Voip Gateway

F X S

F X O

S I P

:: TT211 (2 ports)
:: TT422 (4 ports)
:: TT844 (8 ports)

Operation Manual

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Operation Manual V3.2T

Contents

1. Introduction

Product Overview

The stand-alone VoIP Gateway carries both voice and facsimile over the IP network. It supports SIP industry standard call control protocol to be compatible with free registration services or VoIP service providers' systems. It works in two different modes: UA (User Agent) or Server. As a standard user agent, it is compatible to all well-known Soft Switches and SIP proxy servers. While running the optional server software, the gateway can be configured to establish a private VoIP network over the Internet without a 3rd party SIP Proxy Server.

There are 3 types of gateways in the same series: 2 ports, 4 ports and 8 ports (voice ports, FXS and/or FXO). The gateway can be seamlessly integrated to existing network by connecting to a phone set, PBX, key telephone system, fax machine or PSTN line. With only a broadband connection such as ADSL bridge/router, Cable Modem or leased line router, it allows you to gain access to voice and fax services over the IP in order to reduce the cost of international and long distance calls.

In addition, the in-built 4 ports Ethernet switch supports comprehensive Internet gateway functions to accommodate other PCs or IP devices to share the same broadband stream. QoS function allows voice and data traffic to flow through where voice traffic is transmitted in the highest priority. With TOS bit enabled, it guarantees voice packets to have first priority to pass through a TOS enabled router.

With the support of DDNS, it makes the gateway reachable by its domain name where the ISP dynamically assigns the IP address. It helps users to host a web site or mail server in a PPPoE or DHCP network. By enabling the CDR function & setting up a simple server, administrators are allowed to log in and view all call records such as call duration, time and date of calls and latency.

The gateway can be assigned with a fixed IP address or by DHCP, PPPoE. It adopts the G.711, G.726, G.729A or G.723.1 voice compression format to save the network bandwidth while providing real-time and toll quality voice. In addition, in the event that the power supply fails or Internet connection is lost, the gateway can automatically divert the FXS end to the PSTN network on the FXO port so users can still use the conventional PSTN line to make calls. This feature is especially useful while dialing emergency calls (i.e. 911).

Hardware Description

TT 211 ports gateway model:

TT211S (2FXS) / TT211O (2FXO) / TT211SO (1FXS 1FXO)

Front Panel
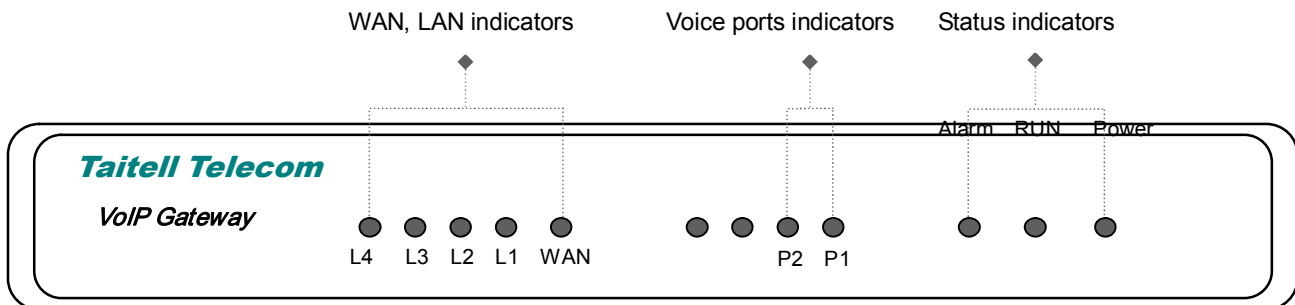
WAN, LAN indicators   Voice ports indicators   Status indicators

Alarm   RUN   Power

Taitell Telecom

VoIP Gateway

L4   L3   L2   L1   WAN   P2   P1

- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- Voice ports indicators: Indicate connection and activity on the port 1 – 2.
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ When starting up the system, the Alarm, Run, and Power indicators will light up. After about 40 seconds, the Alarm indicator will go off, the Run indicator will blink green, and the Power indicator will stay green (under normal operating conditions). If the Alarm indicator continues to blink, then the system is attempting to connect with your ISP and has yet to obtain an IP address.

✓ Once the WAN is connected, the WAN indicator will light up green and, if data is being transmitted over the Internet, the indicator blinks green and orange.

To restore factory default settings (IP address, User's Name, Password):
    (1)   Disconnect the power plug.
    (2)   Press and hold the reset button.
    (3)   Reconnect the power plug while pressing down on the reset button.
    (4)   Release the reset button after 6 seconds. Factory settings will be restored.

Model Description

TT211 S: P1-P2 stand for Phone1-Phone2. Connect to your analog telephone.

TT211 O: P1-P2 stand for Line1-Line2. Connect to your original telephone line on the wall jack with RJ-11 cable.
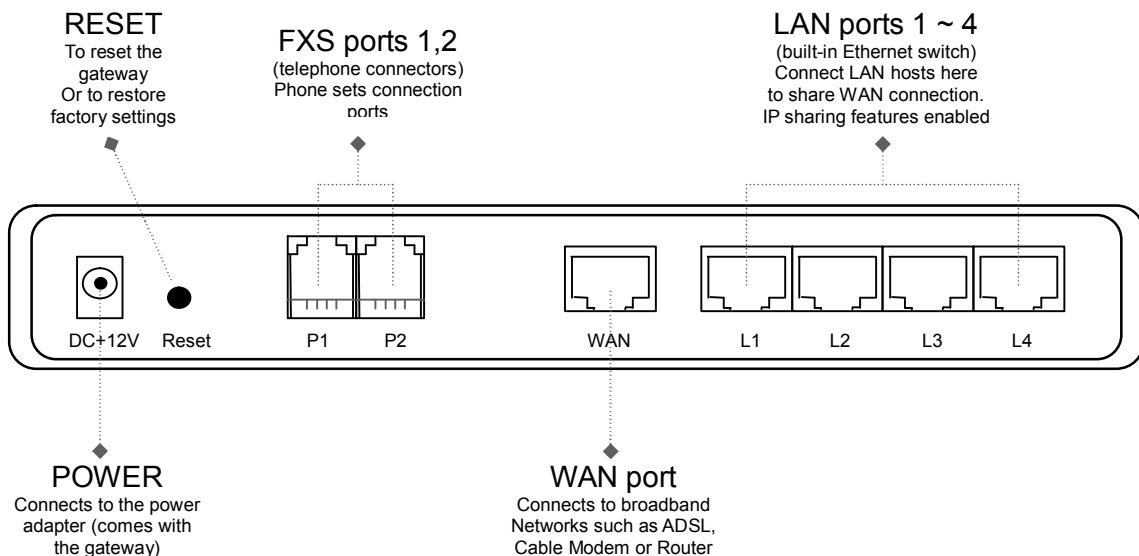
TT211 SO: P1 stand for Phone1 and P2 stand for Line1. Phone port is connected to your analog telephone, and Line port is connected to your original telephone line on the wall jack with RJ-11 cable. P1 will be relayed to P2 for emergency calls before the power is connected or in the occasion of a power failure.

WARNING: DO NOT (1) connect the phone ports to each other (FXS to FXS) or (2) connect any phone port directly to a PSTN line (FXS to PSTN) or to an internal PBX line (FXS to PBX extension). Doing so may damage your VoIP gateway.

Rear Panel

TT 211 S Model (2 FXS ports)

RESET
To reset the gateway
Or to restore factory settings

FXS ports 1,2
(telephone connectors)
Phone sets connection ports

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P1   P2   WAN   L1   L2   L3   L4

POWER
Connects to the power adapter (comes with the gateway)

WAN port
Connects to broadband Networks such as ADSL, Cable Modem or Router

TT211 O Model (2 FXO ports)

RESET
To reset the
gateway
Or to restore
factory settings

FXO ports 1 ~ 2
(PSTN line connectors)
Connect to PSTN lines

LAN ports 1 ~ 4
(Built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P2   WAN   L1   L2   L3   L4

POWER
Connects to the power
adapter (comes with
the gateway)

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

TT211 SO Model (1 FXO and 1 FXS ports)

RESET
To reset the
gateway
Or to restore
factory settings

FXS port: 1
FXO port: 2
FXS connects to phone set;
FXO connect to PSTN line

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P1   P2   WAN   L1   L2   L3   L4

POWER
Connect to the power
adapter (comes with
the gateway)

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

TT422 ports gateway model:

TT422S (4FXS) / TT422 O (4FXO) / TT422 SO (2FXS 2FXO)

Front Panel
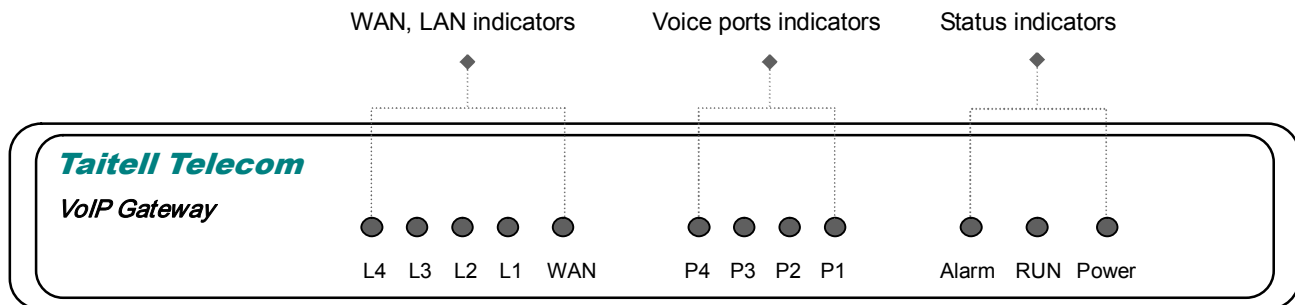
WAN, LAN indicators   Voice ports indicators   Status indicators

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VoIP Gateway

L4 L3 L2 L1 WAN   P4 P3 P2 P1   Alarm   RUN Power

- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- Voice ports indicators: Indicate connection and activity on the port 1 – 4.
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ When starting up the system, the Alarm, Run, and Power indicators will light up. After about 40 seconds, the Alarm indicator will go off, the Run indicator will blink green, and the Power indicator will stay green (under normal operating conditions). If the Alarm indicator continues to blink, then the system is attempting to connect with your ISP and has yet to obtain an IP address.

✓ Once the WAN is connected, the WAN indicator will light up green and, if data is being transmitted over the Internet, the indicator blinks green and orange.

To restore factory default settings (IP address, User's Name, Password):
    (1)   Disconnect the power plug.
    (2)   Press and hold the reset button.
    (3)   Reconnect the power plug while pressing down on the reset button.
    (4)   Release the reset button after 6 seconds. Factory settings will be restored.

Model Description

TT422 S: P1-P4 stand for Phone1-Phone4. Connect to your analog telephone.

