

Table of Contents

TABLE OF CONTENTS	2
BDEE & CES	۶
PREFACES	<u> </u>
0.1 About This Manual	5
0.2 Copyright Declarations	5
0.3 Trademarks	5
0.4 SAFETY INSTRUCTIONS	5
0.5 WARRANTY	5
INTRODUCE	6
1.1 Overview	6
	6
1.3 INTRODUCTION	7
1.4 FRONT PANEL LED INDICATORS & REAR PANELS 1.4.1 VOIP GATEWAY & SIP PROXY SERVER & GATEKEEPER OUTLOOK 1.4.2 FRONT PANEL LED AND CONTAINER DESCRIPTIONS	8 8 9
1.4.3 REAR PANEL DESCRIPTIONS	12
1.5 FEATURES AND SPECIFICATIONS	16
1.5.1 VOIP GATEWAY FEATURES 1.5.2 H.323 GATEKEEPER FEATURES – EMBEDDED SOFTWARE	16 17
1.5.3 SIP Proxy Server Features – Embedded Software	17
1.5.4 VOIP GATEWAY & EMBEDDED H.323 GATEKEEPER & SIP PROXY SERVER SPECIFICATIONS	17
INSTALLATION AND SETUP	19
2.1 PACKAGE CONTENT	19
2.1.1 L200/S200/S400 SERIES GATEWAY & EMBEDDED H.323 GATEKEEPER/ SIP PROXY SEI	RVER20
 2.1.2 S800 SERIES GATEWAY & EMBEDDED H.323 GATEKEEPER/ SIP PROXY SERVER 2.1.3 SB800/S1600/S2400 SERIES HIGH DENSITY GATEWAY 2.1.4 Call manager 	20 21
21	
2.2 INSTALLATION	22
2.3 SETUP	24
2.3.1 FACTORY DEFAULT SETTING 2.3.2 CONSOLE	24 24
2.3.3 TELNET	25
2.3.4 WEB USER INTERFACE	28

WIZARD FOR QUICK SETUP	32
3.1 WAN PORT TYPE SETUP	32
3.2 CONFIGURING NAT OR BRIDGE SETTING:	34
3.3 VOIP CALL PROTOCOL SETUP	35
GATEWAY SETTING	36
 4.1 NETWORK CONFIGURATION 4.1.1 WAN PORT TYPE SETUP 4.1.2 CONFIGURING LAN IP ADDRESS AND DHCP SERVER 4.1.3 VIRTUAL SERVER SETUP 4.1.4 DYNAMIC DNS 4.1.5 NETWORK MANAGEMENT 	38 38 40 41 41 42
4.2 VOIP SETUP 4.2.1 H.323 SETUP 4.2.2 SIP SETUP 4.2.3 DIRECT CALL (PEER TO PEER) SETUP 4.2.4 OTHER VOIP SETTING	43 43 51 57 60
4.3 SIP PROXY SERVER (SVR) SETUP	62
4.4 H.323 GATEKEEPER (GK) SETUP	67
4.5 CALL MANAGER SETUP	70
 4.6 SYSTEM ADMINISTRATOR 4.6.1 SAVE CONFIGURATION AND REBOOT 4.6.2 ACCESS CONTROL 4.6.3 SET TO DEFAULT CONFIGURATION 4.6.4 BACKUP/RESTORE CONFIGURATION TO A FILE 4.6.5 SYSTEM INFORMATION DISPLAY FUNCTION 4.6.6 SNTP SETTING FUNCTION 4.6.7 CAPTURE PACKETS FUNCTION 	71 71 72 72 73 73 73
4.7 UPDATE FIRMWARE(FOR GATEWAY & GK & SVR)	75
APPENDIX	78
A FAO LIST	78
B SIP SETTING VOIPBUSTER	80
C ANSWER SUPERVISION	82
D SIP SPEEDS CALL	84
E INTEROPERABILITY LIST	86
F RJ21 (TELCO 50) CABLE AND PATCH PANEL INSTALL	87
G SB800 / S1600 / S2400 Series module extension install	88
H GATEWAY VALUE SETTING	92
I SCENARIO APPLICATION SAMPLES	93

J FXO ANSWER MODE

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

Copyright 2006 Telephony Corporation. All rights reserved. This publication contains information that is protected by copyright. No part may be reproduced, transmitted, transcribed, stored in a retrieval system, or translated into any language without written permission from the copyright holders.

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	VolP	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	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 Seri	es (24 analo	og lines)							
	S2400	0	24	1	1	1	1	1	1	
	S2412	12	12	1	1	1	1	1	1	
	S2424	24	0	1	1	1	√	1	1	
	51600 Seri	es (16 analo	og lines)							
	S1600	0	16	1	1	1	1	1	1	
	S1608	8	8	1	1	1	1	1	1	
	S1616	16	0	1	1	1	√	1	1	
	SB800 Ser	ies (8 analo	g lines)							
	SB 800	0	8	1	1	1	1	1	1	
	SB 804	4	4	1	1	1	√	1	1	
	SB 808	8	0	1	1	1	√	1	1	
	S800 Serie	s(8 analog	lines Gatewa	ys /Embedded	I H.323 Gateke	epers/Em	bedded S	ip Pro	ky Serve	es)
	S800	0	8	1	1		1	1	V	SK800/SVR800
	S802	2	6	1	1		1	1	1	SK802/SVR802
	S804	4	4	1	1		√	1	1	SK804/SVR804
	S808	8	0	1	1		√	1	1	SK808/SVR808
•	S400 Serie	s (4 analog	lines Gatewa	ays/embedded	Gatekeepers/	Sip Proxy	Servers)			
	S400	0	4	4	1			1	1	SK400/SVR400
	S401	1	3	4	1			1	1	SK401/SVR401
	S402	2	2	4	1			1	1	SK402/SVR402
	S404	4	0	4	1			V	1	SK404/SVR404

Model Compare Table

<mark>S200 Serie</mark>	S200 Series (2 analog lines gateways/ Embedded H.323 Gatekeepers/ Embedded Sip Proxy Servers)								
S200	0	2	4	1			1	1	SK200/SVR200
S201	1	1	4	1			1	1	SK201/SVR201
S202	2	0	4	1			1	1	SK202/SVR202
L200 Serie	L200 Series (2 analog lines Gateways/embedded Gatekeepers/Sip Proxy Servers)								
L200	0	2	1	1			1	1	LK200/LVR200
L201	1	1	1	1			1	1	LK201/LVR201
L202(*)	2	0	1	1			1	1	LK202/LVR202(*)
Call Manager									
C400	4	0	4	1			1	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 Off	ATA is power ON ATA is power Off
2. WAN port	On Flashing Off	ATA network connection established Data traffic on cable network Waiting for network connection
3. LAN port	On Flashing Off	LAN is connected successfully Data is transmitting Ethernet not connected to PC
4. FXS	Off Flashing On	Telephone Set is On-Hook Ring Indication Telephone Set is Off-Hook
5. FXO	Off On	Line is On-hook Line is In-Use

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



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

	ACT On Flashing Off	LAN is connected successfully Data is transmitting Ethernet not connected to PC
5. FXS(Port)	Off Flashing On	Telephone Set is On-Hook Ring Indication Telephone Set is Off-Hook
6. FXO(Port)	Off On	Line is not enabled 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 Off	GW is power ON GW is power Off			
2. RUN port	On Flashing Off	GW connection established Data traffic on cable network Waiting for GW connection			
3. WAN port	100M On Off ACT ON Flashing Off	GW network connection 100MB network GW network connection 10MB network GW network connection established Data traffic on cable network Waiting for network connection			
4. LAN port	100M On Off ACT On Flashing Off	GW LAN connection 100MB network GW LAN connection 10MB network LAN is connected successfully Data is transmitting Ethernet not connected to PC			
5. FXS(Port)	Off	Telephone Set is On-Hook			

	Flashing On	Ring Indication Telephone Set is Off-Hook
6. FXO(Port)	Off On	Line is not enabled Line is busy
6. LCD Panel	Off On	System is Shutdown 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
- 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 :	Т.30 / Т.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.

