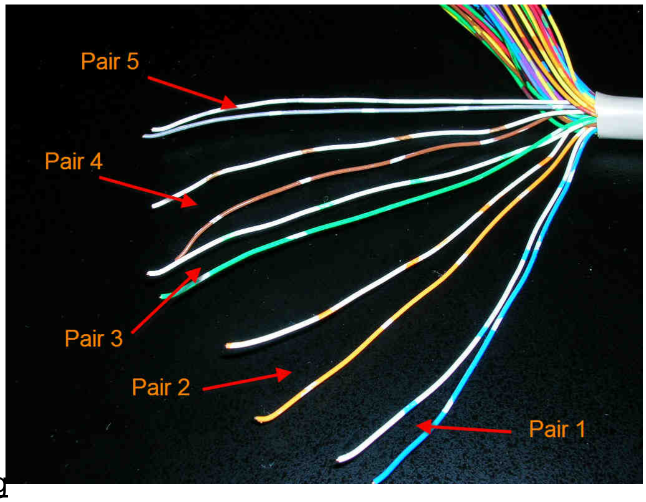
The RJ21 cable consists of 25 pairs of wires for FXS/FXO wiring (S2400 series use 24 pairs). Each wire is colored uniquely to be identified easily. Basically, the color codes are composed of 2 group of colors and the color code for each wire and pair is as the following:

Color Group 1	Blue	Orange	Green	Brown	Gray
Color Group 2	White	Red	Black	Yellow	Purple

Pair 1		Pair 2		Pair 3		Pair 4		Pair 5	
Wire 1	Wire 2	Wire 3	Wire 4	Wire 5	Wire 6	Wire 7	Wire 8	Wire 9	Wire 10
Blue-White	White-Blue	Orange-White	White-Orange	Green-White	White-Green	Brown-White	White-Brown	Gray-White	White-Gray
Pair 6		Pair 7		Pair 8		Pair 9		Pair 10	
Wire 11	Wire 12	Wire 13	Wire 14	Wire 15	Wire 16	Wire 17	Wire 18	Wire 19	Wire 20
Blue-Red	Red-Blue	Orange-Red	Red-Orange	Green-Red	Red-Green	Brown-Red	Red-Brown	Gray-Red	Red-Gray
Pair 11		Pair 12		Pair 13		Pair 14		Pair 15	
Wire 21	Wire 22	Wire 23	Wire 24	Wire 25	Wire 26	Wire 27	Wire 28	Wire 29	Wire 30
Blue-Black	Black -Blue	Orange-Black	Black -Orange	Green-Black	Black-Green	Brown-Black	Black-Brown	Gray-Black	Black-Gray
Pair 16		Pair 17		Pair 18		Pair 19		Pair 20	
Wire 31	Wire 32	Wire 33	Wire 34	Wire 35	Wire 36	Wire 37	Wire 38	Wire 39	Wire 40
Blue-Yellow	Yellow - Blue	Orange-Yellow	Yellow - Orange	Green-Yellow	Yellow - Green	Brown-Yellow	Yellow - Brown	Gray-Yellow	Yellow-Gray

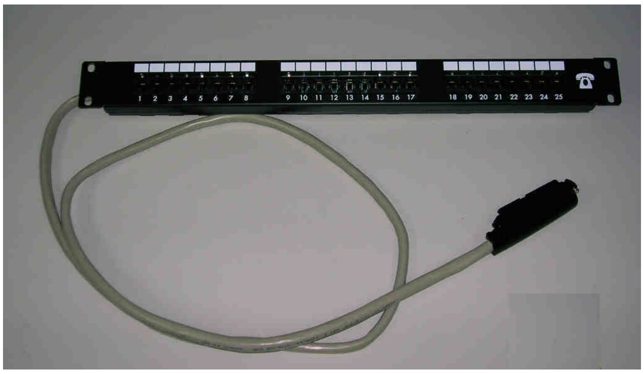
Pair 21		Pair 22		Pair 23		Pair 24		Pair 25	
Wire 41	Wire 42	Wire 43	Wire 44	Wire 45	Wire 46	Wire 47	Wire 48	Wire 49	Wire 50
Blue-Purple	Purple - Blue	Orange-Purple	Purple - Orange	Green-Purple	Purple - Green	Brown-Purple	Purple - Brown	Gray-Purple	Purple-Gray

The picture below shows the pair 1 to pair 5 wiring:

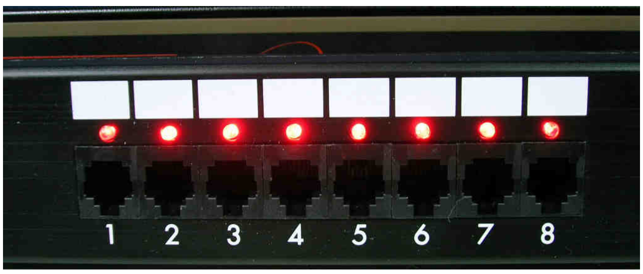


3. RJ21 cable with Patch Panel wiring

There are 25 RJ11 ports on patch panel, and each port is marked from 1 to 25 (S2400 series use port 1 to 24).



Each port has a LED indicator on it. The LED indicator is lighted when this port is for FXS functionality after system booting complete. For FXO port, the LED indicator is not lighted till PSTN plugged with proper polarity (RJ11 pin 4 is positive polarity).



install

Introduce:

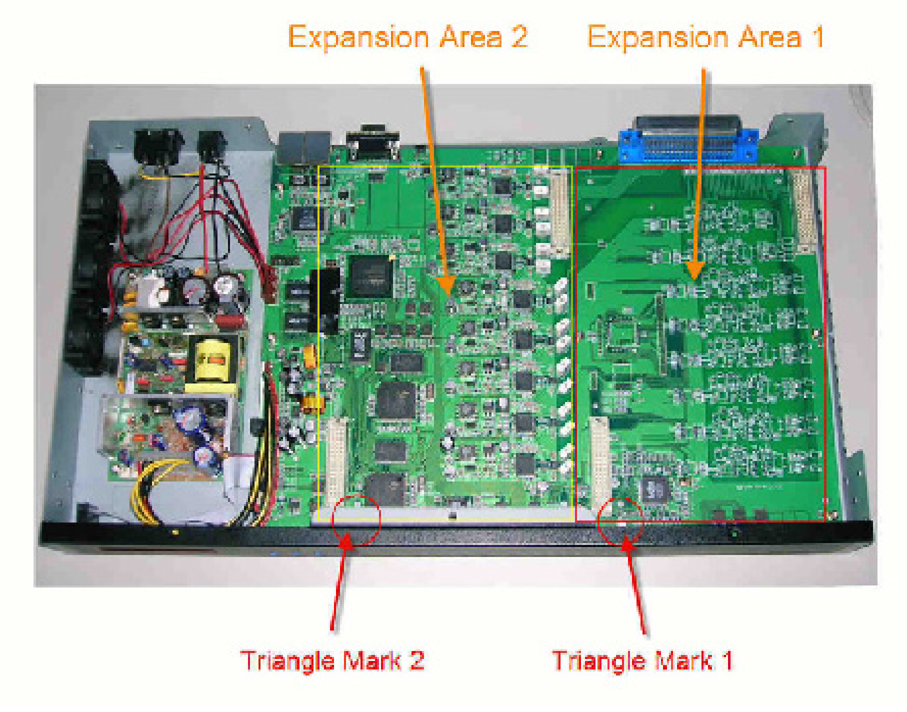
SB800/S1600/S2400 series, according to different module board and motherboard combine to 8/16/24 port high level gateway. The module board and types are as follows:

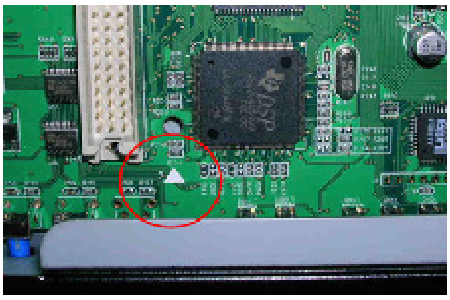
Motherboard	Module_1	Module_2	Type
SB800	M800	--	S1600
SB800	M800	M800	S2400
SB804	M804	--	S1608
SB804	M804	M804	S2412
SB808	M808	--	S1616
SB808	M808	M808	S2424

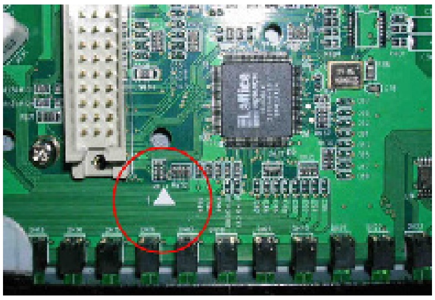
Module group M80x installs:

a.) Open gateway box, let cover on the fixed screws and all take off. Taking down RJ-21 (Telco 50) six angle screws going to, use six angle collets screws. The screwdriver will be more convenient.

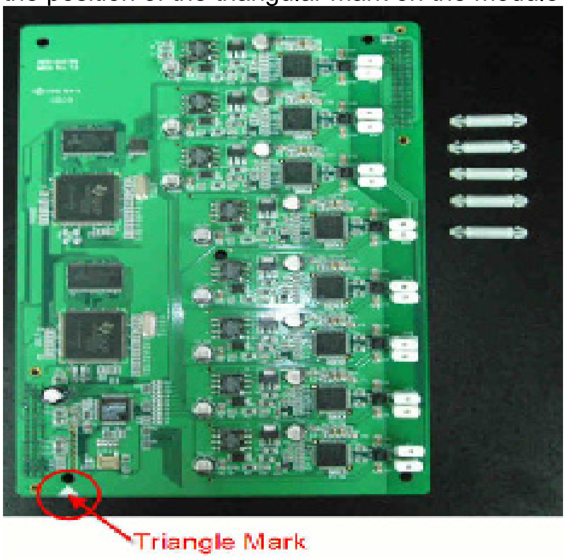
b.) Open the top cover, there is expansion of two groups of DIN41612 model that joins the head on the motherboard. Please pay attention to two groups of triangular marks on motherboards, Position of the mark.



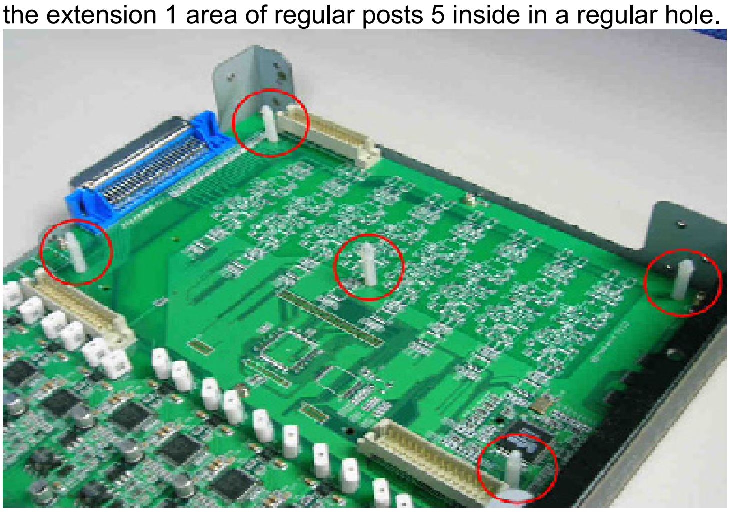




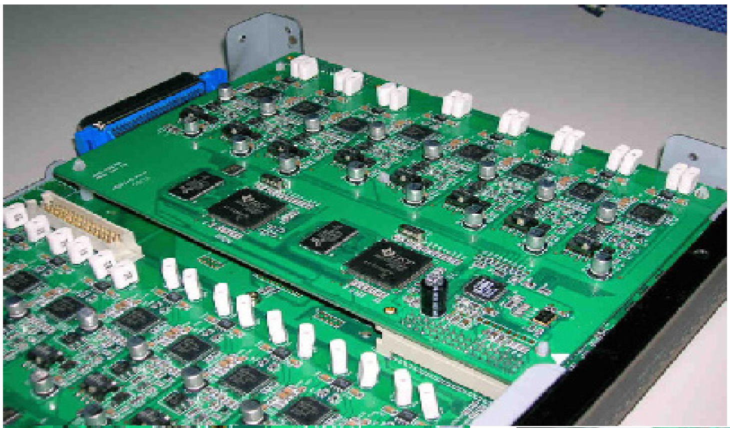
c.) M80x module will enclose 5 regular posts while producing the goods. Please pay attention to the position of the triangular mark on the module

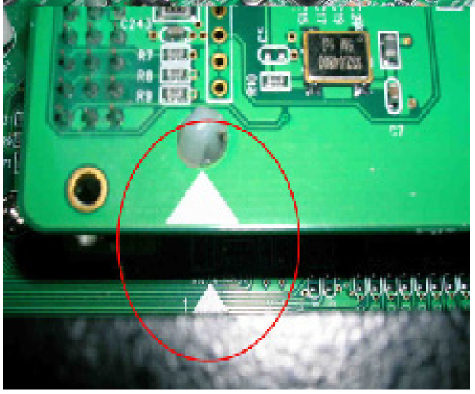


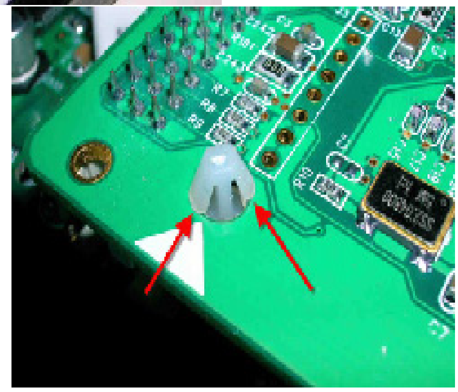
d.)Module 1 (the 9~6th port of FXS/ FXO) Must be put in the extension 1 area. Please insert in the extension 1 area of regular posts 5 inside in a regular hole.