TT422 O: P1-P4 stand for Line1-Line4. Connect to your original telephone line on the wall jack with RJ-11 cable.
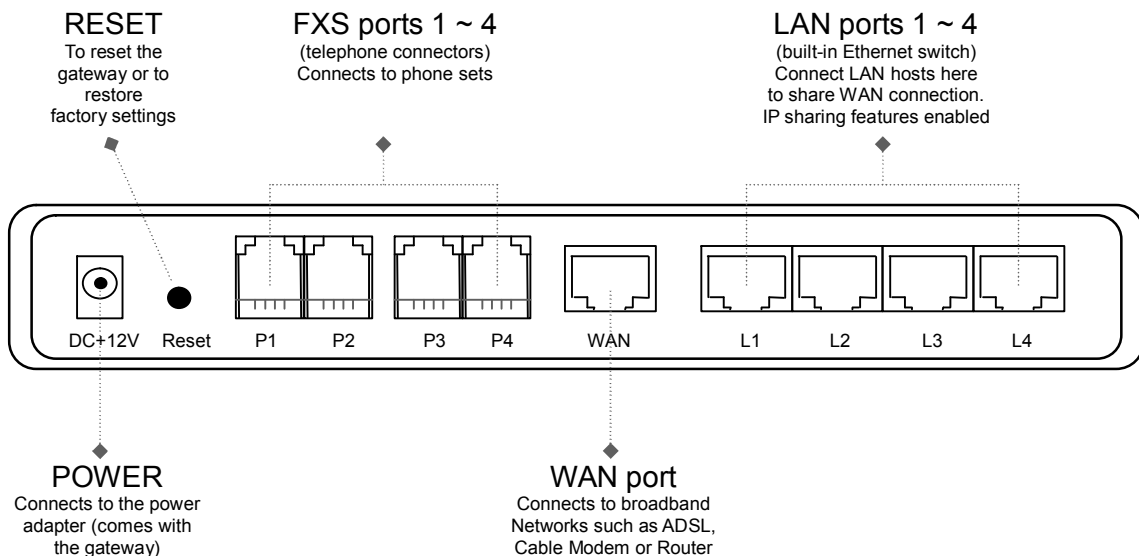
TT422 SO: P1-P2 stand for Phone1-Phone2 and P3-P4 stand for Line1-Line2. Phone ports are connected to your analog telephone, and Line ports are connected to your original telephone line on the wall jack with RJ-11 cable. Each FXS is relayed to each FXO symmetrically before the power is connected or in the occasion of a power failure.
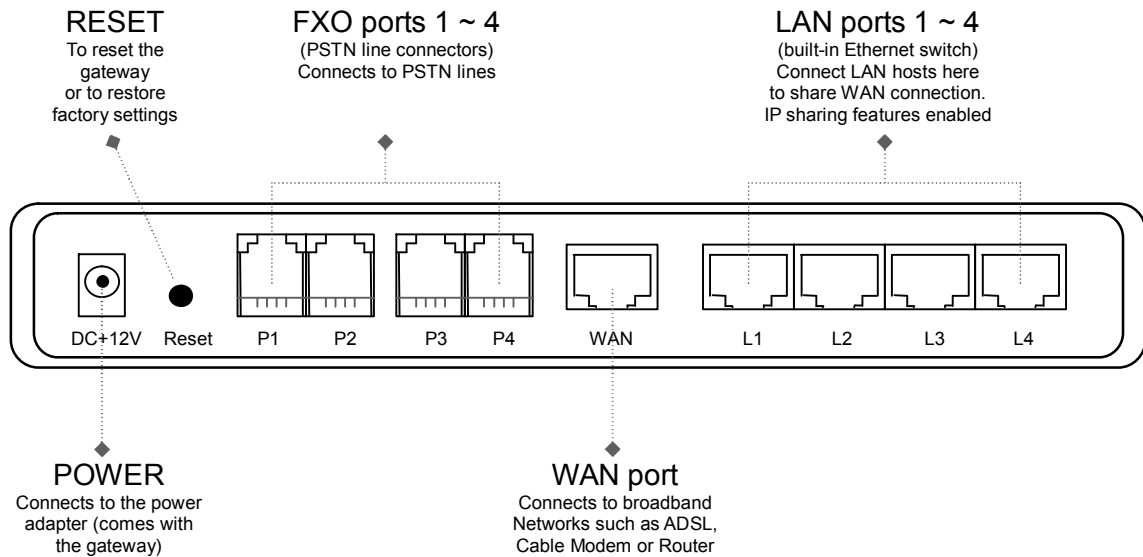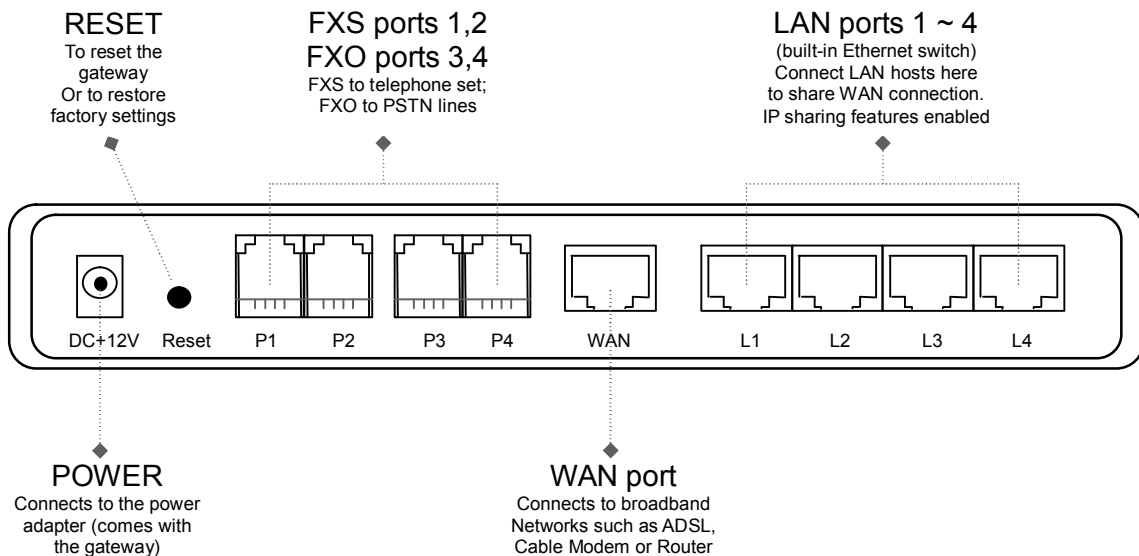
⚠ WARNING: DO NOT (1) connect the phone ports to each other (FXS to FXS) or (2) connect any phone port directly to a PSTN line (FXS to PSTN) or to an internal PBX line (FXS to PBX extension). Doing so may damage your VoIP gateway.
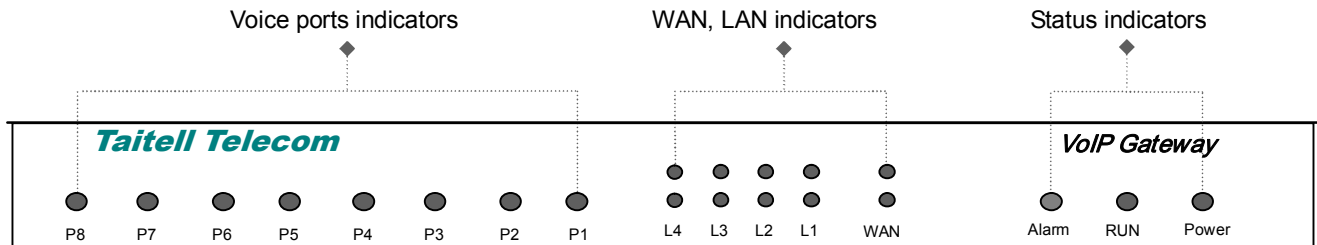
Rear Panel

TT422 S Model (4 FXS ports)

RESET
To reset the
gateway or to
restore
factory settings

FXS ports 1 ~ 4
(telephone connectors)
Connects to phone sets

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P1   P2   P3   P4   WAN   L1   L2   L3   L4

POWER
Connects to the power
adapter (comes with
the gateway)

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

TT422 O Model (4 FXO ports)

RESET
To reset the
gateway
or to restore
factory settings

FXO ports 1 ~ 4
(PSTN line connectors)
Connects to PSTN lines

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P1   P2   P3   P4   WAN   L1   L2   L3   L4

POWER
Connects to the power
adapter (comes with
the gateway)

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

TT422 SO Model (2 FXS and 2 FXO ports)

RESET
To reset the
gateway
Or to restore
factory settings

FXS ports 1,2
FXO ports 3,4
FXS to telephone set;
FXO to PSTN lines

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

DC+12V   Reset   P1   P2   P3   P4   WAN   L1   L2   L3   L4

POWER
Connects to the power
adapter (comes with
the gateway)

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

TT844 ports gateways model:

TT844 (8FXS) / TT844O (8FXO) / TT844SO (4FXS 4FXO)

Front Panel

- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- Voice ports indicators: Indicate connection and activity on the port 1 – 8.
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ When starting up the system, the Alarm, Run, and Power indicators will light up. After about 40 seconds, the Alarm indicator will go off, the Run indicator will blink green, and the Power indicator will stay green (under normal operating conditions). If the Alarm indicator continues to blink, then the system is attempting to connect with your ISP and has yet to obtain an IP address.

✓ Once the WAN is connected, the WAN indicator will light up green and, if data is being transmitted over the Internet, the indicator blinks green and orange.

To restore factory default settings (IP address, User's Name, Password):
(1) Disconnect the power plug.
(2) Press and hold the reset button.
(3) Reconnect the power plug while pressing down on the reset button.
(4) Release the reset button after 6 seconds. Factory settings will be restored.

Model Description

TT844S: P1-P8 stand for Phone1-Phone8. Connect to your analog telephone.

TT844 O: P1-P8 stand for Line1-Line8. Connect to your original telephone line on the wall jack with RJ-11 cable.
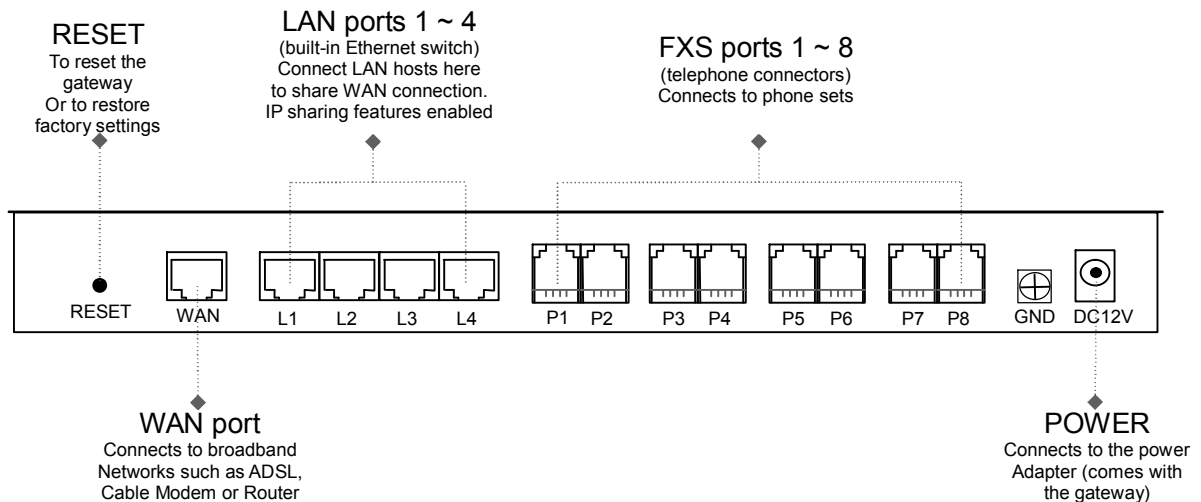
TT844 SO: P1-P4 stand for Phone1-Phone4 and P5-P8 stand for Line5-Line8. Phone ports are connected to your analog telephone, and Line ports are connected to your original telephone line on the wall jack with RJ-11 cable. Each FXS is relayed to each FXO symmetrically before the power is connected or in the occasion of a power failure.

⚠ **WARNING: DO NOT** (1) connect the phone ports to each other (FXS to FXS) or (2) connect any phone port directly to a PSTN line (FXS to PSTN) or to an internal PBX line (FXS to PBX extension). Doing so may damage your VoIP gateway.

Rear Panel

TT844 S Model (8 FXS ports)

RESET
To reset the gateway
Or to restore factory settings

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here to share WAN connection.
IP sharing features enabled

FXS ports 1 ~ 8
(telephone connectors)
Connects to phone sets

RESET   WAN   L1   L2   L3   L4   P1   P2   P3   P4   P5   P6   P7   P8   GND   DC12V

WAN port
Connects to broadband Networks such as ADSL, Cable Modem or Router

POWER
Connects to the power Adapter (comes with the gateway)

TT844 O Model (8 FXO ports)

Erro!

RESET
To reset the
gateway
or to restore
factory settings

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

FXO ports 1 ~ 8
(PSTN line connectors)
Connects to PSTN lines

RESET WAN L1 L2 L3 L4 P1 P2 P3 P4 P5 P6 P7 P8 GND DC12V

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

POWER
Connects to the power
adapter (comes with
the gateway)

TT844 SO Model (4 FXS and 4 FXO ports)

RESET
To reset the
gateway
or to restore
factory settings

LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here
to share WAN connection.
IP sharing features enabled

FXS ports 1 ~ 4
(telephone connectors)
Connects to phone sets

FXO ports 5~8
(PSTN line connectors)
Connects to PSTN lines

RESET WAN L1 L2 L3 L4 P1 P2 P3 P4 P5 P6 P7 P8 GND DC12V

WAN port
Connects to broadband
Networks such as ADSL,
Cable Modem or Router

POWER
Connects to the power
adapter (comes with
the gateway)

2. Installation and Applications

Network Interface

The network interface is divided into 4 basic modes as described below:

- Gateway can be assigned with a Public IP Address
- Gateway can be built under the existing NAT
- Gateway can be assigned with a Public IP address and serves as an IP sharing router.
- Gateway can be assigned with a Public IP address and serves as a bridge

Gateway Assigned with a Public IP Address

The gateway will have a Public IP address for Internet connection regardless of whether it is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dialup / ADSL).

Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	
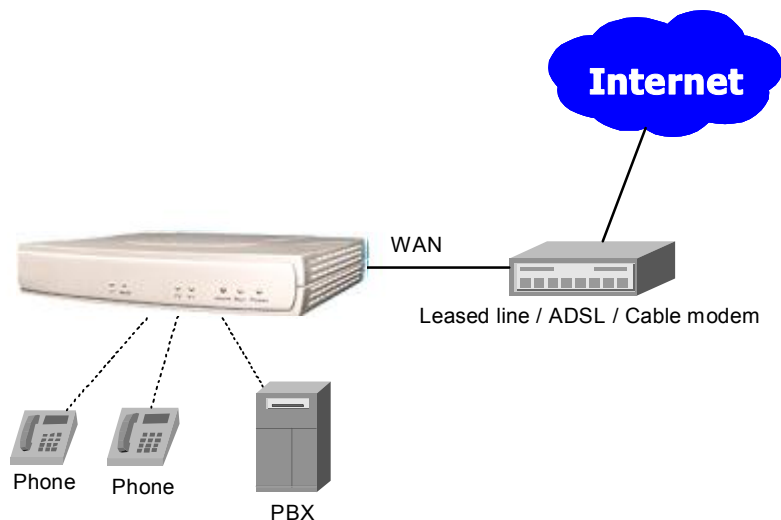
Gateway in a NAT network

The gateway uses a virtual IP address and the IP sharing function of other systems to connect to the Internet.

LAN IP address of IP sharing	Please avoid IP address 192.168.0.1-192.168.8.254 (You may need to change the settings of IP sharing or change SIP series Gateway LAN Port IP address)	
Gateway IP Settings	Set as static IP address, and assign the LAN IP address of the IP sharing to the Default Gateway.	
NAT /STUN Settings	Enable	If the WAN of the IP sharing device has static IP address, then the NAT IP address is set as the Public IP address of the IP sharing.
		If the WAN of the IP sharing device uses a dynamic IP address, then it has to comply with the DDNS settings. When suing NAT, you must enter the URL (Uniform Resource Locator) that is registered to the DDNS server.
DDNS Settings	The WAN of the IP sharing device has a static IP address.	Disabled
	The WAN of the IP sharing device has a dynamic IP address.	Enabled: enter the registered URL (Uniform Resource Locator) into **NAT / DDNS→NAT Public IP**
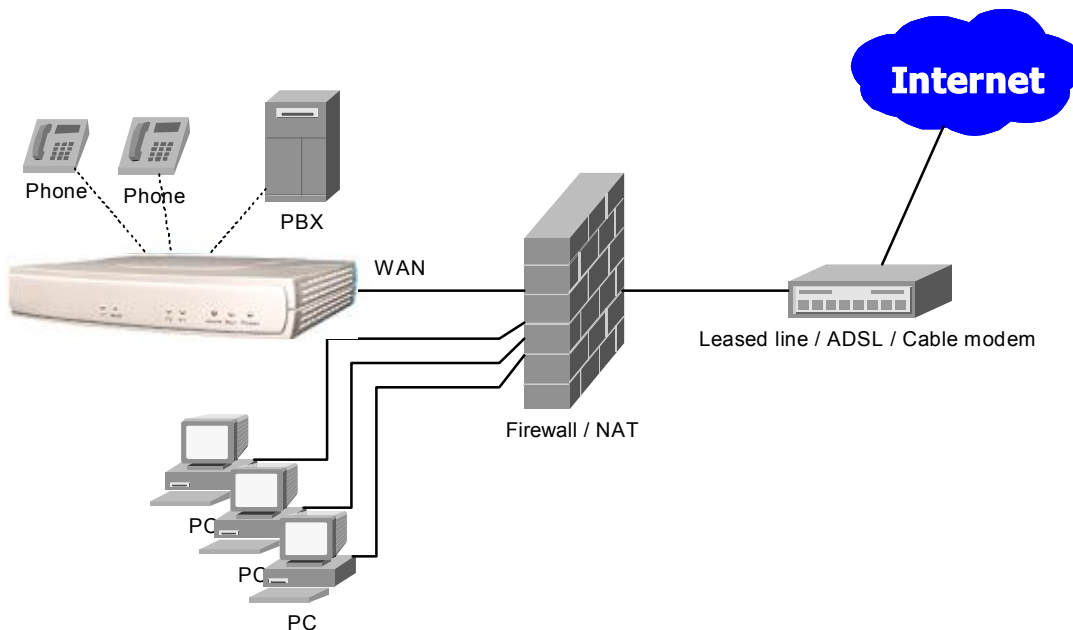
Gateway assigned with a Public IP Address and serving as an IP sharing device

The gateway will have a Public IP address regardless of whether it is a static IP application, DHCP (using a Cable Modem), or PPPoE (To connect to your ADSL account), which can then use the functions of built-in IP sharing function to allow other PCs to be on-line at the same time.

Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE
NAT/STUN Settings	Unnecessary (Disabled)
DDNS Settings	Unnecessary (Disabled)
For settings at PC end, please refer to **IP sharing functions**	PC uses a static IP address ranging from: 192.168.8.1-192.168.8.253 Subnet Mask : 255.255.255.0 Default Gateway : 192.168.8.254
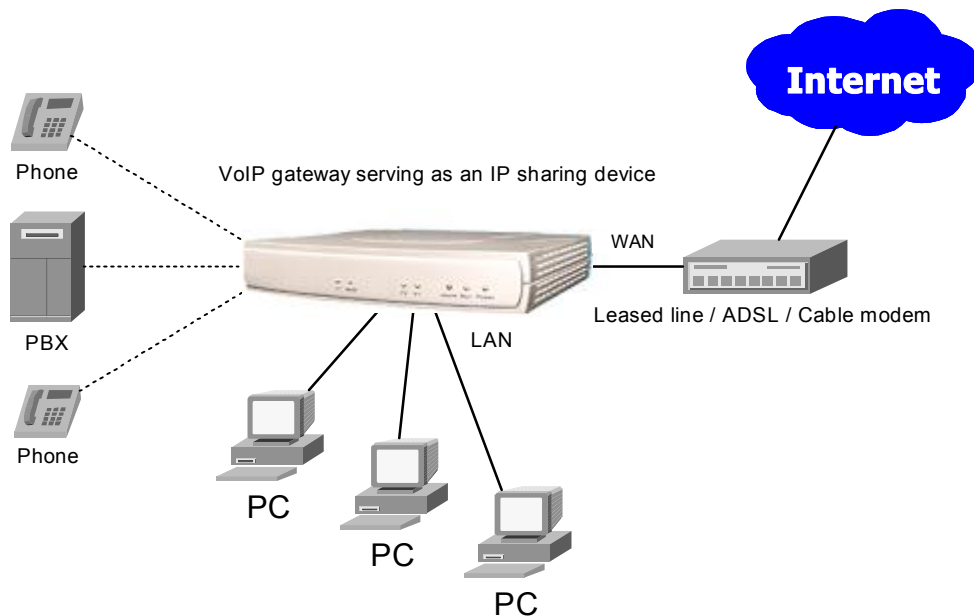
Gateway assigned with a Public IP Address and serving as a bridge

The gateway will have a Public IP address regardless of whether it is a static IP application, DHCP (using a Cable Modem), or PPPoE (To connect to your ADSL account), which can then use the functions of built-in Bridge function to allow a PC to be on-line at the same time.

Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE
NAT/STUN Settings	Unnecessary (Disabled)
DDNS Settings	Unnecessary (Disabled)
For settings at PC end	PC uses the original IP address
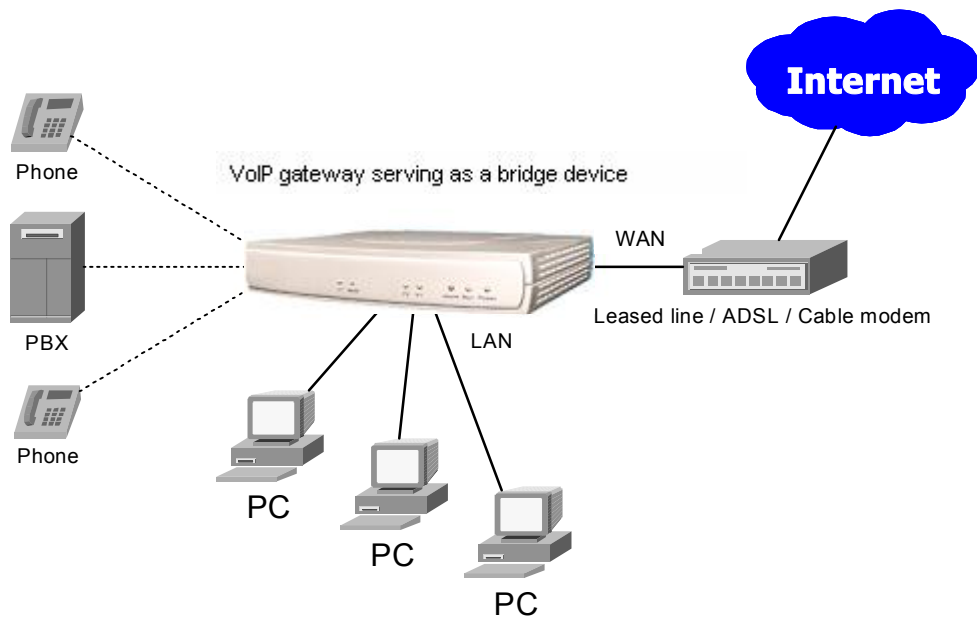
Telephone Interface Description

Example for TT422 S gateway:

TT422 S gateway connecting directly to phone sets

After connecting telephone sets to P1-P4, users can make direct calls, (P1-P4 are FXS interfaces). Each set acts as an independent extension line.
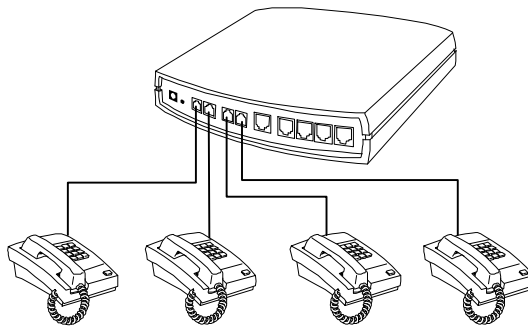
Integrating the TT422 S with PBX

P1-P4 is FXS interfaces, and some of them can be connected to telephone sets for direct calls. Others can be connected to the PBX so other extension lines can make VoIP calls.
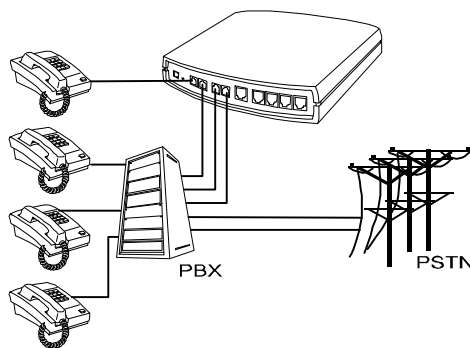
Example for TT422 O gateway:

TT422 O model connecting directly to the Telephone Line of a PSTN

P1-P4 is FXO interfaces and can all be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a user behind the gateway.
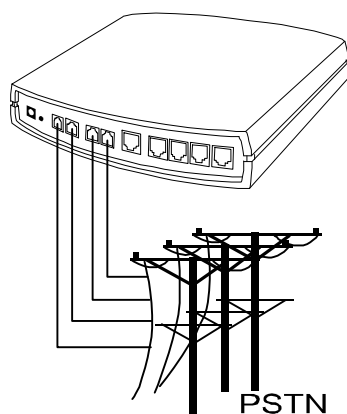
Integrating the TT422 O with PBX

P1-P4 is FXO interfaces and can be connected with PBX extension lines (exclusively for analog interface, not applicable for digital type).
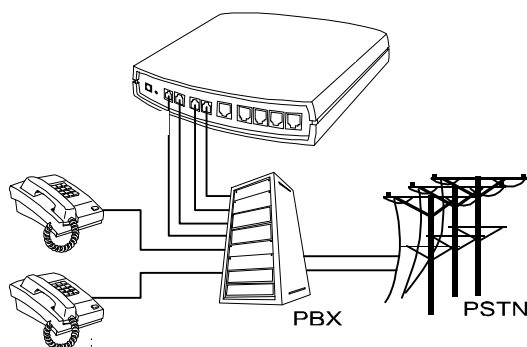
Example for TT422 SO gateway:

P1-P2 is FXS interfaces and can be directly connected to a telephone set for direct calls. P3-P4 is FXO interfaces and can be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a gateway user.
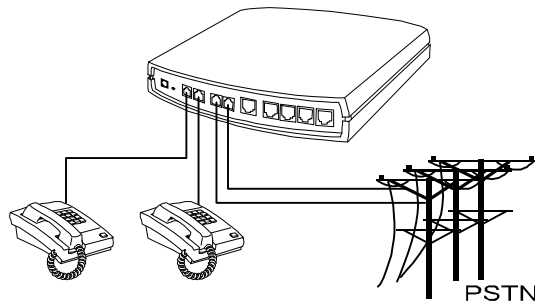
Integrating the TT422 SO with PBX

P1-P2 is FXS interfaces and can be connected to a PBX CO or an analog telephone; P3-P4 is FXO interfaces and can be connected to a PSTN to act as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a gateway user.
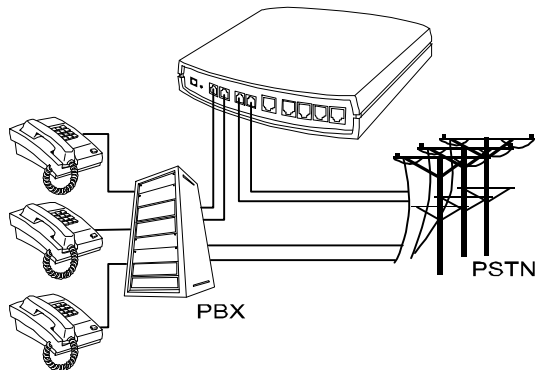
3. Setting the Gateway through IVR

VoIP transmits voice data (packets) via the Internet. One effect of this is that telecommunications quality is closely related to the condition and status of the network environment. If any of the parties involved in VoIP communications has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, excellent telecommunication can only happen when the gateway is connected to the Internet and when the network environment is stable.

Preparation

- Install the gateway according to instructions. Connect the power supply, telephone set, telephone cable, and network cable properly as described in **Installation and Applications**.
- If a static IP is used, confirm the correct IP settings of the WAN Port (IP address, subnet mask, and default gateway). Please contact your local Internet Service Provider (ISP) if you have any questions.
- If using dialup ADSL (PPPoE) for network connection, confirm the dialup account number and password.
- If you intend to operate the gateway under a NAT, the gateway WAN port IP address and LAN port should not use the same range in order to avoid phone failures.

The gateway provides two setting modes:
1. Telephone IVR Configuration Mode
2. Browser Configuration Mode

The IVR provides basic query and setting functions, while the browser provides a full setting function.

IVR (Interactive Voice Response)

The gateway provides convenient IVR functions. Users only need to pick up a handset and enter the function code for the query and setting without using a PC.

NOTE: After finishing the setup, make sure the new settings are saved. This will enable the new settings to take effect after the gateway is restarted.

Instructions

- FXS Port: Connect FXS port to a telephone. To access IVR mode, you should enter factory default code "**** #**". If it requires password, enter ** password #". Character to number conversion information is provide in the **PPPoE Character Conversion Table**. After entering the correct IVR password, you will hear an indication tone after which the gateway is in IVR setup mode. Enter function codes to check or set the gateway configuration. (Please refer to **IVR Function Table** for function codes).

  Example: If your password is "1234", enter ****1234#** so that you are now in IVR setup mode. Next enter a function code to check or configure the gateway. If your password is "admin", enter *****4144534954#**.

- FXO Port: Use extension line to dial the phone number of FXO port. You will hear the instruction "enter value", enter factory default code <u>** #</u> to access IVR mode. If it requires password, enter <u>** password #</u>. Character to number conversion information is provide in the **PPPoE Character Conversion Table**. After entering the correct IVR password, you will hear an indication tone after which the gateway is in IVR setup mode. Enter function codes to check or set the gateway configuration.

- Once the first setting or query has been completed, you will hear a dial tone. Use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

  Example: enter <u>**#</u> (You are now in IVR mode)→ enter <u>101</u> (to query about the current IP address) → the gateway responds with an IP address → you can continue with more settings or queries: enter <u>111</u> (to set a new IP address) → enter <u>192*168*1*2</u> (new IP address).

Save Settings

After completing all of your settings, dial <u>509</u> (Save Settings). Wait for about 3 seconds, you should hear a confirmation tone "1." You can now hang up the phone. Please reboot the gateway to enable new settings.

To inquire about the current gateway's WAN Port IP address

After completing all of your settings, dial <u>101</u>. The gateway will repeat the current WAN Port IP address. If the gateway does not repeat the IP address, this indicates that the gateway is not currently connected to the Internet. Please check to be certain that the cable connection, account number, and password are all correct.

Software Upgrade

IVR provides online upgrades. Once in IVR mode, enter "<u>209</u>" and you will hear "Enter Value". Enter your IP address followed by "#" (i.e.: <u>61*30*25*89#</u>). You will hear a second "Enter Value". Enter the Listen Port Number followed by "#" (i.e.: <u>69#</u>).

NOTE: Please contact your service provider for software upgrade.

IVR Functions Table:

Function Code	Description	Example / Notes
111/101	WAN Port IP address Set/Query	Use function code **114** to select 1 for Static IP connection then setup the IP address
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2.DHCP, 3.PPPoE)	
118	Restart	
311/301	LAN Port IP address Set/Query	
312/302	LAN Port Subnet Mask Set/Query	
131/132	Play/Record greeting message	
133	Saving greeting message	
217/207	Set/Query the gateway web configuration interface port number	
109	Restoring factory default IP address configuration	A static IP address for WAN Port IP：192.168.1.2 Subnet Mask：255.255.255.0 Default Gateway : 192.168.1.254
409	Restoring factory default settings	
509	Save settings	
900	Setting IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Software Upgrade	

IP Configuration Settings of WAN Port

Static IP Settings

NOTE: Complete static IP settings should include a static IP (option 1 under <u>114</u>), IP address (<u>111</u>), Subnet Mask (<u>112</u>), and Default Gateway (<u>113</u>). Please contact your local Internet Service Provider (ISP) if you have any questions.