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

Modules Card Table (SB800/S1600/S2400 Only)

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:



ltem	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)

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 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	X1
(SB800/S1600/S2400 Series)	
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) RJ-45 RS-232(only C800) AC Power Adapter CD-Rom(User manual) Modular Duplex Jack Extension Cord	X1 X1 (X1) X1 X1 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

Mad

	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.
- Open Terminal Program (ie. Windows XP Hyper Terminal) [Start] → [Program file] → [Accessories] → [communications] → [Hyper Terminal]

_	Part A	_					
C.	Accessories	۲	r,	Communications	¢	Fax	۲
6	Adobe	۶		Calculator	9	HyperTerminal	
۵	Adobe Acrobat 4.0	۶	嘗	Paint	8	Internet Connection Wizard	
e	Internet Explorer		2	Notepad	2	NetMeeting	
W	Microsoft Word			×	2	Network and Dial-up Connections	
C 3	Outlook Express		T		G	Phone Dialer	
(b)	Terminal Services Client	۲			C	HyperTerminal	Þ

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

-	at at	
	New Connection Enter a name and choose an icon for the connection: Name Connection Test	
	CK. Cancel	

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

(Setting Table following as Teinet Setting table)
COM1 Properties
Port Settings
Bits per second: 38400
Data bits: 8
Parity: None
Stop bits: 1
The second Divers
Flow control: None
Restore Defaults
OK Cancel Apply

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 222.222.22.1)

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) - 🗆 ×

Login : (1) Welcome to 2FXS+2FXO VoIP Gateway Version 2.7.8	
Main Menu	
WAN Status:PPPoE OK (NAT Mode) DDNS:DDNS OK (4) VoIP Status:SIP Direct Mode (3)	
[1] Advanced Setup. (5) [2] System Administration.	
[3] Save Current Configurations. [4] Upgrade Software.	
[5] Fing. [6] Logout.	
[7] Restart. Please Select 1 - 7:	
	-

4. Gateway Telnet Setting Tab	le, 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.Change WAN Type to DHCP
1.WAN Setting	2.Change WAN Type to Fixed IP
	3.Change PPPoE Username
	4.Change PPPoE Password
[1]Advanced Setup	1.Change to Bridge Mode
2.LAN Setting	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	1.Add Virtual Server
3.Virtual Server	2.Delete Virtual Server
[1]Advanced Setup	1.Change DDNS username
4.Dynamic DNS	2.Change DDNS password
	3.Change DDNS domain name
	4.Change DNS server IP
[1]Advanced Setup	1. Change web server port
[1]Advanced Setun	1 Change VolP Protocol to H 323
6 VolP Basic	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	1.Add Outbound Direct Call
	3 Add Inhound Direct Call
	4 Delete Inbound Direct Call
[1]Advanced Setup	(1)Sip Advance
8.VoIP Advance Setting	1.Set DTMF Relay Mode
g	2.Change FAX Mode
	3.Change RFC2833 Payload(96-127)
	3.Change RFC2833 Payload(96-127) (2)Telephone Advance
	3.Change RFC2833 Payload(96-127) (2)Telephone Advance 1.VAD(Silence Compression)On/Off
	3.Change RFC2833 Payload(96-127) (2)Telephone Advance 1.VAD(Silence Compression)On/Off 2.Change Codec
	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?
	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
	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
	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
	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 FXS Impedance
	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
	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.Elash Detection
	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 Bing Frequency
	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
	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
	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
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume Filash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change FXO Impedance
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume FIash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change FXO Impedance Change Line Out Volume
	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
	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
	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
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume Flash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change Line Out Volume Change FXO Tx Gain Change FXO Rx Gain Flash Duration
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume FIash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change Line Out Volume Change FXO Tx Gain Change FXO Rx Gain Flash Duration TDTMF Tone Power FXO Transmit Hybrid
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume FIASh Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change FXO Rx Gain Flash Duration DTMF Tone Power FXO Transmit Hybrid Answer Supervision Setting
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume Fash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change Line Out Volume Change FXO Tx Gain Flash Duration DTMF Tone Power FXO Transmit Hybrid Answer Supervision Setting Change Ringer Voltage Threshold
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance VAD(Silence Compression)On/Off Change Codec Enable/Disable UK PSTN Tone Detection? Enable/Disable Dial Complete Tone Dial Termination Key Setting FXS Parameters Setting Change FXS Impedance Change Phone In Volume Change Phone Out Volume Fash Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change FXO Impedance Change FXO Impedance Change Line Out Volume Change FXO Impedance Change EXO Tx Gain Change FXO Tx Gain FIash Duration DTMF Tone Power FXO Transmit Hybrid Answer Supervision Setting Change Ringer Voltage Threshold Enable Line Silence Disconnect
	 3.Change RFC2833 Payload(96-127) (2)Telephone Advance 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 Change FXS Impedance Change Phone In Volume Change Phone Out Volume Fish Detection Ring Frequency FXS Battery reversal generation 7.FXO Parameters Setting Change FXO Impedance Change FXO Impedance Change FXO Impedance Change FXO Impedance Change Exo Impedance Change Exo Tx Gain Schange Line Out Volume Change FXO Rx Gain Flash Duration DTMF Tone Power FXO Transmit Hybrid Answer Supervision Setting Change FXO Answer delay time Change FXO Answer delay time

	(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.*

our compu	iter uses to
ompaters.	
A 44	
	our compu omputers.

Select Protocol and click the Add button. The Select Network Protocol dialog box will open.
 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.

🚣 Local Area Connection Properties	? ×
General Authentication Advanced	
Connect using:	
Intel(R) PRO/100 VE Network Connection	
Configure	
This connection uses the following items:	-
Install	
Description Transmission Control Protocol/Internet Protocol. The default wide area network protocol that provides communication across diverse interconnected networks.	
Show icon in notification area when connected	
OK Can	cel

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 222.222.222.100 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.

You can get IP settings assigned aut his capability. Otherwise, you need t the appropriate IP settings.	tomatica to ask y	ally if y our ne	our nei twork	twork s adminis	upports trator for
Obtain an IP address automatic	cally				
- OUse the following IP address: -					
IP address;			+	-	_
Subnet mask:					
Default gateway:		1	1	- 83	
Obtain DNS server address au	tomatic	ally			
— Use the following DNS server a	address	es:			_
Preferred DNS server:		3	10	•22	
Alternate DNS server:		3	e.	•2	
				Adv	anced

Connecting to the Web Configuration via a Web Browser 1. Launch the Web browser(IE or Firefox). Enter http://222.222.22.1 into the browser Address window and press the Enter Key

(j) ▼ [€] http://222.222.22.1	🟉 http://mctony.gotdns.com/ - Windows Internet Explorer	
	() ▼ [] http://222.222.1]	
Co + @http://mctony.gotdns.com/	C + //mctony.gotdns.com/	

2. An authentication dialog box will open.

Enter Netv	vork Passwo	rd	<u>?</u> ×
? >	Please type y	our user name and password.	
3	Site:	222.222.222.1	
	Realm	Web Server	
	<u>U</u> ser Name	admin	
	Password	admin	
	□ <u>S</u> ave this	password in your password list	
		OK Ca	ncel

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.

System Configuration	
Setup Wizard Advance Setup System Administration	Setup Wizard: Let you configure your VoIP gateway in three simple steps and within 3 minutes. Advance Setup: Let you configure advance VoIP gateway features. System Administration: View system information and save system configuration.
©2004 Telephony Solution	

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:

	Setup Wizard Main Menu Reboot
WAN Type Selection: 2.NAT Setting 3.VoIP Call Setting	 1.WAN Type Please specify the WAN connection type required. Please select one of these three types: DHCP Client for Cable modern, Fixed-IP, or PPPoE for ADSL modern. DHCP - Cable Modern This function will be automatically configured when plugged into the cable modern. If there is a Domain Name Server (DNS) that you would input to use. Fixed - IP Internet Service Providers may assign a fixed IP address for your VoIP Router. Select this option and enter the assigned IP address, subnet mask, gateway IP and DNS IP addresses for your Barricade. PPPoE - ADSL If you connect to the Internet using an ADSL Modern, please enter the username and password which provided from ISP.

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.

	Setu	p Wizard	<u>Main Menu</u> <u>Reb</u>	oot
WAN Type Selection: 2.NAT Setting	2 Fixed-IP			
O 3.VolP Call Setting		IP Address	59 . 120 . 54 .	62
		Default Router IP Address	59 , 120 , 54 ,	254
		Subnet Mask	255 . 255 . 255 .	0
	Enter the IP address, Dr above.	efault Router IP address and Subne	et Mask provided to you by yo	our ISP in the appropriate fields

- **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.