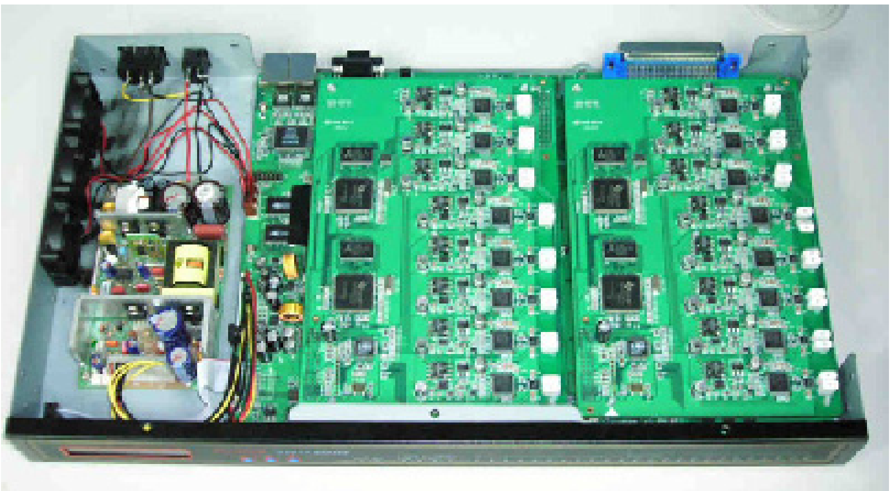
e.)Put and fix the module board on motherboard. Please notice the triangular mark on module and motherboard is on the same position. Please fix DIN41612 connector is regular and good, at the same time the regular post totally opens and blocks the module too.







f.)To the module 2 (17th ~ 24th FXS/FXO port), repeat step d and e are installed by finishing in the expanding area.



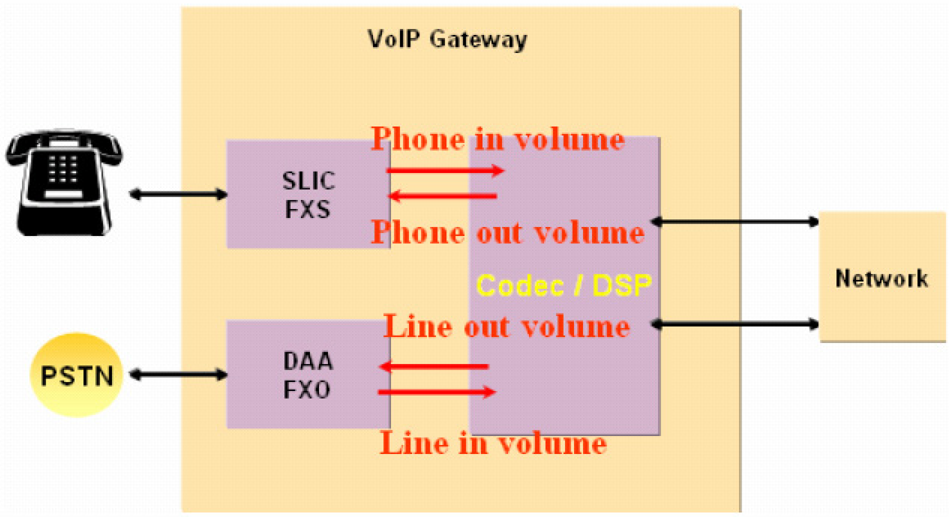
g.) Open power at this moment, type shown in LCD can know module correct to install.

h.)Put back to the top cover, lock all screws of the top cover. Module is installed and finished.

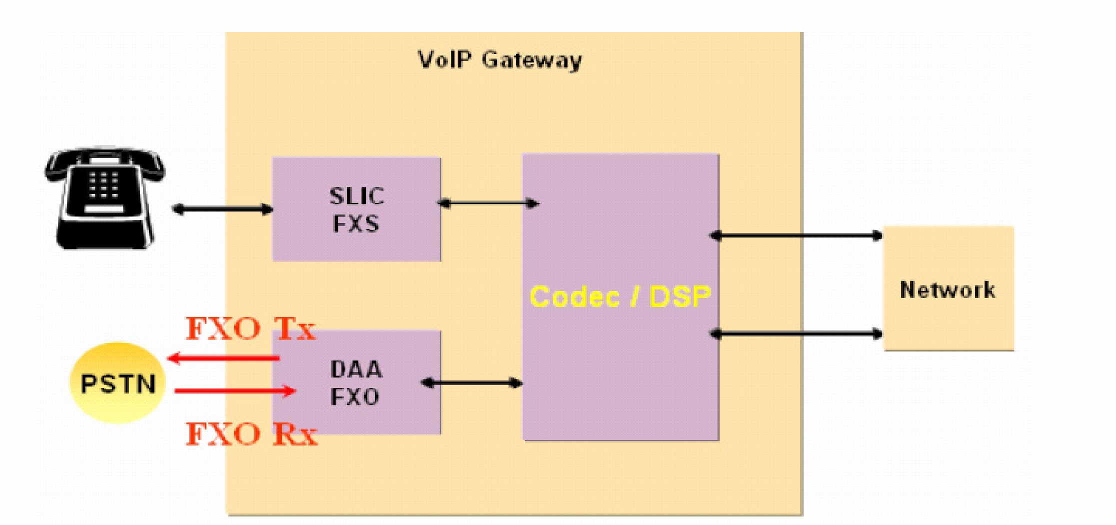
H Gateway value Setting

Phone Volume / Line Volume

Supports user can set DSP volume.