Function	Command
Select a Static IP	After entering IVR mode, dial 114. After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address, followed by "#". **Example:** If the IP address is 192.168.1.200, dial 192*168*1*200#.
Subnet Mask Settings	After entering IVR mode, dial 112. After hearing "Enter value", enter your subnet mask, followed by "#". **Example:** If the mask value is 255.255.255.0, dial 255*255*255*0#.
Default Gateway Settings	After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address, followed by "#". **Example:** If the default gateway is 192.168.1.254, dial 192*168*1*254#.
Save Settings and Restart	To save settings, dial <u>509</u> (Save Settings). The gateway will save the current settings. After hearing "one", dial <u>118</u> to restart the gateway. Wait for about 40 seconds for the gateway to restart, and then enter <u>101</u> to check whether or not the IP address is retained. If the IP address is not repeated, this indicates that the gateway is not properly connected. Please check to be certain that the cable connection, account, and password are all correct.

Dynamic IP (DHCP) Settings

- After entering IVR mode, dial <u>114</u>.
- You will hear "Enter value", Dial 2 to select DHCP.
- Dial 509 to save settings.
- Dial 118 to reboot the gateway.
- Wait for about 40 seconds for restart, and then enter <u>101</u> to check whether or not the IP address is retained.

ADSL PPPoE Settings

NOTE: Before setting PPPoE, you must have PPPoE account (<u>121</u>) and PPPoE password (<u>122</u>) provided by your local Internet Service Provider (ISP).

Select a PPPoE

- After entering IVR mode, dial <u>114</u>.
- You will hear "Enter value".
- Dial 3 to select PPPoE.

Set PPPoE account

- After entering IVR mode, dial <u>121</u>.
- You will hear "Enter value".
- Enter account number and # (speed up dialing).

Example: If the account is "84943122 @ hinet.net", please enter 08 04 09 04 03 01 02 02 71 48 49 54 45 60 72 54 45 60 #.

NOTE: It is necessary to enter two digits for each character/number; for example, enter "01" for "1" and "11" for "A".

PPPoE Password Setting

- After entering IVR mode, dial <u>122</u>.
- You will hear "Enter value".
- Enter password number and # (speed up dialing).

Example: If the password is "3ttixike", please enter "03 60 60 49 64 49 51 45#".

Save Settings and Restart

- Dial <u>509</u> to save settings.
- Dial <u>118</u> to reboot the gateway.
- Wait for about 40 seconds for restart, and then enter <u>101</u> to check whether or not the IP address is retained. If the IP address is not repeated, this indicates that the gateway is not been properly connected. Please check to be certain that the cable connection, account, or password are all correct.

Record Greeting File

- The gateway allows users to record their incoming call greeting messages when FXO receives an incoming call.
- After entering IVR mode, dial <u>132</u>. After hearing "Enter value", start to record the incoming call greeting message. Simply hang up the phone to end recording.
- After recording, to listen to the recorded message, press <u>131</u>. Press <u>133</u> to save the message.

PPPoE Character Conversion Table

The table below provides a list of PPPoE conversion codes. The first column in each pair of columns lists the number, letter or symbol that you want to enter. The second column in each pair ("Input Key") tells you what code to enter for the corresponding number, letter or symbol.

Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	Input Key
0	00	A	11	a	41	@	71
1	01	B	12	b	42	•	72
2	02	C	13	c	43	!	73
3	03	D	14	d	44	"	74
4	04	E	15	e	45	$	75
5	05	F	16	f	46	%	76
6	06	G	17	g	47	&	77
7	07	H	18	h	48	'	78
8	08	I	19	i	49	(	79
9	09	J	20	j	50	)	80
		K	21	k	51	+	81
		L	22	l	52	,	82
		M	23	m	53	-	83
		N	24	n	54	/	84
		O	25	o	55	:	85
		P	26	p	56	;	86
		Q	27	q	57	<	87
		R	28	r	58	=	88
		S	29	s	59	>	89
		T	30	t	60	?	90
		U	31	u	61	[	91
		V	32	v	62	\	92
		W	33	w	63	]	93
		X	34	x	64	^	94
		Y	35	y	65	_	95
		Z	36	z	66	{	96
								97
						}	98

4. Setting a Gateway with WEB Browser

The VoIP gateway allows users to configure its settings using a web interface (Web UI). You can access the Configuration Menu by opening a web-browser (e.g., Internet Explorer or Netscape Navigator) and entering the factory default LAN IP address: 192.168.8.254. The IP address of the Web UI is same as the default LAN IP noted elsewhere in this user's manual.

You can also use an ordinary telephone, connect it to the gateway, and dial "101" to inquire about the current WAN Port IP address and then use the WAN port to log-in.

Instructions

- Open a Web-Browser (e.g., Explorer, Navigator, Opera, Firefox).
- Enter the LAN port IP address. The default LAN port IP address is: 192.168.8.254.
- The log-in screen below will appear after you connect. The factory default settings for **Login ID** and **Password** are left blank (i.e., no login ID, no password).
- Change the default settings of Administrator's Name, Password and Web UI Login ID, Password in **Advanced Options**.

The gateway does not allow multiple people to configure the gateway simultaneously. Please remember to logout or restart the system if you are not using the web configuration function.

Network Settings (WAN)

The network settings are used to set the gateway's communication ports, and IP configurations, etc.

Network Settings (WAN)		
Current WAN IP Address	192.168.1.2	
Listen Port UDP [1 - 65535]	5060	RTP Starting Port UDP [1 - 65500] 9000
DHCP ○		
Static IP ⦿	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254
PPPoE ○	PPPoE Account	
	PPPoE Password	**********
	Confirm Password	**********
PPTP ○	IP address	
	Subnet mask	
	PPTP Server	
	PPTP ID	
	PPTP Password	**********
	Confirm Password	**********
BigPond Cable ○	User Name	
	BigPond Cable Password	**********
	Confirm Password	**********
	Login Server	
Domain Name Server Assignment	○ Auto ⦿ Manual	
Domain Name Server (Primary) IP	168.95.1.1	
Domain Name Server (Secondary) IP		

- Current WAN IP Address: The IP address of the WAN port.

- Listen Port UDP: It is not necessary to change the protocol of the communication port used by the gateway, unless it conflicts with ports used by another device in your network.

- RTP Starting Port UDP: The initial value of port number of transmitting voice data among gateway(s). Each line requires 2 ports (RTP/RTCP). It is not necessary to change these, unless it conflicts with ports used by another network device.

  For example: If the starting port is 9000, then Line 1 is using 9000(RTP) and 9001(RTCP), and Line 2 is using 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

There are five methods of obtaining a WAN port IP address:
1. DHCP, which means a Dynamic IP (Cable Modem)
2. Static IP
3. PPPoE (dial-up ADSL)
4. PPTP.
5. BigPond (for Australia only)

Methods for using DHCP and PPPoE for obtaining an IP address may vary. If you are not familiar with creating a network connection, please contact your local ISP.

Setting Dynamic IP (DHCP)

Click **DHCP** to obtain a Dynamic IP address, and then click the **Accept** button at the bottom of the screen.

Click **System Operation** to select **Save Settings** and **Restart**. Wait for a while (about 40 seconds), and the system will obtain the related IP address from the DHCP Server.

DHCP ⊙	

NOTE: After the system has obtained a new IP address, if you are using a WAN port to enter the Web Configuration Screen, the new IP address has to be used. The system takes about 40 seconds to restart. The same principle applies to the next two settings.

Setting Static IP

Select **Static IP** and enter the IP address, Subnet Mask and Default Gateway values. Then click the **Accept** button at the bottom of the screen. Save the settings, and then restart the system.

Static IP ⊙	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254

ADSL PPPoE Settings

Select **PPPoE** and enter **Account**, **Password** and Re-enter Password to confirm. Then click the **Accept** button at the bottom. Save the settings, and then restart the system.

	PPPoE Account	
PPPoE ⊙	PPPoE Password	
	Confirm Password	

PPTP

Select **PPTP** and enter **IP Address**, **Subnet mask**, **PPTP Server**, **PPTP ID** and **Password**. Then click the **Accept** button at the bottom. Save the settings, and then restart the system.

	IP address	
	Subnet mask	
	PPTP Server	
PPTP ○	PPTP ID	
	PPTP Password	**********
	Confirm Password	**********

BigPond *(for Australia use only)*

Click **BigPond Cable** and enter User Name and Password. Login Server is optional. Click the **Accept** button at the bottom. Save the settings, and then restart the system.

	User Name	
	BigPond Cable Password	
BigPond Cable ⊙	Confirm Password	
	Login Server	

Domain Name Server

Domain Name Server (DNS): While a gateway is accessing another gateway or a computer with hostname, it will look up the IP address from the DNS provided by your ISP. Normally, the ISP assigns DNS information while negotiating with PPPoE or DHCP. If the DNS is not assigned automatically or the WAN port is assigned a static IP address, the DNS settings must be assigned manually.

Domain Name Server Assignment	○ Auto ⦿ Manual
Domain Name Server (Primary) IP	168.95.1.1
Domain Name Server (Secondary) IP	

Auto: The gateway learns primary and secondary addresses from the ISP's DHCP server or PPPoE server.

Manual : Enter the primary and secondary addresses manually. Please be sure that the IP addresses are correct otherwise the gateway will not be able to access hosts using hostnames instead of IPs.

WAN Link Speed

It is used to choose the WAN Ethernet link speed. The default is Auto.
Please choose the same speed with Router/Switch/Hub, if VoIP gateway connected to Router/Switch/Hub has the link problem.

WAN Link Speed	Auto ▼

Clone MAC

Some Internet Service Providers (ISPs) assign bandwidth via MAC (Media Access Control) addresses. You can click the **Clone** button to copy in a MAC address which will be recognized by your ISP. It is only necessary to fill in the field if it is required by your ISP.

The **Your MAC Address** field will be blank as you log-in via the WAN port.

Click **Restore** to disable this feature.

Factory Default MAC Address	000C2A113051	Restore
Your MAC Address	00055D050012	Clone
Current MAC Address		

Using Phone Book Manager

NOTE: This function is only available in Standard version. Please see Private Network if your gateway is Dual Network version.

Enable Phone Book Manager Server	☐ Clients List		
Share Phone Book to Clients	☐	TTL (Expire time: mins) [1 - 60]	1
Register to Phone Book Manager	☐		
Gateway Name for Phone Book Manager			
Phone Book Manager Login Password		Confirm Password	
Phone Book Manager IP/Domain	192.168.1.1		
Phone Book Manager Server Listen Port [1 -65535]	1690		

Server Settings

- Enable Phone Book Manager Server: This allows other gateway users to register the IP address and telephone number in this Phone Book manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- Share Phone Book to Clients: While this option is enabled and the gateway is performing as a Phone Book Manager, the Phone Book Manager server will append its Phone Book entries for other clients to lookup.
- TTL (Expire time): If the gateway that is controlled by the Phone Book Manager does not report back within the deadline set by TTL, the gateway will be excluded from the user's list. Each gateway should report to the Phone Book Manager once every 30 seconds.

Clients Settings

- Register to Phone Book Manager: Register to the Phone Book Manager
- Gateway Name for Phone Book Manager: The alias registered with the Phone Book Manager.
- Phone Book Manager Login Password: Enter the registered password. If this system is serving as the Phone Book Manager, the set password is also the password used for registering other gateway systems
- Phone Book Manager IP/Domain: Enter the IP address of the Phone Book Manager. It supports URL (Uniform Resource Locator).
- Phone Book Manager Server Listen Port: The protocol communication port of transmitting signals between the Phone Book Manager and other clients. Please confirm whether the setting is the same as that of the **Phone Book Manager**.

NOTE: A gateway is able to be a server and a client at the same time.

Network Settings (LAN)

LAN interface mode	
⊙ Router ○ Bridge	

Network Settings (LAN)			
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0
DHCP Server			
Enable DHCP Server	☑		
IP Pool Starting Address	192.168.8.1	IP Pool Ending Address	192.168.8.250
IP Pool Uses Other Default Gw	☐		
IP Pool Default Gateway	192.168.8.254	IP Pool Subnet mask	255.255.255.0
Lease Time [1 - 9999 hours]	1		
Domain Name Server Assignment	⊙ Auto ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

Port of Web Access from WAN [0=disable, 1 - 65535]	80
Enable Web UI	☑
Enable Telnet Service	☑

LAN interface mode

LAN interface mode
⊙ Router   ○ Bridge

- Router: The system serves as a router with NAT.
- Bridge: The system serves as a bridge between WAN port and LAN port without NAT. (LAN default gateway will still be accessible for configuration).

Network Settings (LAN)

The gateway LAN Port IP and Subnet mask settings.

Example: if the LAN IP address of the Internet Sharing Device is 192.168.8.1, then the gateway's LAN IP address cannot be in the range between 192.168.8.1 ~ 192.168.8.254. You can set 192.168.99.254 for the LAN IP.

Network Settings (LAN)		
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask 255.255.255.0

NOTE: If the gateway is setup behind NAT. The gateway's LAN IP address cannot be within the same range as the Internet Sharing Device, otherwise it would be unable to make or receive calls.

DHCP Settings

DHCP Server	
Enable DHCP Server	☑
IP Pool Starting Address	192.168.8.1 IP Pool Ending Address	192.168.8.250
IP Pool Uses Other Default Gw	☐
IP Pool Default Gateway	192.168.8.254 IP Pool Subnet mask	255.255.255.0
Lease Time [1 - 9999 hours]	1
Domain Name Server Assignment	◉ Auto ○ Man
Domain Name Server (Primary) IP	Domain Name Server (Secondary) IP	

- Enable DHCP Server: Enable or Disable DHCP server service of the gateway.
- IP Pool Starting Address: The first IP address to be assigned to DHCP clients.
- IP Pool Ending Address: The last IP address to be assigned to DHCP clients.
- IP Pool Uses Other Default Gw: Tick the check box to give DHCP client the other default gateway.
- IP Pool Default Gateway: Assign the default gateway and subnet mask to DHCP client.
- IP Pool Subnet mask: Assign the default gateway and subnet mask to DHCP client.
- Lease Time: The valid period of an assigned IP address.
- Domain Name Server Assignment: The DNS information to be assigned to DHCP clients.
  Auto: Assign DNS obtained from WAN port to the DHCP clients.
  Manual: Manually assigns the DNS for DHCP clients.