WAN Type Selection: O 2.NAT Setting O 3.VoIP Call Setting	3.PPPoE Type PPPoE Configuration :
	Use PPPoE Authentication
	User Name(MAX. 40 characters) :
	Password(MAX. 40 characters) :
	Confirm password :
	Enter the User Name and Password required by your ISP.

- 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

	Setup Wizard	<u>Main Menu</u>	<u>Reboot</u>
WAN Type Selection: 2.NAT Setting 3.VolP Call Setting	DHCP Client Enabled !		

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:

WAN Type Selection	NAT Settings
O 3. VolP Call Setup	O Bridge Mode ⊙ NAT Mode
	You can use NAT to allow PCs from LAN subnet for accessing Internet.
	LAN IP Setting
	IP Address 222 . 222 . 1
	IP Subnet Mask 255.255.255.0
	BACK FINISH

- **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: 222.222.221.51).
- **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"

O 1. WAN Type O 2. NAT Setting ♂ 3. VoIP Call Setup	VoIP Protocol Selection VoIP Call Settings	SP H.323 SP Port number	Select er Setup :
		Port 1 number	11001
		Port 2 number	11002
		SIP Proxy Server IP address	59.120.54.63/5060
	lf you don'	t use sip proxy server, you sho	uld set the following outgoing dial plan.

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.

N-ID C-II C-III	ltem	Phone Number	Min Digit	Max Digit	Strip Len	Prefix Number	IP Address
. voir can setup	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:
Network Setup	
WAN Setting	
LAN Setting	
Virtual Server	
Dynamic DNS	
Network Management	
VOIP Setup	
VoIP Basic	
Dialing Plan	
Advance Setting	
Hot Line Setting	
Death Obeless	
Port Status	

Advance Setup

WAN Setting: Set WAN port network parameters.

Set wan port network parameters.

LAN Setting: Set LAN port network parameters.

Virtual Server: Set virtual server like WEB,FTP or E-MAIL server.

DDNS Setting: Set DDNS server IP address.

Network Management: Set web server,telnet server port

VoIP Basic: Set VoIP basic parameters such as VoIP protocol selection, phone number.

Dial Plan: Set outbound and inbound dial plan.

Advance Setting: Set advance parameters such as codec,voice volume

Auto Dial Setting: Set auto dial number

Port Status: Display current telephone port status

_	Advanced S	ettin	ig L	abel
ſ	Network Set	un I	ahe	7

Network Setup Laber	
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:

System Administration

Main Menu Reboot

	System Administration
Management	Save Configuration:
Save Configuration	Save current system configuration.
Access Control	Access Control:
Set to default	Set system administrator username and password.
Backup/Restore Configuration File	Set to Default:
System Information	Set to default configuration.
SNTP Setting	Backup/Restore Configuration file:
Syslog setting	Backup current configuration to PC/Restore system configuration from PC backup file.
Capture packet	System Information:
	Display current system information.
	SNTP Setting:
	SNTP parameter setting.

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 <u>http://www.ethereal.com/</u>)

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.

Network Setup	WAN Port Type Configuration		
WAN Setting	WAN Type Setting	Static IP 🗸 Select	
LAN Setting	ID Address	192168116	
Virtual Server	Subnet Mask	255.255.255.0	
Dynamic DNS	Default Router	192.168.1.254	
Network Management			
	Apply		

- **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.

Network Setup			
WAN Setting			PPPoE 🖌 Select
LAN Setting			
Virtual Server			Use PPPoE Authentication
Dynamic DNS		User Name(MAX. 40 characters) :	87993911@hinet.net
Network Management		Password(MAX. 40 characters) :	
	WAN Type Setting	Confirm Password:	•••••
		Get IP Address:	61.216.39.205
		Get Default Router:	61.216.39.206
		Enter the Use	or Name and Password required by your ISP.
VoIP Setup			Apply
VoIP Basic			
Dialing Plan			

- **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

Network Setup	Set Network Parameters	
WAN Setting	WAN Type Setting	DHCP V Select
LAN Setting	IP Address	192 168 1 16
Virtual Server	Subnet Mask	255.255.0
Dynamic DNS	Default Router	192.168.1.254
Network Management		
	Apply	

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

	LAN Mode Selecting
Network Setup	
WAN Setting	Bridge Mode Let WAN and LAN Port as Bridge
LAN Setting	NAT Mode (Default) Let PCs in LAN subnet can access Internet
Virtual Server	LAN IP Address Setting
Dynamic DNS	
Network Management	IP Address 222 222 222 1
	IP Subnet Mask 255.255.255.0
	DHCP Server 💿 Enable 🔿 Disable
	DHCP Server Setting :
	Start IP address 222.222.222. 50
VoIP Setup	End IP address 222,222, 100
VoIP Basic	DNS Server IP 168 95 1 1
Dialing Plan	Leased Time (min =60) 7200 seconds
Advance Setting	
Hot Line Setting	Apply

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 222.222.100.

	Virtual Server Configuration:				
Network Setup	Ulistual Casuar Cattina				
WAN Setting	virtual Server Setting Remote Users can access services such as the Web or FTP at your local site via public IP addresses can be automatically				
LAN Setting	redirected to lo	redirected to local servers configured with private IP addresses.			
Virtual Server		Private IP	Private Port	Public Port	
Dynamic DNS	1	222 222 222 100	21	21	
Network Management	1.	222.222.222.			
	2.	222.222.222.			
	3.	222.222.222.			
	4.	222.222.222.			
	5.	222.222.222.			
VoIP Setup	6.	222.222.222.			
VoIP Basic	7.	222.222.222.			
Dialing Plan	8.	222.222.222.			
Hot Line Setting	9.	222.222.222.			
Port Status	10.	222.222.222.			
		Apply			

- 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)

	DDNS(Dynamic DNS) Service Configuration:			
Network Setup				
WAN Setting		DDNS Service		
LAN Setting	Dynamic DNS allows you to provide Internet users with a domain name (instead of an IP Address) to access your Virtual Servers.			
Virtual Server	Register for this FREE service at <u>http://m</u>	embers.dyndns.org/newacct		
Dynamic DNS				
Network Management	DDNS Data			
	DDNS username	sdicxp		
	DDNS password			
	DDNS domain name	mctony.gotdns.com		
	DNS Server IP	168.95.1.1		
VoIP Setup		Apply		
VoIP Basic				

- 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)

	Access Service Configuration (HTTP Port and TELNET Port Configuration):		
Network Setup			
WAN Setting	Access Port Service		
	Access Port Configuration allows you to modify the HTTP port or TELNET port for accessing VoIP gateway		
LAN Setting			
Virtual Server	11777D. Counting Deart		
Dynamic DNS			
	Telnet Service Port 23		
Network Management			
	Apply		

- 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 VolP 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!

	VoIP Basic Configuration	
Network Setup	VelD Desteral Co	H 323 W Select
WAN Setting	VolP Protocol Se	H.323
LAN Setting	E.164 Number S	etting (SUP) digit) :
Virtual Server	Port 1 E.164 Number	none
Dynamic DNS	Port 2 E 164 Number	none
Network Management		
,,	Port 3 E.164 Number	none
	Port 4 E.164 Number	none
	Caller ID / ANI Setting for O	ff-Net Call Setting (MAX 20 digit) :
1	Port 1 Caller ID / ANI	none
VoIP Setup	Port 2 Caller ID / ANI	none
VoIP Basic		
Dialing Plan	Port 3 Caller ID / ANI	none
Advance Setting	Port 4 Caller ID / ANI	none
Hot Line Setting	<u></u>	1

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)

	Phone Page Select	Page 1 V Select
Network Setup	Thone Tage Select	Page 1
WAN Setting	E.164 Number Se	Page 2 0 digit) : Page 3
LAN Setting	Port 1 E.164 Number	101
Virtual Server		
Dynamic DNS	Port 2 E.164 Number	102
Network Management	Port 3 E.164 Number	103
VoIP Setup	Port 4 E.164 Number	104
VoIP Basic	Port 5 E.164 Number	105
Dialing Plan		
Advance Setting	Port 6 E.164 Number	
Hot Line Setting	Port 7 E.164 Number	107
Port Status	Port 8 E.164 Number	108

- 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.