FXO TX /RX Gain



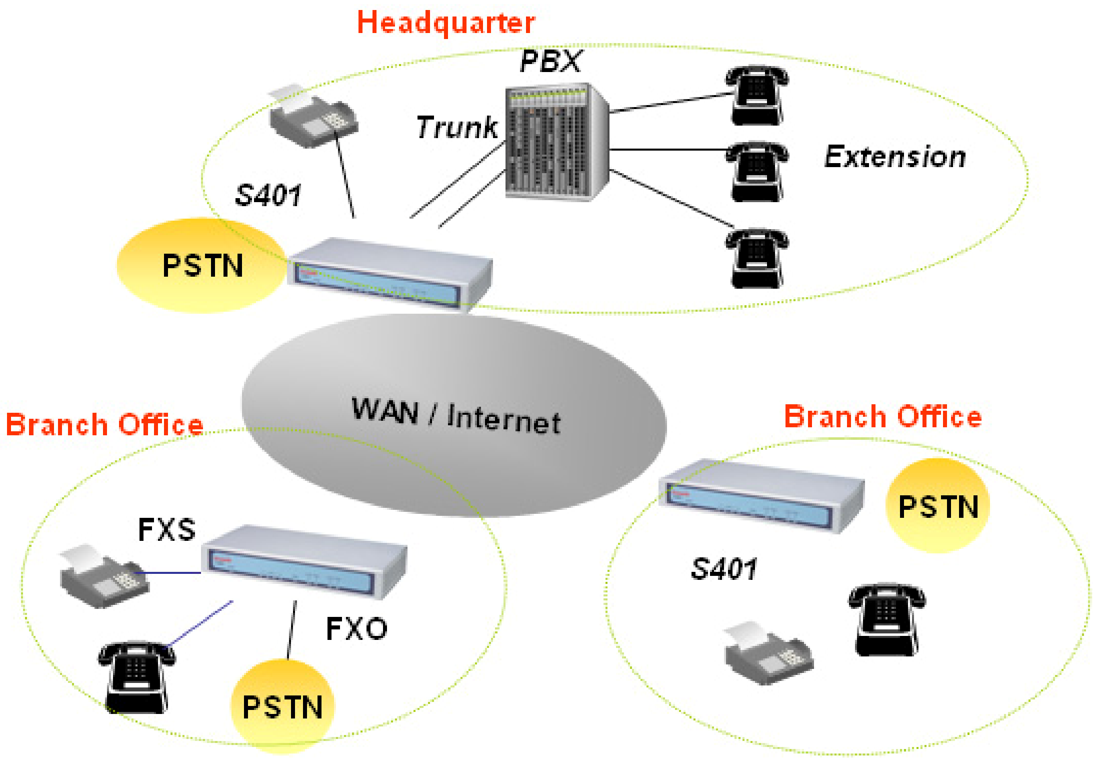
Supports multiple levels of gain and attenuation for the transmit and receive paths of DAA.

1. The FXO tx gain enable gain or attenuation in 1 dB increments for the transmit from DAA site outgoing PSTN site.

2. The FXO rx gain enable gain or attenuation in 1 dB increments for the receive from PSTN site incoming DAA site.
Note: DAA : Direct Access Arrangement.

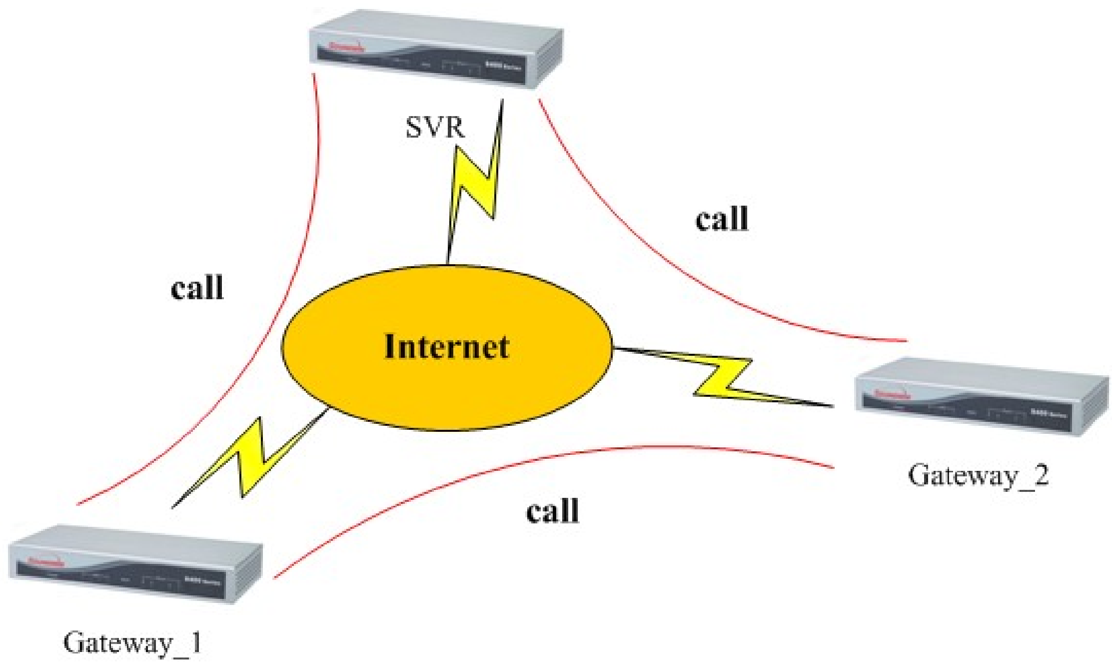
I Scenario Application Samples

Scenario 1:
The following application was use in Multi-Office as an example

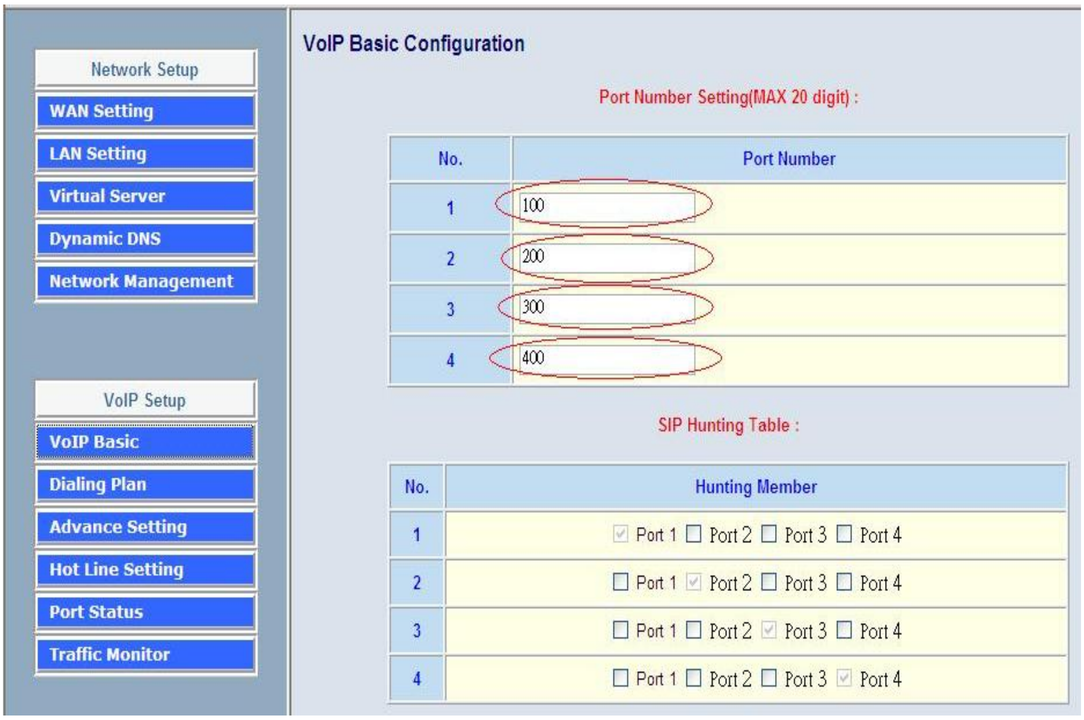


Scenario 2:

SVR and Gateway Setting

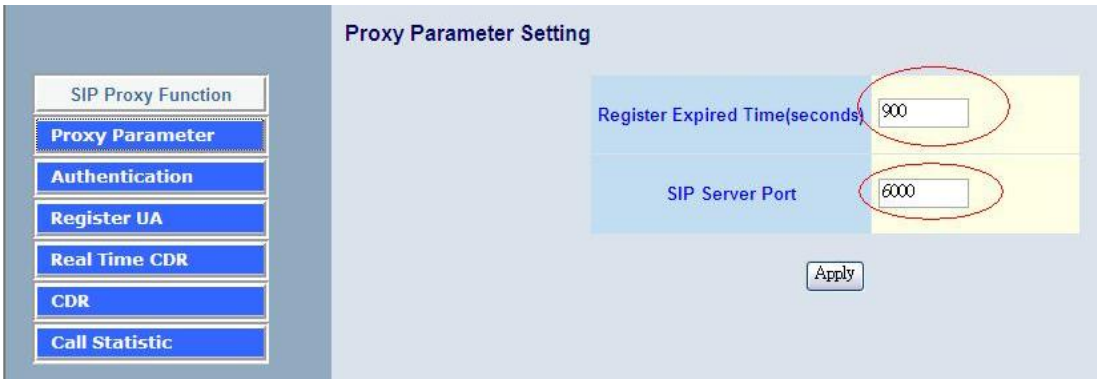


Part 1: SVR Setting

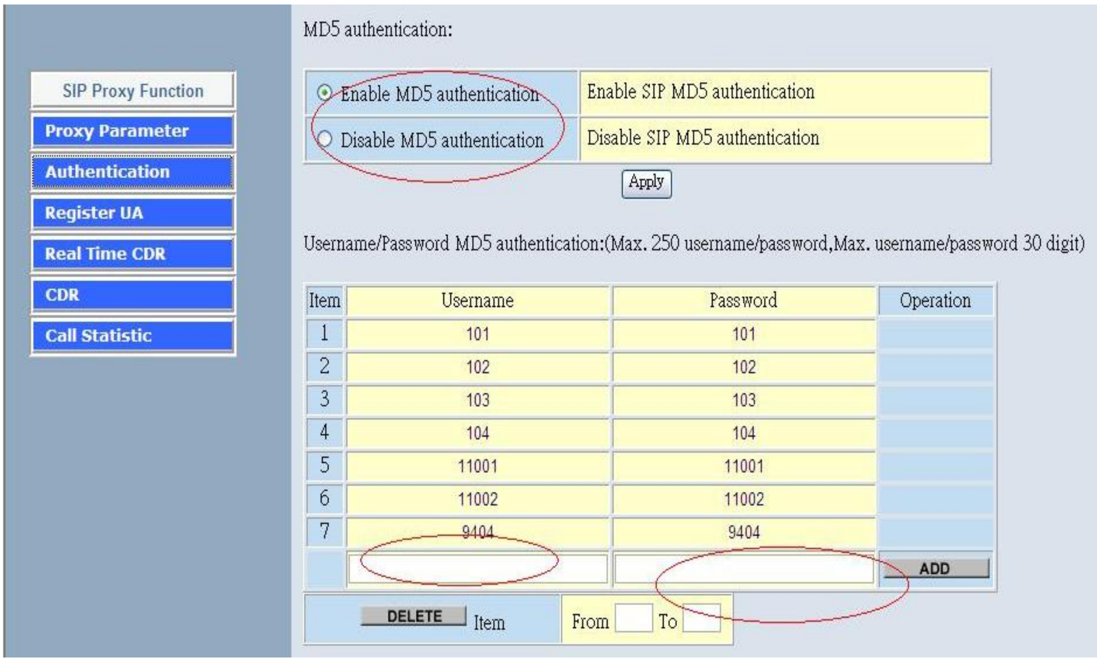


Step_1: Setting SVR Ports Number that is gateway (SVR) port 1~4.If there are other gateway registered on SVR, you can call SVR port number (100~400), or call other gateway registered number. For example, your have a gateway registered SVR number is 1001, you can call 1001 to SVR port 1 (100), or use SVR port 1(100) to call gateway 1001.

Step_2: Select SIP Proxy Server option to setting SIP Proxy Server function. Our SVR is a simple SIP Proxy Server, it support registered and Make call each other.

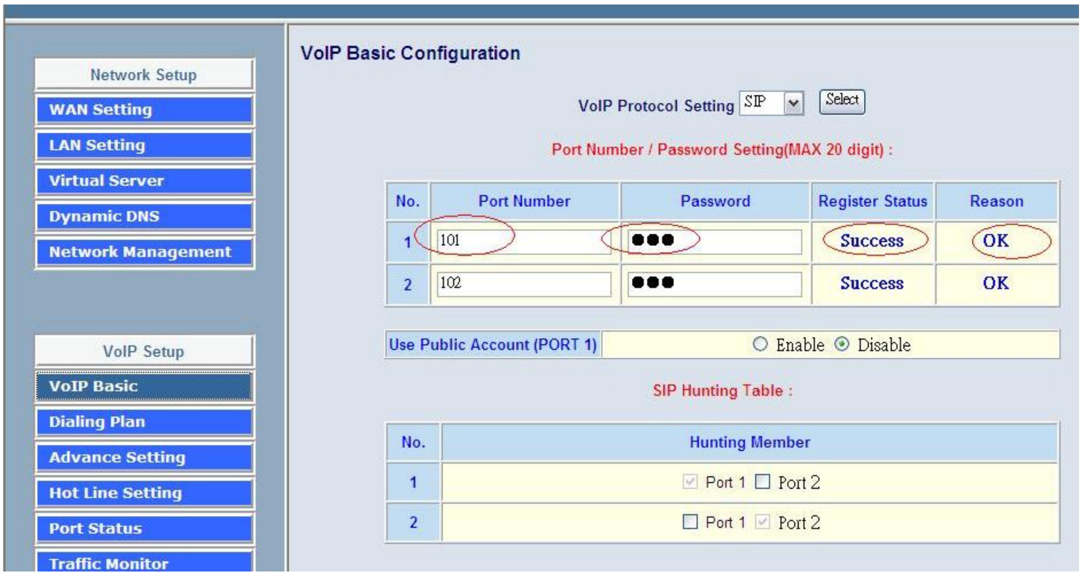


Setp_3: In Proxy Parameter setting, You can set SVR Port and registered time. For example, in this setting Port is 6000 and registered Expired 900 second. Default Proxy port is 5060.