Accessing Settings

Port of Web Access from WAN [0=disable, 1 - 65535]	80
Enable Web UI	☑
Enable Telnet Service	☑

- Port of Web Access from WAN: Http port for WAN. To change this setting, web configuration must be accessed via the gateway's LAN port.
- Enable Web UI: Unclick the check box to disable WEB access from WAN or LAN while necessary.
- Enable Telnet Service: Unclick the check box to disable Telnet access from WAN or LAN while necessary.

QoS Settings

QoS is that according to the actual bandwidth offered by Internet service, set the appropriate value in WAN QoS filed. Reserve bandwidth is recommended to enable the gateway for other transmission application.

WAN QoS

WAN QoS			
☐ QoS	Downstream Bandwidth	Full ▼	
	Upstream Bandwidth	Full ▼	32	kbps
ToS / DiffServ Settings			
ToS IP Precedence ◉	Signaling Precedence	3 (Flash) ▼	
	Voice Data Precedence	5 (CRITIC / ECP) ▼	
DiffServ (DSCP) ○	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31) ▼	
	Voice Data Value	46 (Expedited Forwarding, EF) ▼	

- QoS (Quality of Service): To set true bandwidth of your Internet connection to ensure sound quality during transmission. (When this function is enabled, voice packets have the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission.) Some models of the VoIP gateway without this function will adjust bandwidth automatically.

- ToS IP Precedence/DiffServ: Voice packets have the highest priority to ensure telecommunication quality; the larger the value you set, the higher the priority.

LAN QoS

Users can allocate bandwidth for various types of network traffic with each LAN port and apply the LAN QoS control function on incoming and outgoing traffic flows, or both of them. The real throughput of incoming rate is depended on downstream bandwidth of WAN QoS and that of outgoing is depended on upstream bandwidth. Incoming and Outgoing flow can not exceed the limit bandwidth rate of total throughput.

LAN QoS				
Enable LAN QoS	☐			
Port	Priority	Flow Control	Incoming Rate Limit	Outgoing Rate Limit
LAN Port 1	Low ▾	☐	Full ▾	Full ▾
LAN Port 2	Low ▾	☐	Full ▾	Full ▾
LAN Port 3	Low ▾	☐	Full ▾	Full ▾
LAN Port 4	Low ▾	☐	Full ▾	Full ▾

- Priority: The gateway can prioritize LAN ports. Data from HIGH priority port are delivered prior to those from LOW priority port while they arrive at the same time.
- Flow Control: Enable or Disable Flow control.
- Incoming Rate Limit: Set the incoming (from WAN to LAN) rate limit of a specific LAN port (can not exceed the real downstream bandwidth).
- Outgoing Rate Limit: Set the outgoing (from LAN to WAN) rate limit of a specific LAN port (can not exceed the real upstream bandwidth).

NAT/DDNS

NAT Traversal

If the gateway is set up behind an Internet sharing device, you can select either the NAT or STUN protocol.

NAT Traversal		
NAT Public IP ☐	NAT IP/Domain	
Enable STUN Client ☐	STUN Server IP / Domain	
	STUN Server Port[1 ~ 65535]	3478
Enable UPnP Control Point ☐		

- NAT Public IP: The IP address used by the gateway should be a private address. Furthermore, users must set the Virtual Server Mapping in the Internet sharing device. (For example, a virtual server is usually defined as a Service Port, and all requests to this port will be redirected to this specified the server's private IP address).

  The default port is listed below:

  Listen Port (UDP): 5060
  RTP Starting Port (UDP): 9000~ (Listen Port used for telephone communication).
  Port of Web Access from WAN (TCP): The number you set in this option on the Network Settings page.
- NAT IP/Domain: Enter the NAT Server IP address (real public IP address of the Internet

sharing device); or enter a true URL (Uniform Resource Locator) when DDNS is used. Please refer to **DDNS** for further information.

NOTE: If you are setting a public IP in this field, it has to be a public IP, otherwise VoIP communication may not be established properly.

- Enable STUN Client: Using the STUN protocol prevents problems with setting the IP sharing function, but some NATs do not support this protocol.

- STUN Server IP/Domain and Port: Enter the STUN server IP address and Listen Port number.

- Enable UPnP Control Point: This variable will enable the gateway's IP traffic to pass through an Internet sharing device. This function only works when the Internet sharing device supports UPnP and has it enabled.

NOTE: The "Status → Current Status" page will show the status of UPnP.DDNS

These settings are only necessary when the gateway is set up behind an Internet sharing device that uses a dynamic IP address and do not support DDNS.

The current system allows users to choose either DynDNS, TZO, 3322.org, PeanutHull or a private server. You will need to apply for an account with Please apply for a user account with DynDNS、 TZO、3322.org、PeanutHull or a private server before you type in the following information.

☐ Register to DDNS	

○ DynDNS DDNS Server		Default
Server Address	members.dyndns.org	
Hostname	dyndns.org	
Login ID		
Password	**********	
Confirm Password	**********	
Behind NAT	☐ Yes	
Custom	☐	

- Register to DDNS: Tick the checkbox to enable DDNS function.
- Server Address: The IP address or URL (Uniform Resource Locator) of the DDNS Server.
- Hostname: The URL of the system (or NAT) – applied by a domain name registration providers. (e.g. www.dyndns.org).
- Login ID and Password: The ID and password are used to log-in to the DDNS server.
- Behind NAT: Tick the checkbox to enable this function only when the gateway is set up behind a NAT device.
- Custom: *Only for DynDNS.* Tick the checkbox if you have a custom hostname in DynDNS.

NOTE: If the gateway is set up under NAT, then enter the hostname into the NAT IP/Domain that is the same with Hostname of DDNS.

Example:

NAT

NAT Traversal	
NAT Public IP ☑	NAT IP/Domain	hostname.ddnsserv.com

DDNS

☑ Register to DDNS		
⊙ **DynDNS DDNS Server**		Default
Hostname	hostname.ddnsserv.com	

Caller ID

FXS Caller ID Generation	⊙ Disable ○ DTMF ○ FSK	
FXO Caller ID Detection	☑	Detection Level [0 ▾]
FSK Caller ID Type	⊙ Bellcore ○ ETSI	
Anonymous Caller ID (CLIR)	☐	
CLIR At Transit In W/O Caller ID	☐	

- FXS Caller ID Generation: Select this option to enable the caller ID display function on FXS ports. When enabled, the caller's phone number will be displayed on your phone set when the call comes through. FSK is preferred in some countries.

- FXO Caller ID Detection: It is to detect Caller ID delivered from PSTN to the FXO port.

- Detection Level: It is the gain volume that could be adjusted while detecting Caller ID.

NOTE: You have to enable "Wait for Caller ID before FXO / Trunk pick up" to ensure Caller ID is detected correctly.

- FSK Caller ID Type: Bellcore is preferred in North America and ETSI is in Europe.

- Anonymous Caller ID (CLIR): When enabled, anyone receiving a call from you will not display your number if the have caller ID.

NOTE: If you register the gateway to a Proxy and you check this options, you may be unable to make a call. This is due to the fact that the VoIP gateway doesn't send the number for authorization.

- CLIP At Transit in W/O Caller ID: When disabled, if the FXO detects Caller ID in a call from PSTN, the gateway will use the detected Caller ID as caller identification when it makes transit in calls; if FXO cannot detect Caller ID in a call from PSTN, the gateway will use "anonymous" as caller identification for transit in calls. When it enabled, the gateway will always use "anonymous" as caller identification for transit in calls.

Transit In Caller ID Strip / Replace

You can change the information of the calling party while making calls to Internet.

Transit In Caller ID Strip / Replace	
Scan code("?" = single digit ; "%" = wildcard)	Substitude

- Scan code: Defines the rule of the Caller IDs detected by FXO. It can be a prefix or a full number.

- Substitude: Defines the changed display info. of the calling party while making calls to Internet by FXO.

- Telephony Settings

Prefix Number Rules

Prefix Number Rules	
Trunk Dial Out Verify	01;00
Trunk Dial Out Replace	190601;190200
Trunk Dial Out Deny	020

- Trunk Dial Out Verify/ Trunk Dial Out Replace: Before FXO dials to PSTN, the gateway will check (verify) the numbers in **Trunk Dial Out Verify** field and change (replace) them with the numbers in **Trunk Dial Out Replace** filed.

Example:
If you transit out with 01907123456, the system will transmit to 190601 907123456. If you transit out with 008621123456 the system will replace it with 190200 8621123456. The maximum digit is 40.

- Trunk Dial Out Deny: The system will deny the call with the leading number filled in this column.

Note: This rule only applies to one-stage dial.

Trunk Incoming Prompt Voice	⦿ Default Greeting ○ Custom Greeting ○ Dial Tone	
Custom Greeting Upload / Backup	[ ] Browse... Upload Backup	Clear Greeting
FXO Hunting VoIP call in option	Caller Indicate Dial-Out ▾ FXO Hunting Default Dial-Out	[ ]
FXO Line VoIP call in option	Caller Indicate Dial-Out ▾	

- Trunk Incoming Prompt Voice: Select the greeting (must use the IVR 132 function to record a voice file). When FXO receives an inbound call (transit in).

- Custom Greeting Upload / Backup

  Browse…: Select the preferred voice file for upload.
  Upload: Upload the voice file. The format must be G.723.1.
  Backup: Download the voice file to your PC. It allows you to share custom voice file among gateways.
  Clear Greeting: Remove the voice file.

- FXO Hunting VoIP call in option: To set FXO dial-out mode by using the default setting or the indicated number to dial out when the VoIP call calls FXO hunting number.

  Caller Indicate Dial-Out: When someone makes a call to this FXO port from Internet, it will dial to PSTN with the number assigned in SIP packet.
  Default Dial-Out: When someone makes a call to this FXO port from Internet, it will dial to PSTN with the number filled in **FXO Hunting Default Dial-Out** field. FXO Hunting Default Dial-Out: To set FXO default dial-out number. This will take effect as FXO Line VoIP call in option is set to Default Dial-Out. When someone makes a call to this FXO port from Internet, it will dial to PSTN with that default number.

- FXO Line VoIP call in option: To set FXO dial-out mode when the VoIP call indicates the FXO extension number.

  Caller Indicate Dial-Out: When someone makes a call to this FXO port from Internet, it will dial to PSTN with the number assigned in SIP packet.
  Default Dial-Out: When someone makes a call to this FXO port from Internet, it will dial to PSTN with the number filled in **FXO Line Default Dial-Out** field.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	Dial-Out Prefix	FXO Line Default Dial-Out	FXS Group	Fax / Modem
1	☑	FXS	☐	[ ]	0			☑	T.38 Fax ▾
2	☑	FXS	☐	[ ]	0			☑	T.38 Fax ▾
3	☑	FXO	☐	[ ]	0	[ ]	[ ]		T.38 Fax ▾
4	☑	FXO	☐	[ ]	0	[ ]	[ ]		T.38 Fax ▾

- Enable: Tick the check box to enable a line. If some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to the line.

Hotline Functions

- FXS port: When the user picks up the phone, the gateway automatically dials assigned hotline number. When in hotline mode, other phone numbers cannot be dialed.

- FXO port: When receiving a call from an outside line, the gateway will divert the call to the assigned hotline number.

- Hot Line No.: Enter the hotline number for an automatic dialing function.

- Warm Line: When the Warm Line function is in use, user can dial a number. Otherwise the system will divert incoming calls from an outside line to the Hot Line Number after a set wait time.

Example:

Assume the assigned hotline for the first FXO port is 701 and the Warm Line (Hot Line Delay) is 5 seconds. If no extension number is dialed within 5 seconds, the call will be automatically diverted to the assigned hotline (ext 701). The system allows users to record a voice prompt (e.g. "please enter an extension number or wait for the operator to connect you") to use in this situation.

Assume the assigned hotline for the second FXO port is 702 and the Warm Line is 0 second. When the FXO port receives a call from an outside line, it will be automatically diverted to extension 702.

- Dial-out Prefix: It is the number dialed automatically by the system when the FXO interface diverts a call to the PSTN by VoIP (2S/4S/8S do not support this function).

- FXO Line Default Dial-Out: Default number that FXO will dial out when it receive an incoming call from VoIP.

Example:

If PBX extension needs to dial "0" to make a PSTN call, and the FXO ports are connected to PBX extension. In this case, the Dial-out prefix should be set to "0".   If the PBX requires some delay time before capturing a line, then the trunk prefix should be set as "0," so that after dialing a 0, it will pause for 1 second before dialing the destination number. Each comma represents a 1 second delay. If more delay time is required, simply add more commas.

NOTE: If a Dial-out prefix is set, the line won't be able to dial to any PBX extension line (FXS interface does not have a trunk prefix function).

- FXS Group: Tick the check box to select group hunting when there is an incoming call and the gateway will automatically assign an unassigned call according to the **Hunting Priority**. If Port 2 does not want to be set as an assigned line to receive any inbound calls, the function can be disabled. Users can also use the Up or Down Key to adjust hunting priority (No setting is required for the FXO interface).

- FAX / Modem: Select the mode to detect if there is a fax tone and transfer the call to a fax line.

    Disable: Stop to detect fax tone automatically.
    T.38 Fax: Use T.38 as the protocol for fax transmission. T.38 is used for better and faster facsimile transmission. It is recommended to enable T.38 to gain better fax quality without setting fax and voice parameter.
    T.30 Fax: Use T.30 as the protocol for fax transmission. It will consume more network resources and will affect transmission quality. The gateway is still able to change the protocol from T.38 to T.30 if the called party uses T.38 for fax transmission.
    T.30 Fax/Modem: Use it as the protocol for transmission of fax/modem over IP network.

T.30 Only: Only use T.30 as the protocol for fax transmission. The gateway won't accept T.38 for fax transmission.

Trunk Hunting Order	Sequential ▼
Enable FXO / Trunk Extension Number ☑	Pick up Line by Dialing Extension Number ☑
Wait for Caller ID before FXO / Trunk pick up	☑
Transit in Busy Tone Limit [0 - 60 s]	3
Ring (Early Media) Time Limit [10 - 600 s]	600
Enable End of Digit Tone	☐
VoIP Call Out Notification	☐
Enable Built-in Call Hold Music	☑
Force Calling Thru PSTN Code	
Trunk Early Media Option	One Way Voice ▼
Early Media Treatment	☑
Loop Current Drop Trigger Time [0=disable, 3 - 30 s]	0
Loop Current Drop Duration [1 - 5 s]	2
Enable ROH	☐
Max. External Call	999

- Trunk Hunting Order: To set FXO dial-out mode when there is an incoming call dialed FXO representative or unassigned call.