	Caller ID / ANI Setting for Off-Net Call Setting (MAX 20 digit) :
Network Setup	
WAN Setting	Port 1 Caller ID / ANI
LAN Setting	Dest 2 Collect D (ANI
Virtual Server	
Dynamic DNS	Port 3 Caller ID / ANI
Network Management	Port 4 Caller ID / ANI
VoIP Setup	Port 5 Caller ID / ANI
VoIP Basic	Port & Caller ID / ANI
Dialing Plan	
Advance Setting	Port 7 Caller ID / ANI
Hot Line Setting	Port 8 Caller ID / ANI
Port Status	

- 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 Para	neter Setting :
Network Setup	H323 ID	
WAN Setting	Primary GateKeeper IP address	0.0.0
LAN Setting	Secondary GateKeeper IP address	0.0.0
Virtual Server	Primary H.323 GateKeeper Domain Name	
Dynamic DNS	Secondary H.323 GateKeeper Domain Name	
Network Parameters	H.323 Gatekeeper ID	
ValD Ostur	Voice Caps Prefix	
VoiP Setup	RAS Port Adjustment	1719
VoIP Basic	Q.931 Port Adjustment	1720
Dialing Plan		
Advance Setting	H.323 Call Pass Thro	ugh NAT Configuration :
Auto Dial Setting		
Port Status	NAT Pass Method	● Disable ○ Auto Pass NAT ○ Manual (Need Key In Public IP Address)
Traffic Monitor	Public IP Address	0.0.0.0

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		
Secondary Gatekeeper Domain Name	Let user use Domain Name of H.323 Gatekeeper.	
H.323 Gatekeeper ID	The Gatekeeper ID; usually do not need to set this field unless the gatekeeper must need this value.	
Voice Cap Prefix	Let user set prefix number in RRQ nonstandard 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	 Disable : The Gateway operates in public IP address Auto Detection: When the Gateway register to GNU Gatekeeper / H.323 Gatekeeper (SK Series), please select this option. 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]

Advance Setting

Advance Setting Select VoIP Advance 💉 Select

⊙ Outband (by H.245) ○ Inband (by RTP)
O Normal-Start ⊙ Fast-Start
○ Enable Disable
 T.30 ○ T.38 T38UDP Low Speed Redundancy Level 5 ▼ T38UDP High Speed Redundancy Level 0 ▼
⊙ Gateway ○ Terminal
0 seconds
⊙ On ⊖ Off
○ Enable
O Disable 💿 Enable

Apply

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] Advance Setting

	Advance Setting Select Telephone Advance 🖌 Select
Silence Compression Voice Activity Detection	
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	 ⑥ 600 ○ 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	-3db(from -9 to 8)
Line Out Volume	-3 db(from -9 to 8)
FXO Tx Gain	4db(from -6 to 6)
FXO Rx Gain	0 db(from -6 to 6)
UK PSTN release tone detection	🔿 Enable 💿 Disable

FXS Flash Detection	100 ~ 500 msec
FXO Flash Duration Generation	100 msec
Ring Frequency	20 Hz
DTMF tone power	💿 -7dbm ○ -6dbm ○ -3dbm ○ -1dbm ○ 0dbm ○ +1dbm ○ +3dbm ○ +6dbm
FXO Transmit Hybrid	Mode 0 ○ Mode 1 ○ Mode 2
FXO Ringer Voltage Threshold	⊙ Low ○ Medium ○ High
FXO Ringer Voltage Filter	O Disable ○ Enable
FXS Battery Reversal Generation	
FXO Answer Supervision	O Disable ○ Battery Reversal Detection ○ Voice Detection
Line Silence Disconnect	🔿 Enable 💿 Disable
FXO Answer Delay Time	0 msec(from 0 to 8000 msec)
FXO Answer Mode	

Apply

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] Advance Setting

Advance Setting Select Network Advance 💌 Select

Smart QOS	⊙ Enable ○ Disable	
Bandwidth Control	Downstream 512 Kbps Upstream 64 Kbps	
G.723 Bandwidth	◯ 18kbps ⊙ 12kbps ◯ 10kbps ◯ 8kbps	
G.729 Bandwidth	○ 40kbps ○ 24kbps ○ 19kbps ○ 16kbps ○ 15kbps ④ 14kbps	
IP TOS	O Enable 💿 Disable	

Apply

ltem	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.

	VoIP Basic Configuration						
Network Setup	ValD Protocol Satting SP V Select						
WAN Setting							
LAN Setting			Po	rt Number / Password Set	ting(MAX 20 digit) :		
Virtual Server	No.	Number	Reg	Account	Password	Register Status	Reason
Dynamic DNS	1	500		500		Success	OK
Network Management	2	501		501		Success	ОК
	3	502		502		Success	ОК
	4	503		503	•••	Success	ОК
		Jse Public Account (PORT	1)		○ Enable ⊙ Disable		
VoIP Setup	SIP Hunting Table :						
VoIP Basic	No).		Hunting	Member		
Advance Setting	1			🗹 Port 1 🗖 Port 2	🗆 Port 3 🗖 Port 4		
Hot Line Setting	2	2 Dort 1 🗹 Port 2 🗌 Port 3 🗌 Port 4					
Port Status	3		Port 1 Port 2 Port 3 Port 4				
	4	4 Port 1 Port 2 Port 3 Port 4					

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)

No.	Number	Reg	Account	Password	Register Status	Reason
1	500		500		Success	ОК
2	501		501		Success	ОК
3	502		502		Success	ОК
4	503		503		Success	ОК
Use Public Account (PORT 1) O Enable Disable						

- 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.

No.	Hunting Member
1	🗹 Port 1 🔲 Port 2 🔲 Port 3 🔲 Port 4
2	🗖 Port 1 🗹 Port 2 🗖 Port 3 🗖 Port 4
3	🗖 Port 1 🗖 Port 2 🗹 Port 3 🗖 Port 4
4	🗖 Port 1 🗖 Port 2 🗖 Port 3 🗹 Port 4

SIP Hunting Table :

No.	Hunting Member
1	2,3,4,5,6,7
2	1,3,5,6,7,8,9,10,11,12,13,14,15,16,17,18,19,20,21,22,23,24
3	
4	
5	
6	
7	
8	

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 Setting :				
Domain/Realm	sdlcxp.dyndns.biz			
SIP Proxy Server	sdlcxp.dyndns.biz/5060			
Register Interval(seconds)	900			
SIP Authentication	💿 Enable 🔿 Disable			
Outbound Proxy Server	00.0.00			

SIP Proxy Server Label	
SIP Proxy Server Setting	 Enter the SIP service IP address or domain name in this field (the domain name that comes after the @ symbol i n a full SIP URI). Use Net2Phone Service Provider
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.

NAT Pass Setting:				
NAT Pass Method	○ STUN ⊙ Symmetric RTP			
STUN Server address	64.69.76.21			
STUN Server port	3478			
Local Setting:				
Local SIP Port	5060			
Apply				

- **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]

Advance Setting	
	Advance Setting Select VolP Advance v Select
DTMF Relay for SIP	O Inband
RFC2833 Payload	101 (from 96 to 127)
FAX Mode	 T.30 ○ T.38 T38UDP Low Speed Redundancy Level ⁵ ▼ T38UDP High Speed Redundancy Level ⁰ ▼
Watchdog	O Disable 💿 Enable

Apply

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						
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	Ͽ 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	-3db(from -9 to 8)					
Line Out Volume	-3db(from -9 to 8)					
FXO Tx Gain	4db(from -6 to 6)					
FXO Rx Gain	0db(from -6 to 6)					
UK PSTN release tone detection	O Enable 💿 Disable					

FXS Flash Detection	100 ~ 500 msec			
FXO Flash Duration Generation	100 msec			
Ring Frequency	20 Hz			
DTMF tone power	⊙ -7dbm ○ -6dbm ○ -3dbm ○ -1dbm ○ 0dbm ○ +1dbm ○ +3dbm ○ +6dbm			
FXO Transmit Hybrid	Mode 0 ○ Mode 1 ○ Mode 2			
FXO Ringer Voltage Threshold	⊙ Low ○ Medium ○ High			
FXO Ringer Voltage Filter	⊙ Disable ○ Enable			
FXS Battery Reversal Generation	⊙ Enable 🔿 Disable			
FXO Answer Supervision	O Disable ○ Battery Reversal Detection ○ Voice Detection			
Line Silence Disconnect	🔿 Enable 💿 Disable			
FXO Answer Delay Time	0 msec(from 0 to 8000 msec)			
FXO Answer Mode	💿 Ringing Answer 🔿 Connectting Answer 🔿 No Answer			
	Apply			