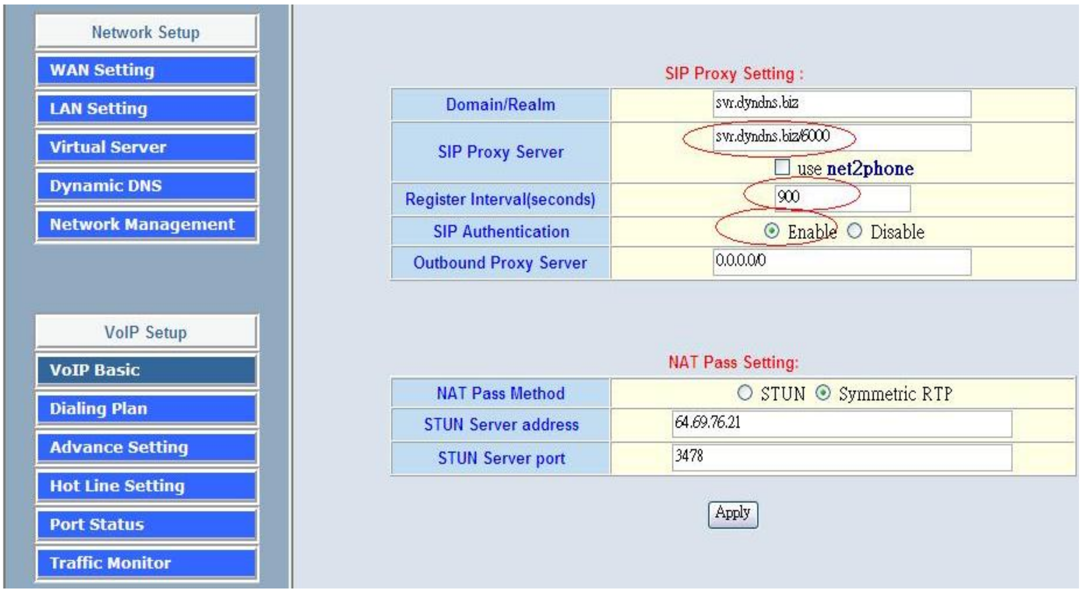


Step_4: Setting Authentication, add account. For example, in this setting, you can Enable/Disable authentication, if you don't use authentication, every gateway can registered SVR. If you use authentication, you must add username/password. Only in username /password table, you can register in SVR.

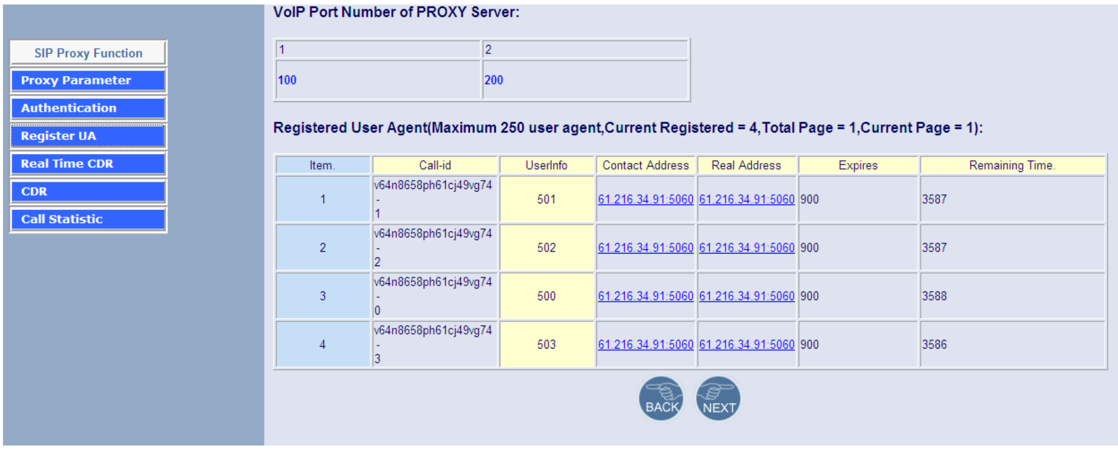
Part 2: Gateway Setting



Step_1: Setting Gateway Port Number, If your SVR use MD5 authentication, you must input Username/Password that on SVR authentication table to register SVR.

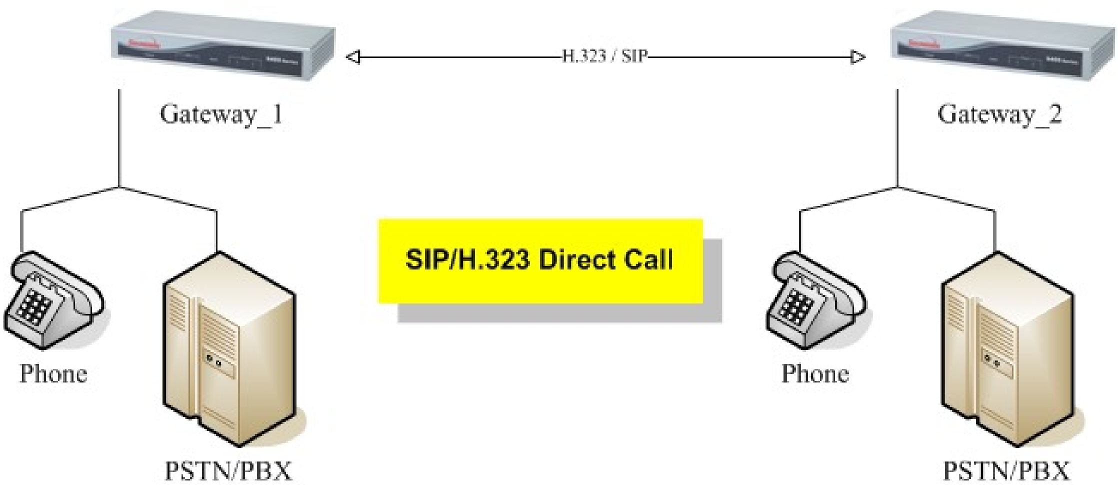


Step_2: Input Server information. For example, red line cycle is must input field. Input SVR IP address or domain name, Register Interval time setting as the same as SVR registered Expired time. If you SVR have MD5 Authentication, you must enable SIP Authentication.



After Setting Gateway and SVR, If your setting is right, you can call each other. you can see all registered UA and number on SVR.