  First Idle: The gateway will assign each unassigned call from first FXO port.
  Sequential: The gateway will automatically assign the first unassigned call to the first FXO port. The second FXO port will dial the second unassigned call out. Each line be used.

- Enable FXO/Trunk Extension Number: Selects this function only when FXO is connected to different PBX or PSTN, or under special circumstances. Users are free to call out from a desired channel, if assigned. If you register to a Proxy it MUST be checked.

Description:

Assume a user at Port 1 of the gateway would like to assign Port 4 (FXO) to make a call and **Enable FXO/Trunk Extension Number** is checked. The user can dial 704 22520199 to assign Port 4 to dial out.
If **Enable FXO/Trunk Extension Number** is unchecked, the gateway will select a FXO line automatically to call out. The user can dial 22520199 without adding FXO extension number 703 or 704.

- Pick up Line by Dialing Extension Number: Allows user to dial the extension number first, after hearing the second dial tone, and dial a PSTN number. If you are registered to a Proxy, it MUST be checked.

- Wait for Caller ID before FXO / Trunk pick up: To detect caller ID from FXO port.

- Transit in Busy Tone Limit: The duration gateway plays a busy tone before FXO hook-on. It is to notify the caller from PSTN that this call is finished.

- Ring (Early Media) Time Limit: The timeout to cancel a call when no one answers.

- Enable End of Digit Tone : The gateway will play a "Beep-Beep" tone to notify that the call is in progress.

- VoIP Call Out Notification: The gateway will play a tone to notify that the call is via VoIP.

- Enable Built-in Call Hold Music: The default setting is that when receiving a call hold request, the gateway will play music on hold. Unclick the check box to disable this function while necessary.

- Force Calling Thru PSTN Code: Dial the code to get a PSTN line for dialing out.

Example: If you specify "*33 "and you would like to dial "23456789" via a PSTN, dial "*33 23456789".

- Trunk Early Media Option: Early Media refers to media that is generated prior to connection or answer of a call is established by the called user. It may be unidirectional or bidirectional, and can be generated by the caller, the callee, or both. The gateway supports three early media mechanisms. These mechanisms occur from the moment "200 OK" being sent in response to an "INVITE" message. It can be Both Way Voice, One Way Voice and Ring Back.

NOTE: This function is active when PSTN Answer Detection is enable (Advanced Options\Line Setting).

- Early Media Treatment: If this variable is disabled, the system will send RTP immediately after a connection with a proxy is set up. The default setting is enabled, If communicating with other gateways encounters problems, please disable this function.

- Loop Current Drop Trigger Time: To set the trigger time for FXS drops loop current. A setting of zero is to disable this function. It is used to avoid the line engaged if FXS is connected to PBX.

- Loop Current Drop Duration: To set the drop duration.

- Enable ROH: The gateway will play Receiver Off-Hook tone to notify user of hanging up the phone set if FXS is off-hook for 20 seconds.

- Max. External Call: To control network voice quality according to bandwidth, defines the maximum concurrent Internet call is allowed by the gateway.

FXS Group Hunting/Ring Priority

FXS Group Hunting / Ring Priority			
Hunting / Ring	Hunting ▼	Sequential Ring Time [1 - 100 s]	6
1	Line 1	▽	
2	Line 2	�	

- Hunting/Ring: This variable is able to set FXS group hunting using simultaneous ring or sequential ring.

- Sequential Ring Time: Set the ring time of each port, when sequential ring is chosen.

- Hunting Priority: This variable can be adjusted using the Up and Down arrows.

- SIP

FXS/ FXO Representative Number registers to Proxy:

Assuming that your registered ID and password are individual, the settings should be as above.

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
FXS Representative Number		23425926	☑		account1	**********	**********
FXO Representative Number		1746518	☑		account2	**********	**********
1	FXS	701 Auto	☐	☐		**********	**********
2	FXS	702	☐	☐		**********	**********
3	FXO	703	☐	☐		**********	**********
4	FXO	704	☐	☐		**********	**********

- FXS Representative Number: Register all FXS ports as a hunting group.

- FXO Representative Number: Register all FXO ports as a hunting group. All the grouped FXO ports will be hunted automatically. It is available when you register FXO to Proxy.

- Register: Tick the check box to register to Proxy if selected

- Invite with ID / Account: Tick the check box if SIP server requests authentication.

NOTE: Please ensure that if Proxy Server allows one account for many ports using before using representative number to register.

Each line registers to Proxy independently:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
FXS Representative Number		23425926	☐			**********	**********
FXO Representative Number		1746518	☐			**********	**********
1	FXS	701 [Auto]	☑	☑	account1	**********	**********
2	FXS	702	☑	☑	account2	**********	**********
3	FXO	703	☑	☑	account3	**********	**********
4	FXO	704	☑	☑	account4	**********	**********

As there are various Proxy Server providers, according to RFC standard our company has designed the gateway to be compatible with them. If any registration problem occurs, please consult your Proxy Server Provider.

NOTE: When you register with a Proxy, dialing principles may vary with different Proxy Servers, especially when dialing through a remote end FXO port. Please consult your Internet Telephony Service Provider.

DNS SRV Settings

Use DNS SRV	☐
DNS SRV Auto Prefix	☑
Proxy Fallback Interval [0 - 10800 s]	1800

- Use DNS SRV: The gateway asks for the related IP address of SIP Server from the records of DNS SRV. DNS SRV uses several servers for a single domain for SIP proxy, to move services from host to host and design some hosts as primary servers (the highest priority) for a service and others as backups. If the primary server is not reachable, the gateway will go for backup server, and so forth…
- DNS SRV Auto Prefix: This option tells the gateway to send packet with service type when using DNS SRV.
- Proxy Fallback Interval: Set the preferred Proxy Fallback Interval. After the time expires, the gateway gets back for registration with the primary server.

NOTE: Be sure that your Internet Telephony Service Provider supports DNS SRV. If you fail to make a call, please contact your Internet Telephony Service Provider.

SIP Proxy Server / Soft Switch Settings

☐ Enable Support of SIP Proxy Server / Soft Switch			
	☑ **Enable SIP Proxy 1**		
Proxy Server IP / Domain	192.168.1.1	Proxy Server Port [1 - 65535]	5060
Proxy Server Realm		TTL (Registration interval) [10 - 7200 s]	600
SIP Domain		Use Domain to Register	☐
	☐ **Enable SIP Proxy 2**		
Proxy Server IP / Domain	192.168.1.1	Proxy Server Port [1 - 65535]	5060
Proxy Server Realm		TTL (Registration interval) [10 - 7200 s]	600
SIP Domain		Use Domain to Register	☐
VoIP failure announcement	☐		
Bind Proxy Interval for NAT [0 - 180 s]	0		
Initial Unregister	☐		
Support Message Waiting Indication (MWI)	☐		
MWI Subscribe Interval [0=disable, 60 - 86400 s]	7200		

- Enable Support of SIP Proxy Server / Soft Switch: Tick the check box to enable the functions to inter-work with Proxy Server / Soft Switch. When SIP Proxy 1 and 2 are enabled, the gateway will register to SIP Proxy 2 after all lines have failed to register to SIP Proxy 1.

NOTE: SIP Proxy 2 is a backup system.

- Proxy Server IP/Domain: Enter the Proxy Server IP address or URL (Uniform Resource Locator).

- Proxy Server Port: Enter the Proxy Server listen port number. (The factory default value is 5060).

- Proxy Server Realm: This variable is used for gateway SIP account authentication in a SIP server. In most cases, the gateway can automatically detect your Proxy Server realm. You can leave this option black. However, if your Proxy Server requires you to use a specific realm you can manually enter it here.

- TTL (Registration interval): Enter the desired time interval at which the gateway will report to your Proxy Server.

- SIP Domain: Enter SIP Domain (URI) if required by Proxy Server.

- Use Domain to Register: Tick the check box to make the gateway register with SIP Domain; otherwise the gateway will register to a Proxy with the IP it resolves.

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- VoIP failure announcement: As soon as VoIP call or the registration to proxy server is failed, the gateway will drive IVR system to play out failure announcements for the caller.

- Bind Proxy Interval for NAT: This function is able to keep the binding that exists when the gateway is behind a NAT and Proxy Server is not able to keep the binding.

- Initial Unregister: During the gateway start up, it sends UNREGISTER packet first to release the possible invalid binding on SIP Proxy server.

- Support Message Waiting Indication (MWI): Tick on the check box to enable voice mail function. The system will play a tone to remind user if there are messages in the voice mail.

- MWI Subscribe Interval: The subscribe interval is for the gateway check of the voice mail.

NOTE: If you fail to make a call, please contact your Internet Telephony Service Provider.

OutBound Proxy Settings

An outbound proxy server handles SIP call signaling as a standard Proxy Server would. Further, it receives and transmits phone conversation traffic (media) between two communication parties. This option tells the gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote gateway. This might help VoIP calls to pass through any NAT protected network without additional settings or techniques.

NOTE: Please make sure your Internet Telephony Service Provider supports outbound proxy before enable it.

OutBound Proxy Support	☐		
OutBound Proxy IP / Domain		OutBound Proxy Port [1 - 65535]	5060

Session Timer (RFC 4028) Settings

Session Expiration [0=disable, 10 - 1800]	0	
Session Refresh Request	⊙ UPDATE ○ re-INVITE	
Session Refresher	⊙ UAS ○ UAC	

- Session Expiration: It is used to avoid billing for abnormally dropped calls due to Internet problems. The default is disabled.

- Session Refresh Request: Used to resend UPDATE or re-INVITE requests to the server.

- Session Refresher: Selects which user agent is the session refresher. UAS (User Agent Server) is an originator, and UAC (User Agent Client) is a replier.

P-Asserted (RFC 3325) Settings

Enable P-Asserted	☐
Privacy Type	id

- Enable P-Asserted: It is for caller ID protection.
- Privacy Type: Privacy type is used to disguise the caller ID when queried via an ITSP/Third-Party Assertion.

Other Settings

SIP Message Resend Timer Base [s]	0.5 ▼
Max. Response Time for Invite [1 - 20]	8
Invite URL need 'user=phone'	☑
Reliability of Provisional Responses	☐
Compact Form	☐
SIP CallerId Obtaining	Remote-Party-Id Display Name ▼
Support URI Percent-Encoding (RFC 3986)	☐
Compare SIP 'To' Header for Transit Out	☐

- SIP Message Resend Timer Base: This parameter let users define the base of timeout to fit the real network. SIP packet will resend if response did not arrive in the base time set in this column. It will send again at "base time" * 2, and send again at "base time" *2 *2. The maximum resend time is four seconds. Resend will stop and restart when the total resend time has reached 20 seconds.
- Max. Response Time for Invite: If the destination does not reply in the set time, the call is failed.
- Invite URL need 'user=phone': There is 'user=phone' in invite packet.
- Compact Form: It decreases the size of SIP header. Tick the check box to enable this function.
- Reliability of Provisional Responses: Defines a type of SIP responses that provide information on the progress of request procession. Tick the check box to achieve reliability for provisional responses.
- SIP CallerId Obtaining: Defines from which part of the SIP packet will the gateway obtain caller ID. There are several places where you can put your caller ID.

  Remote-Party-Id Display Name: It is locate at SIP→Remote-Party-ID→Before [<sip:]

  Remote-Party-Id User Name: It is locate at SIP → Remote-Party-ID → After [<sip:], Before [@]

> **From-Header Display Name:** The standard way is in SIP → Message Header → From → SIP Display info.

- Support URI Percent-Encoding (RFC 3986): It follows RFC 3986 to encode/decode the letters of the basic Latin alphabet, digits, and a few special characters.

- Compare SIP 'To' Header for Transit Out: When FXO is called and the number of Request line and "To" is different, FXO will use the number of "To" to dial out. Please consult your Internet Telephony Service Provider about the format of invite packet from Proxy.

E.164

This is optional. E.164 is to replace number that you dial out into [country code]+[area code] + [destination number]. This is done automatically by VoIP gateway without changing user dialing habit. If your VSP accept only E.164 numbering rule in SIP invite. You will have to fill information in the current VoIP gateway according to the dialing habit. These information are, what will user dial when he tries to make international call? What is the country code of the VoIP gateway? What will user dial when he wants to dial long distance call? What is the local area code? If all information are filled, the dial out invite will be changed from [destination number] to [country code]+[area code]+[destination number].

International Call Prefix Digit	
Country Code	(Other) ▼
Long Distance Call Prefix Digit	
Area Code	

E.164 Numbering	To Invite Proxy	☐
	Transform to Transit Out	☐
ENUM Header Exception	070

- International Call Prefix Digit: Dial out prefix for international calls. These prefixes will be scanned for further process if "E.164 Numbering" is enabled.

- Country Code: Select the desired country code where the gateway is located.

- Long Distance Call prefix Digit: It is used for making a long-distance call.

- Area Code: Local area code where the gateway is located.

- E.164 Numbering

  To Invite Proxy: Invite Proxy to follow the E.164 rule.
  Transform To Transit Out: The call from FXO to PSTN follows the E.164 rule. It applies to one-stage dialing.

- ENUM Header Exception: Do not change the prefix.

NOTE: All settings in this section are specific to your VoIP network. Please ask your Internet Telephony Service Provider whether or not they require these settings.

Example of To Invite Proxy:

    International Call Prefix Digit: 00

    Country Code: 1

    Long Distance Call Prefix Digit: 0

    Area Code: 567

    ENUM Head Exception: 070

Phone Number Dialed By The User	The True Phone Number Dialed By VoIP gateway	Description
23456789	1 567 23456789	Exclude International Call Prefix Digit and Long Distance Call Prefix Digit. Add Country Code(1) and Area Code(567).
0 223 98765432	1 223 98765432	Include Long Distance Call Prefix Digit. Delete Long Distance Call Prefix Digit(0) and add Country Code(1).
00 852 987654321	852 987654321	Include International Call Prefix Digit. Delete International Call Prefix Digit(00).
070 12345678	070 12345678	Include ENUM Head Exception(070). Do not change the number.

Example of Transform to Transit Out:

    International Call Prefix Digit: 00

    Country Code: 1

    Long Distance Call Prefix Digit: 0

    Area Code: 567

    ENUM Head Exception: 070

Phone Number Dialed To FXO From the Remote End	The True Phone Number Dialed By VoIP gateway From FXO to PSTN	Description
1 567 23456789	23456789	Include Country Code(1), Area Code(567). Delete Country Code and Area Code.
1 765 8527413	0765 8527413	Include Country Code(1) and exclude Area Code(567). Delete Country Code(1) and add Long Distance Call Prefix Digit(0).
852 987654321	00 852 987654321	Exclude Country Code. Add International Call Prefix Digit(00).