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]

Advance Setting Advance Setting Select Network Advance Select Smart QOS Enable Disable Downstream Stype Upstream Kbps G.723 Bandwidth 18kbps 12kbps 10kbps 12kbps 10kbps 12kbps 10kbps 12kbps 14kbps 14kbps 14kbps

Apply

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
 Operation

 Port
 Outgoing no.
 Length of Number
 Delete Len
 Add digit no.
 Destination IP/DNS
 Destination
 Operation

 DELETE
 Outbound Dial Plan
 From
 To
 To
 To
 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	Dutgoing Dial Plan:(Maximun 50 entries,Maximun length of Prefix Digits is 16 digit,Maximun length of number is 20 digit)									
Item	Outgoing no.	Length of Number	Delete Len	Add digit no.	Destination IP/DNS	Destination Port	Operation			
1	001x	4 ~ 20	0	None	215.214.1.1	5060				
2	002x	4 ~ 20	0	None	h323.gw.net	5060				
		~					ADD			
	DELETE Outbound Dial Plan From To									

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

Example2: Speed Dial

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

Item	Outgoing no.	Length of Number	Delete Len	Add digit no.	Destination IP/DNS	Destination Port	Operation
1	401	3~3	3	1334588712	211.22.3.14	5060	
2	402	3~3	3	2212345612	211.21.2.76	5060	
		~					ADD
DELETE Outbound Dial Plan			From	To			

1. If user dial "401",

Gateway automatically dial "1334588712" to Destination IP address: 211.22.3.14 2. If user dial "402",

Gateway automatically dial "2212345612" to Destination IP address: 211.21.2.76 In the "Incoming Dial Plan Configurations" settings: Maximum Entries : 50

Incoming Dial Plan(Maximun 50 entries,Maximun length of Prefix Digits is 16 digit,Maximun length of number is 20 digit):									
Item	Incoming no.	Length of Number	Delete Len	Add Digit no.	Destination tele port	Operation			
		~				ADD			
DE	ELETE Inbound Di	al Plan Fr	om To						

- "Inbound number" is the leading digits of the dialing number.
- "Length of Number" has two text fields need filled: "Min Length" and "Max Length" is the min/max allowed length you can dial.
- "Delete Length" is the number of digits that will be stripped from beginning of the dialed number.
- Add Digit Number" is the digits that will be added to the beginning of the dialed number.
- "Destination Tele port" is "Tel-port"; this is for local dial plan setting phone number.

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.

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		
DE	DELETE Inbound Dial Plan From To							

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			
DE	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 81xxxxxxx).

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 92xxxxxxx).

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	092×	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.

	Hot Line Number Setting (Hotline Setting)									
Network Setup										
WAN Setting		Port 1 number	123							
LAN Setting		Port 2 number	None							
Virtual Server		Port 3 number	None							
Dynamic DNS		Dort 1 number	None							
Network Management		Fort 4 number								
		A	pply							

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.	Ρ	ort	Status:
--------------	---	-----	---------

	Phone Page Select Page 3 V Page 1 Page 1 Page 2											
Port No.	Туре	Status	Codec	Direction	Dial No.	Pag Pag	e2 e3	er 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.

System Configuration								
Advance Setup SIP Proxy Server System Administration	Advance Setup: Let you configure advance VoIP gateway features. SIP Proxy Server Setting: Let you configure SIP Proxy Server parameters. System Administration: View system information and save system configuration.							
©2004 Telephony Solution								

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



2. Setting Proxy Parameter

	Proxy Parameter Setting		
SIP Proxy Function Proxy Parameter		Register Expired Time(seconds)	3600
Authentication Register UA		SIP Server Port	5060
Real Time CDR		A	pply

- 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.

	MD5 authentica	ation:				
SIP Proxy Function	 Enable MI 	⊃5 authentication	Enable SIP MD5 authentication			
Proxy Parameter	O Disable M	D5 authentication	Disable SIP MD5 authentication			
Authentication			Apply			
Register UA						
Real Time CDR	Username/Pass	word MD5 authentication:(Max.)	250 username/password,Max. usernar	ne/password 30 digit)		
CDR	Item	Username	Password	Operation		
Call Statistic	1	500	500			
	2	501	501			
	3	502	502			
	4	503	503			
	5	504	504			
	6	505	505			
	7	506	506			
	8	507	507			
	9	508	508			
	10	509	509			
	11	510	510			
	12	1000	test			
	13	2000	tt2000			
	14	7600	7600			
				ADD		
		DELETE Item	From To			

- 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.

Provy Eunstion	1	2					
Proxy Function	100		00				
Parameter	100	2	00				
entication	Peristered II	ser Agent/Maximun	n 250 user age	nt Current Pegie	tered = 4 Total P	age = 1 Current	Page = 1):
ster UA	Registered o	sei Agendinaximun	ii 200 user age	in,ourient Regia	tereu - 4, rotarr	age - 1,ourrent	1 age - 1/.
Time CDR	Item.	Call-id	UserInfo	Contact Address	Real Address	Expires	Remaining Time.
Statistic	1	qhdhg4as438c1mv67pg: - 1	x 501	<u>61 216 34 158 5060</u>	61.216.34.158:5060	900	3439
Sutistic.	2	qhdhg4as438c1mv67pg: - 2	x 502	61.216.34.158:5060	61.216.34.158:5060	900	3439
	3	qhdhg4as438c1mv67pg: - 0	× 500	61.216.34.158.5060	61.216.34.158:5060	900	3439
	4	qhdhg4as438c1mv67pg; - 3	× 503	61.216.34.158:5060	61.216.34.158:5060	900	3440
				-	<u>I</u>		

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

Call Detail Record(Maximum CDR = 50,Current CDR = 6,Total Page = 1,Current Page = 1)										
SIP Proxy Function	You ca system	n install syslog serv administration and	ver on your PC to let I set syslog ip addre	t SIP proxy server s ss to your PC	end these CDR to y	/our PC,just go to s	yslog setting at			
Authentication	No.	Caller IP Address	Callee IP Address	Calling Number	Caller Number	Duration(Second)	Call Date			
	6.	218.168.182.243	61.218.109.83	501	100	2	2006/4/3 11:10:49			
Register UA	5.	218.168.182.243	61.218.109.83	501	100	13	2006/4/3 10:41:53			
Real Time CDR	4.	218.168.182.243	61.218.109.83	501	100	11	2006/4/3 10:39:23			
CDR	3.	61.218.109.83	218.168.182.243	100	501	5	2006/4/3 10:39:5			
Call Statistic	2.	218.168.182.243	218.168.182.145	501	100	6	2006/4/3 10:30:47			
	1.	218.168.182.243	218.168.182.145	501	100	8	2006/4/3 10:29:42			

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

	Top 20 by:	Top 20 by:										
SIP Proxy Function	Duration	✓ Select										
Proxy Parameter	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						
Authentication	5	218.168.182.243	61.218.109.83	501	100	11						
Register UA	4	218.168.182.243	218.168.182.145	501	100	8						
Real Time CDR	3	218.168.182.243	218.168.182.145	501	100	6						
CDR	2	61.218.109.83	218.168.182.243	100	501	5						
	1	218.168.182.243	61.218.109.83	501	100	2						
Call Statistic		,			,							

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

SIP Proxy Server						<u>Main Me</u>	nu <u>Rebo</u>	ot	
						<u>Save Co</u>	onfiguratio	n	
Trur -sir Trunk 1	1k S ⁹ trunk	etting ^{route-}				Trunk1	Trunk2	Trunk3	Trunk4
	ITSE	>	~	Select					
	ITSF))				
	Tern	nination Gateway	Set	ting(MAX 20	digit):				
	Reg	Account			Passw	ord	Register Status	Reason	
		0949102666		06852	64664		Success	ок	
	SIP Serv	vice Provider Setting :							
	Domain/Realm S			avecom	i.net.tw				
	SIP Proxy Server / Host Name S			sip.savecom.net.tw/5060					
		Register Interval	900	00					
		Outbound Provy	0.0.0	0.0			7		

- 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.