Scenario 3:
SIP or H.323 Direct call Setting



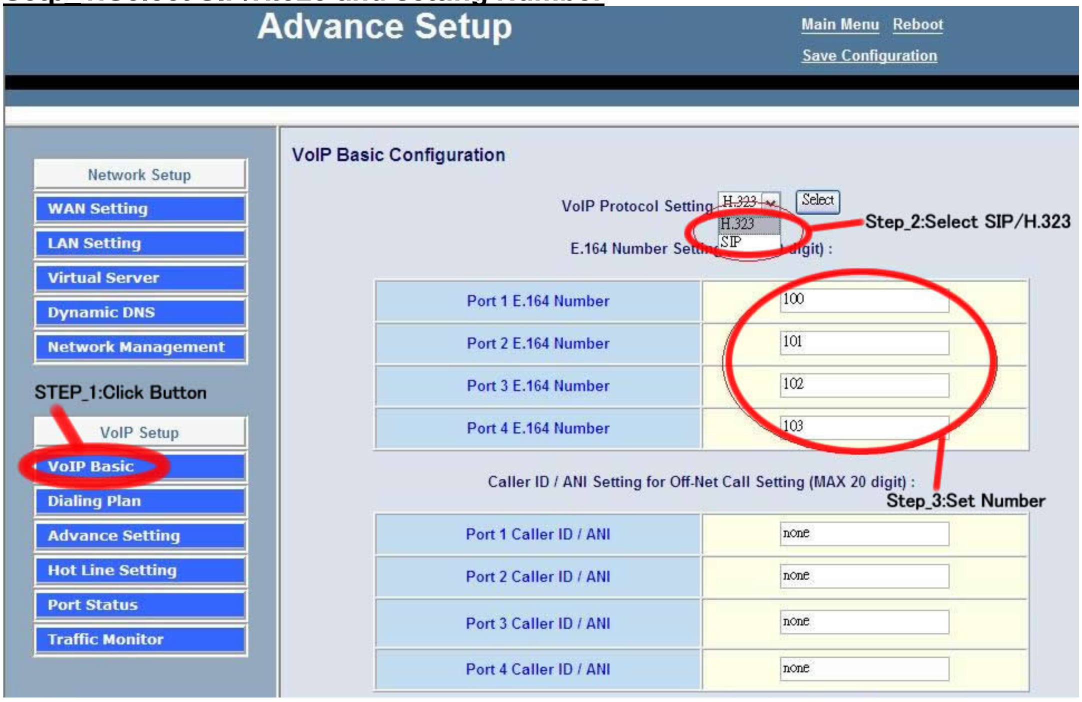
H.323/SIP How to Setting Peer to Peer mode:
-Environment Setting for Demo-
Two Gateway (2/4/8/16/24 port)

Gateway Setting

Information\device	Gateway_1	Gateway_2
IP Address	IP:192.168.1.1	IP:192168.1.2
Port Number	Port_1~Port_4:100~103 (Phone Number)	Port_1~Port_4:200~203 (Phone Number)

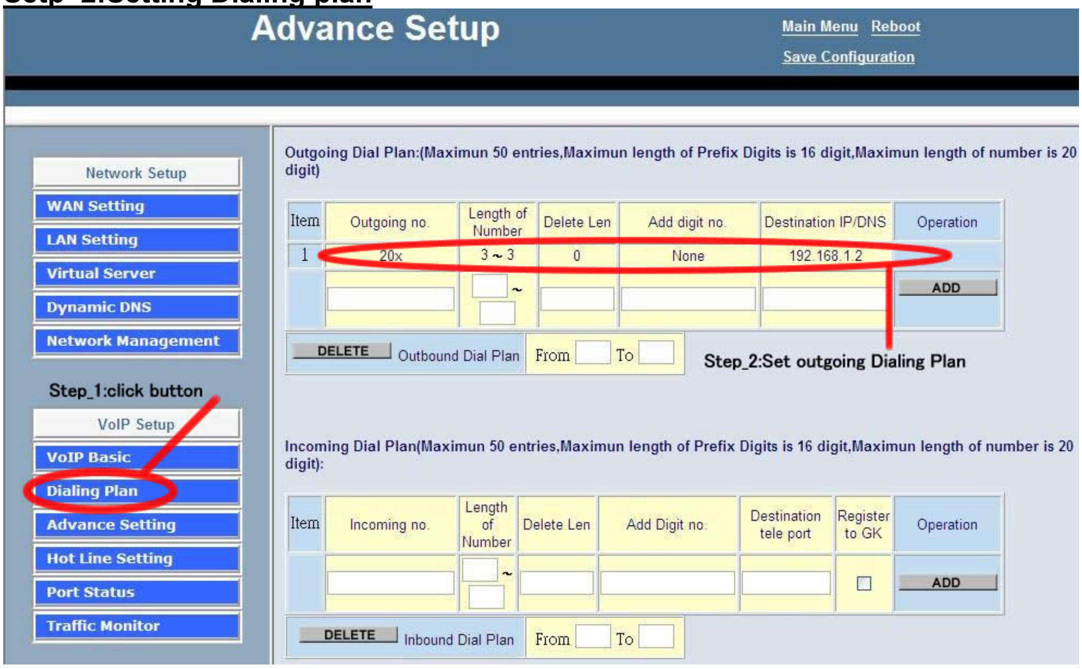
Attentions: If you want to use P2P (Direct mode), please select the same protocol SIP/H.323 both of the gateway. If you want to use different band gateway, please check the other band gateway have support Dialing Plan, we don't premise all of other band gateway can complete support P2P mode.,

Setp 1:Select SIP/H.323 and setting Number



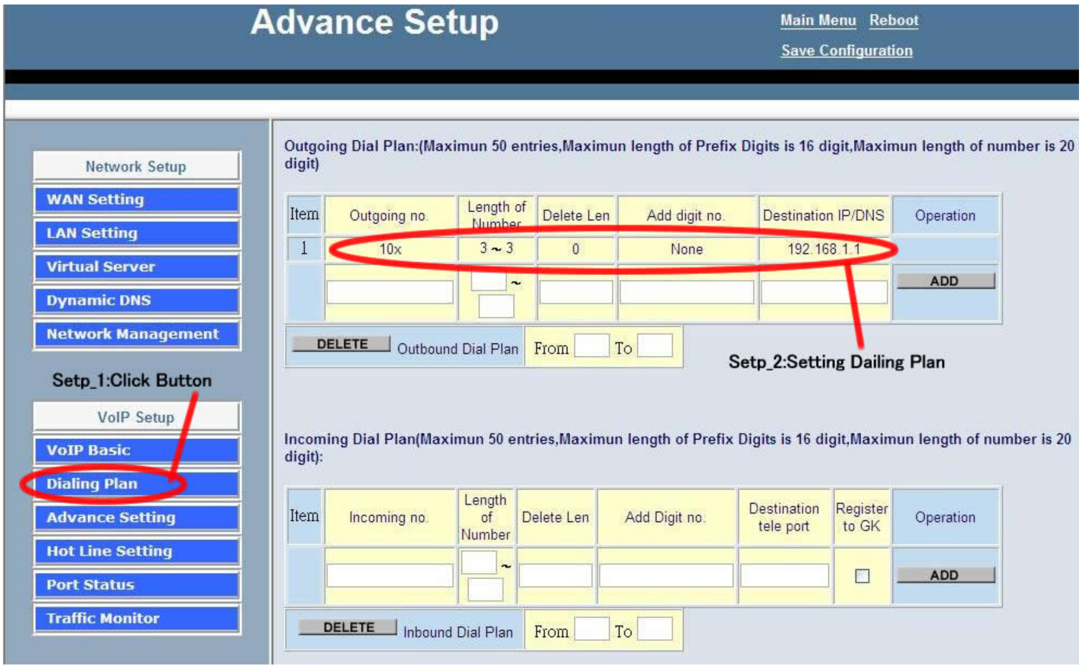
1. **Choose "VoIP Basic". Login in web interface, and in "Advance Setting".**
2. **Select you wan to use protocol (SIP/H.323).**
3. **Input you want to use call number.**

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 call.