070 12345678	070 12345678	Include ENUM Head Exception(070). Do not change the number.

Private Network

This section provides a SIP implement of traditional telephony services.

NOTE: This function is only available in Dual Network version.

Phone Book Manager Service

Server Settings

- Enable Phone Book Manager Server: This allows other gateway users to register the IP address and telephone number in this Phone Book manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- Share Phone Book to Clients: While this option is enabled and the gateway is performing as a Phone Book Manager, the Phone Book Manager server will append its Phone Book entries for other clients to lookup.
- TTL (Expire time): If the gateway that is controlled by the Phone Book Manager does not report back within the deadline set by TTL, the gateway will be excluded from the user's list. Each gateway should report to the Phone Book Manager once every 30 seconds.

Clients Settings

- Register to Phone Book Manager: Register to the Phone Book Manager
- Gateway Name for Phone Book Manager: The alias registered with the Phone Book Manager.
- Phone Book Manager Login Password: Enter the registered password. If this system is serving as the Phone Book Manager, the set password is also the password used for registering other gateway systems
- Phone Book Manager IP/Domain: Enter the IP address of the Phone Book Manager. It supports URL (Uniform Resource Locator).
- Phone Book Manager Server Listen Port: The protocol communication port of transmitting signals between the Phone Book Manager and other clients. Please confirm whether the setting is the same as that of the **Phone Book Manager**.

NOTE: A gateway is able to be a server and a client at the same time.

Private Network Numbers

Private Network Numbers		
Gateway Number	20392072	
Line	Type	Number
1	FXS	901
2	FXS	902
3	FXS	903
4	FXS	904
5	FXS	905
6	FXS	906
7	FXO	907
8	FXO	908

- Gateway Number: Enter the representative number for registering to Phone Book Manager.

NOTE: The gateway will not register to Phone Book Manager if the Gateway Number is blank.

- Number: The extension number for each line.

Calling Features

This section provides a SIP implement of traditional telephony services.

Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID
FXS Representative Number			☐ _____	☐ _____	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
FXO Representative Number			☐ _____	☐ _____	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
1	FXS	☐	☐ _____	☐ _____	☐ After[10 - 60] 20 s _____	☐	☐	☐	☐ _____
2	FXO	☐	☐ _____	☐ _____	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)

- Do Not Disturb: Tick the check box to reject all incoming calls from VoIP. Allow only to call out.

- Unconditional Forward: All incoming calls will be forwarded to the **Forwarding Number** automatically. If it forwards to FXO, it only make FXO hook off, not make FXO dial out.

- Busy Forward: Forward incoming calls to the **Forwarding Number** when the line is busy.

- No Answer Forward: Forward incoming calls to the **Forwarding Number** after ring timeout expires without answer.

- Call Hold: Click the check box to enable the call hold on the specific FXS port.

NOTE: Call Hold must be checked; Call Transfer or Call Waiting is active.

- Call Transfer: Click the check box to enable the call transfer feature on the specific FXS port.

- Call Waiting: Click the check box to enable the call-waiting feature on specific FXS port.

- Three-Way Calling / Service ID: It is used for conference all and must work with Proxy Server that supports Three-Way Calling service.

NOTE: The availability of the above features also depends on your VoIP network. Please also check with your Internet Telephony Service Provider on these services.

Calling Feature Instructions:

- Call Hold: The call will be put on hold after the FLASH button is pressed on the phone set. The gateway will play hold music (provided by your VoIP network) to the remote end.

- Call Transfer: Call will be put on hold after FLASH button pressed on local phone set (gateway plays on-hold music to the remote end). Meanwhile, local user can dial out to another number after dial tone is observed. After the handset is replaced back on-hook, the call on hold will then be transferred to the new call regardless of the status of the new call. If wrong number is dialed for the new call, just press the FLASH button to get back the call on hold. In another case, if the local user doesn't hang up the phone after the new call is set up, press the FLASH button to switch between the first call and the new call. Please be informed that the PBX between phone sets and the gateway must support FLASH features in order to use this function. If a phone set is connecting directly to the

FXS port of the gateway and not functioning to FLASH, please adjust the settings
Advanced Options→Line Settings→Flash Time.

- Example of a Three-Way calling:

  1. Alex dials to Bob, Bob answers the call.
  2. Alex presses Flash and calls to Coral (Bob is on hold), Coral answers that call.
  3. Alex dials *61 and then presses Flash.
  4. Thus the conference call is established.
  Or
  5. Alex dials to Bob, Bob answers the call.
  6. Coral dials to Alex (Call Waiting), presses Flash and talks to Coral.
  7. Alex dials *61 and then presses Flash.
  8. Thus the conference call is established.

Advanced Setting

The gateway provides advanced settings: **Call Pickup** and **Automatic Redial**.

NOTE: Automatic Redial is only used for the latest call (no two calls reserved for Automatic Redial). The duration of Automatic Redial is set to 10 minutes. If the callee is still not available after 10 minutes, gateway will not dial again.

- Call Pickup: FXS lines can pick up each others calls. When one of FXS does not answer a call. Another FXS can pick up the call with the function code ***40#**.
  For Example: If Alice calls Bob (9901701) who does not answer. Carol can pick up the call by dialing *40 9901701#.

- Automatic Redial: The callee is initially busy when you call. Hang up the phone and then pick up to dial ***41#** and then hang up. You are hearing a ring tone when the callee is available. You are alerted and then pick up the phone to wait for the called party answering.

- Cancel the latest Automatic Redial: ***42#**.

- Query the time to redial: ***43#**. You can query how long the gateway shall wait to redial.

- Adjust the duration of waiting for Automatic Redial: ***44#**.
  Method: Dial ***44 + Expiry Time#**

- Query the duration of waiting for Automatic Redial: ***45#**.

Advanced Options

Administrator's Name			
Administrator's Password	**********	Confirm Password	**********
Web UI Login ID			
Web UI / IVR Password	**********	Confirm Password	**********
Web UI auto logout [30 - 300 s]	60		

NOTE: There are two operating levels when entering the Web UI. Logging-in as the Administrator allows you to change all settings. A Web UI user only has access to some settings.

- Web UI auto log out: If a user does not act within the effective time range when logging into the web interface, the user will be disconnected from the web page to allow others to log-in.

Dial Wait Timeout [1 - 60 s]	10	Inter Digits Timeout [1 - 60 s]	4
Minimum DTMF ON Length [40 - 500 ms]	80	Minimum DTMF OFF Length [40 - 500 ms]	80
DTMF Detection Sensitivity	(less) ○ 1 ○ 2 ◉ 3 ○ 4 ○ 5 (more)		
FXO Dial Type	DTMF ▾	Pulse Dial Mark/Space Ratio	US (61:39 %) ▾
FXO Impedance	Taiwan 600 Ohm ▾		
FXS Impedance	Taiwan 600 Ohm ▾		
Enable Out-of-Band DTMF ☐	☐ Enable Hook Flash Event		
	◉ RFC 2833 Payload Type 101 Volume 0 dB ▾		
	○ SIP Info		
Use Second CPT after SIP registered	☐		
Enable Non-SIP Inbox Call	☑		

- Dial Wait Timeout: It is to set the waiting time for the user's first key pressing when dialing a number. The user will hear busy tone if the first key is not pressed within the set time frame.

- Inter Digits Timeout: It is to set the waiting time between each key press after the first digit detected. This variable defines the timeout gateway should wait for later digits.

- Minimum DTMF ON Length / Minimum DTMF OFF Length: Set the ON and OFF length of DTMF tone.

- DTMF Detection Sensitivity: Adjust the sensitivity of the telephone keys.

- FXO Dial Type: Choose dialing type of FXO. There are DTMF and Pulse.

- Pulse Dial Mark/Space Ratio: Duration and break of pulse dial ration.

- FXO/FXS Impedance: Choose correct impedance in your country/area.

- Enable Out-of-Band DTMF: Send DTMF keys (0~9, *, #), follow the RFC2833 rules or via SIP Info.

- Enable Hook Flash Event: According to RFC2833 or SIP info the gateway will deliver Hook Flash signal to the remote party.

- Payload Type：Payload type of RFC2833.

- SIP Info: This is an alternative for DTMF event over IP. When enabled, DTMF is relayed over SIP signaling path using SIP NOTIFY messages.

- Uses Second CPT after SIP registered: This function is usually applied when the user select VoIP as the primary path for outgoing calls. The gateway will generate a different set of tones to inform the user that VoIP is in service. When VoIP call is failed, the user will hear the first set CPT instead of the second one. (for CPT settings, refer to **CPT/Cadence Settings**)

- Enable Non-SIP Inbox Call: Untick on the check box to disable Non-SIP inbox call if all calls need to go through ITSP. Non

Line Settings

Line	Type	Listening Volume	Speaking Volume	Tone Volume	Min. FXS Hook Flash Time [50-950 ms]	Flash Time FXS [200-950 ms] FXO [30-900 ms]	Enable Polarity Reversal	PSTN Answer Detection	PSTN Ring OFF Length [1000 - 20000 ms]	FXS Chip Option 1
1	FXS	0 ▼ All	0 ▼ All	5 ▼ All	90 All	600 All	☐			☑
2	FXS	0 ▼	0 ▼	5 ▼	90	600	☐			☑
3	FXO	0 ▼	0 ▼	5 ▼		600	☐	Disable ▼	4000	
4	FXO	0 ▼	0 ▼	5 ▼		600	☐	Disable ▼	4000	

- Listening Volume: Adjusts the hearing volume.

- Speaking Volume: Adjusts the speaking volume.

- Tone Volume: Adds a new option to make tone volume adjustable. This setting will be applied to all tones generated by the gateway including Dial Tone, Busy Tone, and so on.

- Flash Time:

  FXS: Adjust the detecting period of flash signal from the phone set connected to the FXS port. For example, if pressing the HOLD key will disconnect a call, increase the "Flash Time" should fix this issue.
  FXO: Set the time frame that FXO generates a FLASH signal.

- Enable Polarity Reversal:

  FXS: As the remote site answer this call or hook on the FXS port will reverse the polarity.
  FXO: This option forces the gateway to detect the reversal of polarity on FXO port as the primary signal to drop a call. Some telephone switches or PBX reverse the line polarity to inform the remote site to drop an ongoing call. Please consult with the telephone service provider for availability of this feature.

- PSTN Answer Detection: This is only used with ITSP. When someone makes a call to this FXO port from Internet, it could identify if the remote party of PSTN port answer this call. After it dials to PSTN, it will send "183" to another UAC/UAS. After the remote party of PSTN port answers this call, it will send "200ok" to another UAC/UAS.

- PSTN Ring OFF Length: It is used to detect if the PSTN remoter party is on-hook through the ring length from PSTN by FXO port. If the ring length form PSTN is larger than this setting, it is going on-hook by FXO port, and it makes FXO not answering the call.

- FXS Chip Option 1: It is to avoid mis-detecting the loop state of a subscriber line or PBX user loop by FXS interface. In some places, the voltage of off-hook makes it mis-detect the idle state and the active state by FXS interface. Untick this variable if it mis-detects the state by FXS interface in your place.

Codec Settings

Preferred Codec Type	G.729 8kbps				
Jitter Buffer [60 - 1200 ms]	120				
Silence Detection / Suppression	☑	Echo Cancellation	☑		
Codec	☑ G.711 u-law	☑ G.723.1 / G.723.1 6.3k	☑ G.726	☑ G.729	☑ G.711 a-law
Packet Interval (ms)	20	30	20	20	20
Approximate Bandwidth Required (kbps)	85.6	20.8	53.6	29.6	85.6

- Preferred Codec Type: Since different voice codec have different compression ratios, the sound quality and occupied bandwidths are also different. It is recommended to use the default provided (G.723.1) because it occupies less bandwidth and will provide better sound quality.

- Jitter Buffer: Adjust the jitter to receive a packet. If the jitter range is too wide, it will delay voice transmission.

- Silence Detection / Suppression: If one side of a connection is not speaking, the gateway will stop sending voice data (package) to decrease bandwidth usage.

- Echo Cancellation: Prevents poor telecommunication quality caused by echo interference.

- Codec: Choose the codec that you needs.

- Packet Interval: Defines how long the gateway sends a RTP packet (voice packet) to the remote end. The larger the value, the more voice delay.

- Approximate Bandwidth Required: The bandwidth required varies with codec format and packet time.

Fax Settings

T.38	☑ Enable High Quality	
T.30	FAX Codec	G.711 u-law 64kbps
	FAX Jitter Buffer [60 - 1200 ms]	200

NOTE: When a fax tone is detected in a call, the gateway will automatically switch from voice mode to fax mode. So fax settings will be temporarily applied to a specific port which detects fax tones, instead of its default voice settings.

- T.38: T.38 FAX relay function is the best choice fro reliable and efficient facsimile transmission over network. It transmits and receives FAX waveform (relaying) over the

codec negotiated during call setup this bandwidth consumed is lowered. T.38 protocol also supports redundancy to get better FAX quality.

- Enable High Quality: To compensate possible loss of packet during transmission, this function will send T.38 packet twice over network. It increases approximately double bandwidth but offers good and reliable quality.

- T.30: T.30 provides another choice for FAX over IP without compression. It transmit FAX signal as voice thus uncompressed G.711 would be the choice. (G.726 also works but not recommended). Due to this nature, T.30 always requires a SDP change (change of codec within a session, SIP Re-Invite required) after FAX tone detected by the callee. This is a key even to identify if T.30 works in a new environment.

- FAX Jitter Buffer: Adjusts the jitter to receive fax packets. If the jitter range is too large, it will delay fax transmission.

NOTE: When you send fax over an IP network it needs your network to support fax over IP functionality (either T.38 or T.30). Please consult your Internet Telephony Service Provider for this setting.

Drop Inactive Call

This is used as a standard to determine whether or not to hang up the phone. The gateway will hang up the phone automatically to avoid keeping the line engaged if the detected volume is below the **Silence Detection Threshold** and the exceeds the **Drop Silent Call Timeout**.

Silence Detection Threshold [0=disable, 1 - 60 dB]	0
Drop Silent Call Timeout [0=disable, 1 - 3600 s]	120

- Silence Detection Threshold: The volume below the threshold is used as a standard to determine whether or not to hang up the phone.

- Drop Silent Call Timeout: If the detected volume is below the threshold and the time exceeds the silence detection interval, the gateway will hang up the phone automatically to avoid keeping the line engaged.