SI	P Pi	oxy Server		<u>Main Menu</u> <u>Reboot</u>					
				<u>Save Co</u>	nfiguratior	1			
	Reg	Account		Password	Register Status	Reason			
		0949102666		0685264664	Success	ок			
	SIP Ser	vice Provider Setting :							
		Domain/Realm	sip.sa	avecom.net.tw]				
	SIP P	roxy Server / Host Name	sip.sa	avecom.net.tw/5060]				
		Register Interval	900]				
		Outbound Proxy	0.0.0	0.0.0/0					
	·		,	Apply					
Incoming	Incoming Call Attendant								
	Incomir	g Call Attendant: Max 1 Er	ntries.						
		Item	Extensio	ons number (one of the list registser number)	Operat	tion			
	1	1	703		DELE	ETE			

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.

	SIP Proxy S	erver	Main Menu Reboot									
	Save Configuration											
Outgo	Outgoing Dial Rule :											
Outgo	oing Dial Rule : Max 8 Ent	ries.										
ltem	Outgoing Number	Length Min	Length Max	Delete Length	Add Digit	Select	Ext. Number					
1	001×	4	20	0		Trunk 1						
2	0044×	5	20	0		Trunk 2						
3	0086×	5	20	0		Trunk 3						
4	00×	3	20	0		Trunk 4						
New						Trunk l 💌						

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.

System Configuration	
Setup Wizard Advance Setup Gatekeeper System Administration 中文設定	Setup Wizard: Let you configure your VoIP gateway in three simple steps and within 3 minutes. Advance Setup: Let you configure advance VoIP gateway features. System Administration: View system information and save system configuration. Gatekeeper: H.323 Gatekeeper function.
©2004 Telephony Solution	

1. Select Gatekeeper option and setting gatekeeper.

	Gatekeeper Function
Gatekeeper Function	Register Gateway:
Register Gateway	Display register gateway information
Security Policy	Security Policy:
CDR	con-
Realtime CDR	Display call detail record(CDR)
Call Statistic	Realtime CDR:
Pre-Granted GW	Display current call detail record(CDR)
RAS Port	Call Statistic: Show call statistic information such as top 10 call duration
	Pre-Granted GW: Set pregranted gateway IP address and phone number Syslog Server: Set syslog server IP address RAS Port: Set RAS Port when standard 1719 port is block

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

	R	egistered	Gateway(Maximum = 250,Curren	t Registered = 7,Tota	Il Page = 1,Current Page = 1)	
Gatekeeper Function		No.	Gateway IP Address	H323 ID	Phone Number	MAC Address
Register Gateway		1.	<u>61.6.204.75:1720</u>	MTM Office	6732666,6732777,6732888,6732999	
Security Policy		2.	61.221.176.236:2501(under nat)	Rich	1781,1782	
CDR		3.	83.70.203.184:2501(under nat)	2tel-2	353300,353400	
Realtime CDR		4.	<u>61.6.192.155:1720</u>	alex	6732211,6732233,6732277,6732288	
Call Statistic		5.	<u>61.6.212.142:1720</u>	mtm	6732000,6732111,6732222,6732333	
Pre-Granted GW		6.	<u>218.186.154.185:1720</u>	sulaiman	65100,65200,65300,65400	
		7.	<u>61.221.187.93:1720</u>	Uniphone	101756,804	
RAS Port			, ,	BACK	REXT.	

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

	Security Policy(Check IP Address or H323 ID or MAC Address or All,Maximum Policy = 50)			
Gatekeeper Function	Item	Gateway IP Address	H323 ID	MAC Address
Register Gateway Security Policy	ADD			
CDR	DELET	DELETE Security Policy From To		
Realtime CDR				
Call Statistic	⊖ Enable Security Policy ⊙ Disable Security Policy			
Pre-Granted GW				
RAS Port	⊙ Allow Policy ⊖ Block Policy			
	APPLY			

- 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)

ail Record(Maximum CDR = 250 Current CDR = 234 Total Page = 24 Curr

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

Gatekeeper Function	No.	Caller IP Address	Callee IP Address	Calling Number	Caller Number	Duration(Second)	Call Date
Register Gateway	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
Security Policy	3.	202.132.196.190	83.70.203.184	353400	0000	11	2006/4/8 0:13:40
CDR Realtime CDR	4.	61.6.205.218	61.6.212.27	6732222	6732777	68	2006/1/25 16:25:45
Call Statistic	5.	61.6.205.218	218.186.154.185	65400	6732777	82	2006/1/25 15:17:16
Pre-Granted GW	6.	61.6.205.218	218.186.154.185	65400	6732777	509	2006/1/25 15:12:15
RAS POR	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
				BACK N	EXT		

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

	Top 20 by:					
Gatekeeper Function	Duration	✓ Select				
egister Gateway	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
ecurity Policy	2	61.6.201.49	61.221.187.93	101756	6732888	942
DR	3	61.6.205.218	218.186.154.185	65400	6732777	509
ealtime CDR	4	61.6.195.15	61.6.201.49	6732888	6732111	358
all Statistic	5	220.130.54.184	219.137.246.57	10	900	308
	6	61.6.192.61	218.186.154.185	65400		286
re-Granted GW	7	220.130.54.184	219.137.246.57	10	900	273
AS Port	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.

	Pre-Granted Gateway(Current Pre-Granted Gateway = 0,Maximum Pre-Granted Gateway = 20)			
Gatekeeper Function	Item	Gateway IP Address	Phone Number	
Register Gateway		,		
Security Policy	ADD			
CDR	DELETE Pre-Granted	Gateway From To	1	
Realtime CDR]	
Call Statistic	APPLY			
Pre-Granted GW				
RAS Port				

- **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.

	RAS Port Setting(You can set up to 5 RAS po	prt)
Gatekeeper Function	RAS Port1	1810
Register Gateway Security Policy	RAS Port2	1820
CDR	RAS Port3	0
Realtime CDR	RAS Port4	0
Call Statistic Pre-Granted GW	RAS Port5	0
RAS Port		APPLY

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]

4x4 Configuration:		
Default Mode	O PSTN Mode O VoIP Mode	
Switch Key	⊙ * ⊖ **	

4x4 PSTN Route:

	PSTN Number
1.	
2.	

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.

	System Administration
Management	Save Configuration:
Save Configuration	Save current system configuration.
Access Control	Access Control: Set system administrator username and password.
Set to default	Set to Default
Backup/Restore Configuration File	Set to default configuration.
System Information	Backup/Restore Configuration file: Backup current configuration to PC/Restore system configuration from PC backup file.
SNTP Setting	System Information:
Syslog setting	Display current system information.
Capture packet	SNTP Setting: SNTP parameter setting.
	Syslog Setting: Syslog parameter setting.

4.6.1 Save Configuration and Reboot

Management]
Save Configuration	
Access Control	
Set to default	
Backup/Restore Configuration File	
System Information	
SNTP Setting	
Syslog setting	-
Capture packet	

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

Administrator Username	admin	
Administrator Password		
Confirm Password		
Guest Username	guest	
Guest Password		
Confirm Guest Password		

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

	Set to Default
Management	All configuration will be set to default setting!
Save Configuration	
Access Control	Apply
Set to default	
Backup/Restore Configuration File	
System Information	
SNTP Setting	
Syslog setting	
Capture packet	

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

	Backup System Configuration to file		
Management	Restore System Configuration from file:		
Save Configuration			
Access Control	Apply		
Set to default			
Backup/Restore Configuration File			
System Information			
SNTP Setting			
Syslog setting			
Capture packet			

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

	System Information :	
Management		
Save Configuration	Software Version	2.7.5
	WAN Type	PPPoE OK
Access Control	WAN MAC Address	00-0f-fd-70-01-23
Set to default	LAN MAC Address	00-0f-fd-70-01-24
	VoIP Status	SIP Direct Mode
Backup/Restore	VoIP Codec	G723.1
Configuration File	Phone Interface	2FXS+2FXO
System Information	Current system time	2006/5/12 16:09:05
SNTP Setting		
Syslog setting		
Capture packet		
Click System Inform	ation Display to open the Online Statu	s page. In the example, on the

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,

		Contra rotico, terror accontioning - toruta	
		(GMT +02:00) Bucharest	
		(GMT +02:00) Chairo	
Monogoment		(GMT +02:00) Helsinki, Riga, Tallinn	
Management		(GMT +02:00) Athens, Istanbul, Minsk	
Save Configuration	Time Zone Setting	(GMT +02:00) Jerusalem	Select
	Time Zone Sealing	(GMT +02:00) Harare, Pretoria	
Access Control		(GMT +03:00) Moscow, St., Petersburg	
Cat to default		(GMT +03:00) Volgograd	
Set to default		(GMT +03:00) Bahgdad, Kuwait, Riyadh	
Backup/Restore		(GMT +03:00) Nairobi	
Configuration File		(GMT +04:00) Abu Dhabi, Muscat	
		(GMT +04:00) Baku, Tbilisi	
System		(GMT +05:00) Ekaterinburg	
Information		(GMT +06:00) Astana, Almaty, Dhaka	
CNITD Catting		(GMT +06:00) Colombo	
SIMP Setting		(GMT +07:00) Bangkok, Hanoi Jakarta	

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.