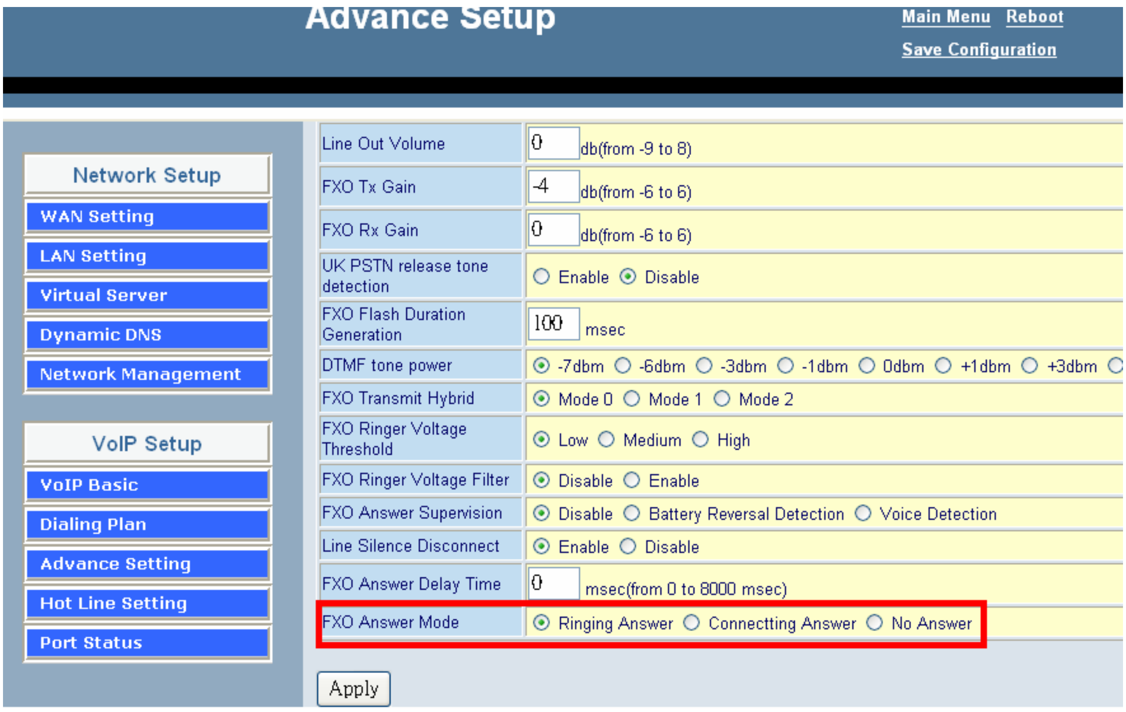
J FXO Answer Mode

FXO Answer Mode Setting

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

7. Ringing Answer Mode (Default Setting): FXO answer the call once the ring coming from PSTN line.
8. Connecting Answer Mode:
   Case A: "Hot Line Number" was NOT assigned in the FXO port. FXO answer the call once the ring come from PSTN line.
   Case B: "Hot Line Number" was assigned and the Hot line number belongs to remote VoIP device.
   In this case, FXO port will not answer (off-hook) the PSTN till the user picks up the call.
   (Note: This case can avoid charging for the Local PSTN call when the remoting VoIP devic still ring.)
   Case C: "Hot Line Number" was setting and the Hot line number was assigned to another FXS port in same Gateway. FXO port will not answer (off-hook) till the Phone (connected to the FXS port) was picked up by user.
   (Note: This case can avoid the Local PSTN charge when the FXS port still ring.)

9. Non Answer Mode: FXO will NOT answer the call in any time.
   (Note: Some ITSP only let the FXO for termination function, they do not user use the FXO port for origination)

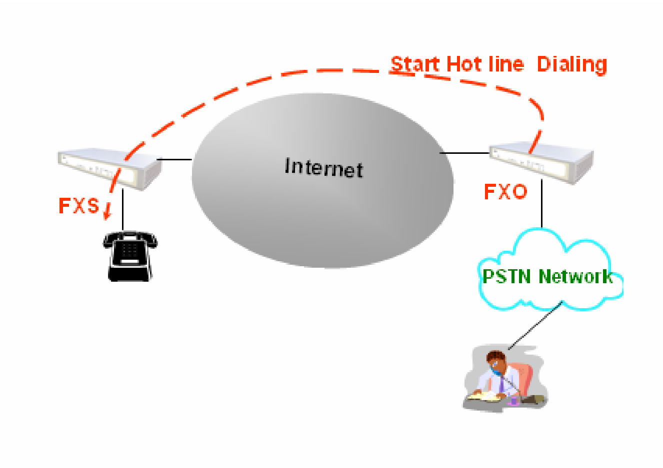


SIP Call Connecting Answer Mode

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

1. When the call com from PSTN to FXO, FXO start the Hot line dialing to remote SIP gateway
2. The phone of remote SIP gateway start ring.

3. When the phone was picked up, the remote SIP Gateway sends “SIP 200 OK” signal to FXO port.
4. Once FXO port receive the “SIP 200 OK” signal, FXO port would off-hook to answer the PSTN call.



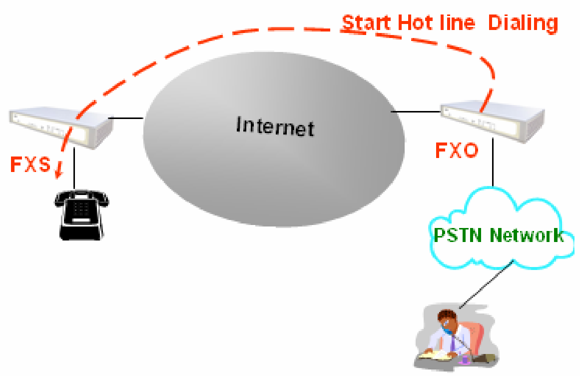
Case C: “Hot Line Number” was setting and the Hot line number was assigned to another FXS port in same Gateway.

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

H.323 Call Connecting Answer Mode

Case B: Hot Line Number” was assigned and the Hot line number belongs to remote H.323 device.(**Note: The remote H.323 device need Disable the “Auto Answer”**)

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



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

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