Management		
Caus Configuration		(GMT ±08:00) Beijing Chongging
Save Configuration	Time Zone Setting	Charl 100.00) Defining, enongqing
Access Control		
Set to default		
Backup/Restore Configuration File		
System Information		
SNTP Setting		

4.6.7 Syslog Setting Function

	Syslog Server Configuration:					
Management	Sys	log Server Setting				
Save Configuration	Syslog is a method to collect messages fr	Syslog is a method to collect messages from devices to a server running a syslog daemon. Logging				
Access Control	their log messages to a SYSLOG service.	their log messages to a SYSLOG service.				
Set to default	The Syslog messages including CDR(Call Detail Record) and system parameters. (Note: Default Syslog port: 514)					
Backup /Bostoro						
Configuration File	Sy	rslog Server Data				
	Syslog Server IP address	Syslog Server IP address				
System Information	Syslog Server Port 514					
SNTP Setting						
Syslog setting		Apply				

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

	System Administration Main Menu Reboot
Management Save Configuration Access Control Set to default Backup/Restore Configuration File System Information SNTP Setting Syslog setting Capture packet	To troubleshoot what is going on on the network level, you can generate PCAP files on this page. These files can be read with Ethereal network tool. Press the start button to start recording, and press the stop button to stop.Please remember that the data stored in a 15KB buffer and that the recording may have a negative impact on the phone's performance. Start Stop Click <u>here</u> to save the current pcap trace. (0 packets, 0 octets, duration 0 seconds)

Use "Capturer Packets" to record Gateway packets. You can start and stop the capture then save the file to PC Use the Ethereal Tool (<u>www.ethereal.com</u>) 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):



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



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 <u>331 Please specify the password.</u> [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 51 Socket closed. 226 File send OK. [3] Socket closed. Jpgrade(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.

System Administration Main Menu Reboot					
	Set to Default				
Management	All configuration will be act to default actting!				
Save Configuration	All configuration will be set to default setting:				
Access Control					
Set to default	Apply				
Backup/Restore Configuration File					
System Information					
SNTP Setting					
Syslog setting					
Capture packet					

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: <u>sales@soundwin.com</u> Web Site: <u>http://www.soundwin.com</u> 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 VolPBuster

VolPBuster Service Using Soundwin VolP 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.

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

Basic Configuration				
Vo Por)IP Pro t Numl	tocol Setting SIP 💉 Sele	ct git) :	
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.voipbuster.com/5060				
SIP Proxy Server	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

• VolPBuster Provides Free Land Line (Fixed Line) Calls

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

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.

1	G.125 Baridwidtri	🕛 τόκαρς 🕑 τζκάρς 🔘 τύκαρς 💛 όκαρς
Network Setup	G.729 Bandwidth	○ 40kbps ○ 24kbps ○ 19kbps ○ 16kbps ○ 15kbps ⊙ 14kbps
WAN Setting	FXO Transmit Hybrid	Mode 0 ○ Mode 1 ○ Mode 2
LAN Setting	Dial Complete Tone	⊙ Enable ○ Disable
Virtual Server	IP TOS	O Enable ⊙ Disable
Dynamic DNS	Ringer Voltage Threshold	⊙ Low ○ Medium ○ High
Network Management	FXS Battery Reversal Generation	O Enable ⊙ Disable
VoIP Setup	Answer Supervision	⊙ Disable ○ Battery Reversal Detection ○ Voice Detection
VoIP Basic	Line Silence Disconnect	O Enable ⊙ Disable
Dialing Plan	FXO Answer Delay Time	0 msec(from 0 to 8000 msec)
Advance Setting		
Hot Line Setting	Apply	
Port Status		

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.

0 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

Network Setup	VolP Basic Configuration					
WAN Setting		VolP Protocol Setting SIP VolP Select				
LAN Setting						
Virtual Server		Port Number / Password Setting(MAX 20 digit) :				
Dynamic DNS	No	No Number Per Account Pactword				
Network Management				7.000 unit		
VolB Setup	1	33670021		33670021	•••••	
Voli Setup	2					
Vuir Bosic	2					
Advance Setting	3					
Hot Line Setting	4					
Hot Line Setting						
Network Setup						
WAN Setting				SIP Proxy Setting :		
LAN Setting		Domain/Realm 🛛	ervice.sip	.com		
Virtual Server		54	ervice.sip	.com/5060		
Dupamic DNS		SIP Proxy Server				
Network Management		Deviator Interval		use net2pl	hone	
Network Management		(seconds)		900		
VoIP Setup		SIP Authentication				
VoIP Basic		Outbound Proxy		0.0.0/0		
Dialing Plan		Server		L		

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.

	Ădv	ance Setup		·	<u>Main</u> Save	<u>Menu</u> <u>Reboot</u> Configuration	
work Setup	Outgo digit)	ing Dial Plan:(Maximun	50 entri	es,Maximun le	ngth of Prefix Digits is 16 o	digit,Maximun length of	number is 20
tting	Item	Outgoing no.	Length of Number	Delete Len	Add digit no.	Destination IP/DNS	Destination Port
Server	1	х	2~15	0	810	service.sip.com	5060
ic DNS							
k Management			~				
oIP Setup		ELETE Outbound Dial	Plan <mark>Fro</mark>	m To			

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

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

	Ααν	ance Setu	þ		<u>Ma</u> Sa	in Menu <u>Reboot</u> ve Configuration	
Network Setup	Outgoi digit)	ng Dial Plan:(Maximu	n 50 entri	es,Maximun le	ngth of Prefix Digits is 1	5 digit,Maximun length of	number is 20
N Setting	Item	Outgoing no.	Length of Number	Delete Len	Add digit no.	Destination IP/DNS	Destination Port
ual Server	1	86x	3 ~ 15	0	810	service.sip.com	5060
amic DNS							
work Management			~				
VoIP Setup	DE	ELETE Outbound Dia	l Plan <mark>Fro</mark>	m To			

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

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

work Setup	Outgo digit)	ing Dial Plan:(Maximun	50 entrie	es,Maximun le	ngth of Prefix Digits is 16	digit,Maximun length of i	number is 20
etting			Length				Destination
tting	Item	Outgoing no.	of Number	Delete Len	Add digit no.	Destination IP/DNS	Port
Server	1	999	3~3	3	810861111222333	service.sip.com	5060
iic DNS							
rk Management			~				
olP Setup		ELETE Outbound Dial	Plan <mark>Fro</mark>	m To			

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

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 -	Orange-	Purple -	Green-	Purple -	Brown-	Purple -	Gray-	Purple-
Purple	Blue	Purple	Orange	Purple	Green	Purple	Brown	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).





G SB800 / S1600 / S2400 Series module extension

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	Туре
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 $_$



Triangle Mark

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.

S2400(24FXS) SIP Re9 Fail	

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



Gateway_1

Part 1: SVR Setting

Network Setup WAN Setting	VoIP Basic Configurat	ion Port Number Setting(MAX 20 digit) :
LAN Setting	No.	Port Number
Virtual Server	1	
Dynamic DNS	2	200
Network Management	3	300
	4 <	400
VoIP Setup		SIP Hunting Table :
Dialing Plan	No.	Hunting Member
Advance Setting	1	🗹 Port 1 🔲 Port 2 🔲 Port 3 🔲 Port 4
Hot Line Setting	2	Port 1 🗹 Port 2 🗌 Port 3 🔲 Port 4
Port Status	3	Port 1 🗌 Port 2 🗹 Port 3 🔲 Port 4
Traffic Monitor	4	🔲 Port 1 🔲 Port 2 🔲 Port 3 🗹 Port 4

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.

System Configuration	
Advance Setup SIP Proxy Server System Administration	Advance Setup: Let you configure advance VoIP gateway features. SIP Proxy Server Setting: Let you configure SIP Proxy Server parameters. System Administration: View system information and save system configuration.
82004 Telephony Solution	

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.

	Proxy Parameter Setting
SIP Proxy Function	Pagistar Expired Time/seconds 900
Proxy Parameter	Register Expired Time(seconds)
Authentication	SIP Server Part
Register UA	
Real Time CDR	Apply
CDR	
Call Statistic	

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.

SIP Proxy Function	📀 Enable	MD5 authentication E	nable SIP MD5 authentication	
oxy Parameter	O Disable	e MD5 authentication)	visable SIP MD5 authentication	
thentication			Apply	
gister UA				
al Time CDR	Username/P	assword MD5 authentication	::(Max. 250 username/password,I	Max. username/passwor
DR	Item	Username	Password	Operation
ll Statistic		101	101	operation
	2	102	102	_
	3	103	103	
	4	104	104	
	5	11001	11001	
	6	11002	11002	
	6	11002 9404	11002 9404	

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

Network Setup WAN Setting LAN Setting	VOIP Basic Configu	VoIP F Port Numt	rotocol Setting SP	 Select (MAX 20 digit) : 	
Virtual Server	No.	Port Number	Password	Register Status	Reason
Network Management	1 101	$\supset \triangleleft$	•••	Success	OK
	2 102			Success	ОК
VoIP Setup	Use Public /	Account (PORT 1)	O e	nable 💿 Disable	
VoIP Basic			SIP Hunting Table	:	
Dialing Plan	No.		Hunting Mem	ber	
Advance Setting Hot Line Setting	1	1 Port 1 Port 2			
Port Status	2 Port 1 🗹 Port 2				

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.

WAN Setting		SIP Proxy Setting :
LAN Setting	Domain/Realm	svr.dyndns.biz
Virtual Server	SIP Proxy Server	svr.dyndns.biz6000
Dynamic DNS	Register Interval(seconds)	900
Network Management	SIP Authentication	⊙ Enable ○ Disable
	Outbound Proxy Server	0.0.0.0
1		
VoIP Setup		NAT Pass Setting:
VoIP Setup VoIP Basic	NAT Pass Method	NAT Pass Setting: O STUN ⓒ Symmetric RTP
VoIP Setup VoIP Basic Dialing Plan	NAT Pass Method STUN Server address	NAT Pass Setting: STUN ③ Symmetric RTP 64.69.76.21
VoIP Setup VoIP Basic Dialing Plan Advance Setting	NAT Pass Method STUN Server address STUN Server port	NAT Pass Setting: O STUN O Symmetric RTP 64.69.76.21 3478
VoIP Setup VoIP Basic Dialing Plan Advance Setting Hot Line Setting	NAT Pass Method STUN Server address STUN Server port	NAT Pass Setting: STUN ③ Symmetric RTP 64.69.76.21 3478
VoIP Setup VoIP Basic Dialing Plan Advance Setting Hot Line Setting Port Status	NAT Pass Method STUN Server address STUN Server port	NAT Pass Setting: STUN Symmetric RTP 64.69.76.21 3478 Apply

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.

Function 1	2					
eter 100	20	200				
Registered Us	er Agent(Maximum	n 250 user age	ent,Current Regis	stered = 4, Total	Page = 1,Current	t Page = 1):
Item.	Call-id	UserInfo	Contact Address	Real Address	Expires	Remaining Time.
1	v64n8658ph61cj49vg74 - 1	501	61.216.34.91.5060	61.216.34.91:5060	900	3587
2	v64n8658ph61cj49vg74 - 2	502	61.216.34.91:5060	61.216.34.91:5060	900	3587
3	v64n8658ph61cj49vg74 - 0	500	61.216.34.91:5060	61.216.34.91:5060	900	3588
	v64n8658ph61cj49vg74	503	61.216.34.91:5060	61.216.34.91:5060	900	3586

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 "VolP 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

Advance Setup					<u>Main Menu</u> <u>Reboot</u> <u>Save Configuration</u>			
Network Setup	Outgoing Dial Plan:(M digit)	aximun 50 ent	ries,Maximun	length of Prefix	Digits is 16 di	igit,Maxin	nun length of numb	
WAN Setting	Item Outgoing no.	Length of Number	Delete Len	Add digit no.	Destination	IP/DNS	Operation	
LAN Setting	120x	3~3	0	None	192.16	8.1.2		
Dynamic DNS							ADD	
Network Management	DELETE Outbo	und Dial Plan	From To	Step	2:Set outg	oing Dia	ling Plan	
Step_1:click button	1					_		
VoIP Setup	Incoming Dial Plan/Ma	avimun 50 ont	rios Maximun	longth of Profix	Digite is 16 di	ait Maxim	up longth of number	
VoIP Basic	digit):	Aximum 50 end	nes,maximun	lengul of Frenx	Digits is to di	git,inaxini	un lengar of numbe	
Dialing Plan Advance Setting	Item Incoming no.	Length of D Number	elete Len	Add Digit no.	Destination tele port	Register to GK	Operation	
Hot Line Setting Port Status							ADD	
Traffic Monitor	DELETE	nd Dial Plan	From To	»				

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.

Advance Setup						<u>Main Menu</u> <u>Reboot</u> <u>Save Configuration</u>			
Network Setup	Outgoi digit)	ng Dial Plan:(Ma)	kimun 50 e	entries,Maximu	in length of Prefix	< Digits is 16 c	ligit,Maxin	nun length of numbe	
WAN Setting	Item	Outgoing no.	Length	of Delete Len	Add digit no.	Destinatio	n IP/DNS	Operation	
LAN Setting	1	10x	3 ~ 3	0	None	192.1	58.1.1		
Virtual Server			-					ADD	
Dynamic DNS									
Network Management	Di		nd Dial Plan	From	То				
Setp_1:Click Button						Setp_2:Setti	ng Dailin	g Plan	
VoIP Setup									
VoIP Basic	Incomi	ng Dial Plan(Max	timun 50 e	ntries,Maximu	n length of Prefix	Digits is 16 d	igit,Maxim	un length of numbe	
Dialing Plan	aigių.								
Advance Setting	Item	Incoming no.	Length of Number	Delete Len	Add Digit no.	Destination tele port	Register to GK	Operation	
Hot Line Setting									
Port Status								ADD	
							-		

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.

<u>Step 3:make call each other</u> 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)

	Advance Set	UP <u>Main Menu</u> <u>Reboot</u> <u>Save Configuration</u>
	Line Out Volume	0 db(from -9 to 8)
Network Setup	FXO Tx Gain	-4 db(from -6 to 6)
WAN Setting	FXO Rx Gain	0 db(from -6 to 6)
LAN Setting	UK PSTN release tone	C Enable Disable
Virtual Server	detection	
Dynamic DNS	Generation	100 msec
Network Management	DTMF tone power	⊙ -7dbm ○ -6dbm ○ -3dbm ○ -1dbm ○ 0dbm ○ +1dbm ○ +3dbm ○
	FXO Transmit Hybrid	⊙ Mode 0 ○ Mode 1 ○ Mode 2
VoIP Setup	FXO Ringer Voltage Threshold	⊙ Low ○ Medium ○ High
VoIP Basic	FXO Ringer Voltage Filter	⊙ Disable ○ Enable
Dialing Plan	FXO Answer Supervision	● Disable ○ Battery Reversal Detection ○ Voice Detection
	Line Silence Disconnect	💿 Enable 🔿 Disable
Auvance Setting	FXO Answer Delay Time	0 msec(from 0 to 8000 msec)
Hot Line Setting	FXO Answer Mode	Ringing Answer Connectting Answer No Answer
Port Status		
	Apply	

• 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.