NOTE: Please be careful with these settings. Improper values might cause unexpected automatic disconnection of a call. Default values are recommended.

Voice Menu Options

Voice Menu Options	
Enable	☑

- Voice Menu Options: Tick the check box to enable or disable IVR function.

NOTE: When disabled, call pickup, Automatic Redial and unattend transfer will be disabled.

Digit Map

Digit Map now is combined the original feature of Digit Map and Speed Dial. You can use "?" or "%" in the column of Scan Code, VoIP Dial-out and PSTN Dial-out. "?" is a single digit, and "%" is wildcard. It provides a mapping between the number received from user and the replaced or modified number for real dial out. With this function, user can easily add certain leading digits to replace full number. There are 50 sets of leading digit entries to choose voice routing interface.

Alert if Auto fails	☐
Enable Pound Key '#' Function	☑
Default Call Route	Auto (VoIP first) ▼

- Alert if Auto fails: Tick the check box to play a voice announcement before calling out. It reminds user that this call is through PSTN.

- Enable Pound Key ' # ' Function: It is to speed up the connection of a call by entering ' # ' after a complete phone number is dialed.

- Default Call Route: Defines the default call route of the gateway. If **Default Call Route** is Deny, all numbers will not be accepted.

  Auto (VoIP first): The call route is VoIP first, and the next is PSTN.
  VoIP: The call route is VoIP only.
  PSTN: The call route is PSTN only.
  Deny: The call will be denied.

Digit Map Testing

Digit Map Testing		
Test Dial No.		Run
Result		

- Test Dial No.: You have to set some rules in Digit Map Setting first and enter the number for test.

- Result: The gateway will show the number for VoIP Dial-out and PSTN Dial-out according to the Digit Map Setting as below.

Digit Map Setting

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	☐				10	Auto (VoIP first) ▼
2	☐				10	Auto (VoIP first) ▼
3	☐				10	Auto (VoIP first) ▼

- Enable: Tick the check box to enable detection of this entry.
- Scan Code: Defines the digits for the gateway to scan while user is dialing.
- VoIP Dial-out: Defines the dialed number rule for the gateway to call through Internet.
- PSTN Dial-out: Defines the dialed number rule for the gateway to call through PSTN.
- User Dial Length: Defines total number of digits that user dialed. A setting of zero tells the gateway scans digits only and disregards the total digit count.
- Route: Determine the interface calls should go through if above conditions satisfied.

Methods of Digit Map:

Method 1- Single mapping: Fill a short code into the **Scan Code** column, and enter the desired phone number into the **VoIP Dial-out** or **PSTN Dial-out** column.

Example - Single mapping,

Scan Code: 55
VoIP Dial-out: 07021234567
User Dial Length: 2
Route: VoIP

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length	Route
1	☑	55	07021234567		2	VoIP
2	☐				10	Auto (VoIP first)
3	☐				10	Auto (VoIP first)

Pick up the handset and dial 55 and the gateway will dial 07021234567. You also can use Digit Map Testing to know that the gateway will dial 07021234567 and go through Internet.

Digit Map Testing		
Test Dial No.	55	Run
Result	#1: VoIP=07021234567	

Method 2- Multi mapping; Fill the prefix code into the **Scan Code** column and the format to transfer into the **VoIP Dial-out** or **PSTN Dial-out** column.

Example 1 - Multi mapping,

Scan Code: 2???
PSTN Dial-out: 351006???
User Dial Length: 4
Route: PSTN

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	☑	55	07021234567		2	VoIP
2	☑	2???		351006???	4	PSTN
3	☐				10	Auto (VoIP first)

Pick up the handset and dial 2301. the gateway will dial 351006301 and go through FXO. You also can use Digit Map Testing to know that the gateway will dial 351006301 and go through FXO.

Digit Map Testing	
Test Dial No.	2301 Run
Result	#2: PSTN=351006301

Example 2 - Multi mapping,

Scan Code: 0%
VoIP Dial-out: 0%
PSTN Dial-out: 1805%
User Dial Length: 0
Route: Auto

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	☑	55	07021234567		2	VoIP
2	☑	2???		351006???	4	PSTN
3	☑	0%	0%	1805%	0	Auto (VoIP first)

Pick up the handset and dial 0423456789. the gateway will dial 0423456789 and go through Internet first. If the call is fail to Internet, the gateway will dial 1805423456789 and go through

FXO. You also can use Digit Map Testing to know that the gateway will dial 0423456789 to Internet and 1805423456789 to FXO.

Digit Map Testing	
Test Dial No.	0423456789 Run
Result	#3: VoIP=0423456789 PSTN=1805423456789

Method 3- Substitution; It helps you dial to destination that you can not dial by phone. Destination like: test@1.1.1.1. Fill the number into the **Scan Code** column and enter the desired name into the **VoIP Dial-out** column.

Example,

Scan Code: 11
VoIP Dial-out: test
User Dial Length: 2
Route: Auto

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	☑	11	test		2	Auto (VoIP first)
2	☐				10	Auto (VoIP first)
3	☐				10	Auto (VoIP first)

Pick up the handset and dial 11. The gateway will dial "test" and go through Internet. You also can use Digit Map Testing to know the dialing result.

Digit Map Testing	
Test Dial No.	11 Run
Result	#1: VoIP=test PSTN=11

NOTE: In the example of Method 3, the result also shows that the gateway will dial 11 and go through FXO. That means the gateway will dial 11 to FXO if the call is fail to Internet. Please select the route is VoIP in this rule if the route is only able to Internet.

Phone Book

The gateway can set up and store 100 phone numbers into a phone book and provides an IP address query when calling to other gateway(s). If no Phone Book manager is set within a gateway group, then all gateways have to set up phone data for each gateway to communicate with each others.

	Phone Book 1 - 5	6 - 10		
#	Gateway Name	Gateway Number	IP / Domain Name	Port
1				5060
2				5060
3				5060

- Gateway Name: Enter another gateway's code or an easy-to-remember name. This parameter is optional.

- Gateway Number: Enter the desired number of another gateway.

- IP/Domain Name: Enter the IP address or URL (Uniform Resource Locator) of another gateway.

- Port: Enter another gateways' listen port.

Caller Filter

This function is used at allow or deny SIP Invite from the Proxy list ONLY.

Allow	Deny	
Enable	Filter IP address	Subnet mask
☐		
☐		
☐		

- Allow: Defines the entries are allowed.

- Deny: Defines the entries are denied.

- Filter IP address: Enter the start IP you would like to allow/deny.

- Subnet mask: Enter the subnet mask you would like to allow/deny.

CDR Settings

The user can set up a CDR Server to record call details for every phone call with TCP protocol. The present CDR provides the call event such as HOOK ON, HOOK OFF, DIALED NUMBER, DATE…recording in a text file and which can be imported to prepare an analysis report.

☐ Send record to CDR Server	
CDR Server IP / Domain	
Port [1 - 65535]	8080
Support RADIUS	☐
RADIUS Accounting Port [1 - 65535]	1813
RADIUS Server Secret	**********
RADIUS User ID	
RADIUS Password	**********

- Send record to CDR Server: Tick the check box to enable the call detail recording.

- CDR Server IP: Enter the IP address of the CDR server.

- Port: Enter the listen port of the CDR server.

- Support RADIUS: Tick the checkbox to enable RADIUS as database and enter the information of RADIUS needed. It includes RADIUS Accounting Port, RADIUS Server Secret, RADIUS User ID and RADIUS Password.

Language

The system provides English, Traditional Chinese, and Simplified Chinese for displaying text on web pages. Changing the language setting also changes the language for IVR (Interactive Voice Response).

Web UI / IVR Language	English ▼

Transit Call Control

This is to control outgoing call and incoming call through FXO. Transit Call Control is effective when it cooperates with Long-Distance Control Table. Long-Distance Exception Table is for an exception and it will not be restricted by Transit Call Control and Long-Distance Control Table. You have to enable both of **Inbound/Outbound Call Control** and **PIN Code**.

NOTE: Transit Call Control is active in one-stage dial.

Inbound Call Control	☐	Outbound Call Control	☐

- Inbound Call Control: Check the inbound PIN code when users make phone calls from a PSTN network to FXO and then using a VoIP - only effective for incoming calls calling from PSTN network.

- Outbound Call Control: Check the outbound PIN code when users utilize FXO interface to divert to PSTN network - only effective for outgoing calls being diverted to PSTN network.

#	PIN Code	Enable	Privileges
1		☐	0 ▾
2		☐	0 ▾
3		☐	0 ▾

- PIN Code: Enter the PIN code (4-6 digits or leave blank. A blank indicates no PIN code is required at this level. Generally, the PIN at level 5 can remain blank to simplify the phone number.)

- Enable: Tick the check box to enable the PIN code at each level.

- Privileges: The level is divided into 0~5 (The levels are in descending order; 0 stands for the highest authority and 5 stands for the lowest.)

The dialing principle to PIN Code is below:

* inbound call control PIN code* outbound call control PID code* phone number

Using * to separate PIN code and the phone number is based on actual settings.

Long-Distance Control Table

This table controls the level of authority of an outgoing (transit out) call that is dialed through FXO and diverted to PSTN network, as below

#	0	1	2	3	4	5
1	0204					
2						
3						

Descriptions:

- Digit strings in this table are prefixes that the gateway will check on dialed numbers in transit out calls.

- This table is used to prohibit dialing any numbers started with specified prefixes.

- If Level 0 (the highest level) is set to prohibit dialing any number started with prefix 0204, then any level below 0 (including Levels 1 to 5) is also prohibited.

- If Level 1 is set to prohibit dialing any number with prefix 0, then any level below 1 (including Levels 2 to 5) is also prohibited. Since Level 0 is not restricted to any prefix, therefore at level 0 users can dial a number with the prefix 0.

NOTE: Downward Restriction — If the users at a higher level cannot dial a number with a certain prefix, then users at lowers level also cannot dial a number with the same prefix.

Long Distance Exception Table

This table handles any exceptions to the long-distance call table.

According to the Long Distance Control Table, users at Level 0 are prohibited from dialing a number with the prefix 0204. But, if the number 020488988 is set in the Exception Table as above, then users could then dial this number.

#	0	1	2	3	4	5
1	020488988					
2						
3						

NOTE: Upward Opening —If the users at a lower level can dial a number with a certain prefix, then the users at higher levels can also dial a number with the same prefix.

CPT/Cadence Settings

CPT/Cadence parameters serve as the basis of an FXO interface to determine whether or not a PSTN-call receiving party has hung up the phone. If the following parameters differ from the parameters of the actual assigned lines, it could cause the FXO to continue to engage a line.

Busy Tone Cadence Measurement

BTC Enable	Busy Tone Cadence Measurement				
	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning
BTC # 1	0	0	0	0	Yes
BTC # 2	0	0	0	0	Yes
BTC # 3	0	0	0	0	Yes
BTC # 4	0	0	0	0	Yes
BTC # 5	0	0	0	0	Yes
BTC Detection Sensitivity	(less) ○ 1 ○ 2 ○ 3 ◉ 4 ○ 5 (more)				

- Busy Tone Cadence Measurement: Provide a solution of FXO integrated with PSTN or PBX. FXO will learn the busy tone automatically.

- BTC Detection Sensitivity: The more sensitivity, the more quickly the gateway will cut off the call. If the gateway often cut off an un-finished call, select less sensitivity.

CPT parameters Table

The CPT has 3 sets of parameter tables. Please adjust the CPT based on local PSTN or PBX.

Moreover, users can use CPT Auto Detect to detect CPT parameters. Instructions are shown in the following section. The method to detect CPT described as below.

# 1 Enable	Setting 1					Default
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	350	440	3000	0	0	0
Congestion Tone	480	620	250	250	0	0
Busy Tone	480	620	500	500	0	0
Ring-Back Tone	440	480	1000	2000	0	0

UDT Detection

If the CPT auto detect function is not able to determine whether or not a PSTN-call receiving party has hung up the phone, then the UDT detection function can serve as a back up. To do this, enter the high/low frequency parameters from the CPT table into the UDP table.

Disable UDT	UDT Setting	# 1 Disable	# 2 Disable	# 3 Disable	# 4 Disable
Congestion Tone / Busy Tone	Low Frequency	0	0	0	0
	High Frequency	0	0	0	0

NOTE: To cope with different local PSTN and different PBX models, the gateway provides CPT Auto Detect function to prevent the FXO from engaging a line. However, if the line of the receiving party is engaged and his/her PSTN uses a voice prompt to replace the traditional beep sound, then the gateway would not be able to detect a busy tone. Drop Inactive Call should then be used to determine whether or not to end the call.

CPT Auto Detect

- 2 PSTN phone numbers or 2 PBX extension lines are needed.
- Connect one of the phone sets to the FXO port (P3 or P4 of MODEL 2S2O).
- The line of **Dial Number** must be hook on. Set the outgoing phone number the same as the phone line that is in use as above, and click the **Accept** button to start detection.
- If detection is successful, the parameters obtained will automatically be inserted into the CPT parameter.
- Save settings and restart the system.

Detailed description is given below:

Click **CPT auto detect** at the bottom of Trunk CPT Settings.

Use CPT auto detect

If CPT auto detect is used, the function would halt every operation of the gateway. Select **I am sure of it**, and then click the **Accept** button. Wait for 15 seconds, and then you will enter CPT auto detect window.

Direct Connection to PSTN

- Connect one of the trunk lines to the FXO Port (For MODEL 2S2O, please connect to P3).

- The line of **Dial Number** (36008913) must be hook on.

- Detect Channel: Enter 3 (The trunk line is connected to P3, and uses P3 for outgoing detection).

- Phone Number: Enter the number of the FXO line.

- Dial Number: Enter the number of the end to be tested—36008913.

- CPT group: Enter the group that after testing to replace with.
- Finally, click the **Accept** button.

Detection in Progress: during detection, the following windows will appear.

Current Operation

```
Detecting Dial Tone...
```

Current Operation

```
Detecting Reorder Tone...
```

Once detection of a busy tone is in progress, the gateway will dial the number to be tested (in this case 36008913). After it rings pick up the phone and enter "#", then hang up. The gateway will then detect a busy tone automatically.

Current Operation

```
Detecting Busy Tone...
Please hook-off after the extension ring up, then press
[#] before hook-on.
```

After detecting it will be as below:

Detect again	Detection Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

#2	Current Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

If you sure overwrite Current Value, please click Accept

Accept

<u>Connected to a PBX Extension Line</u>

If the gateway is connected to a PBX extension line, then the busy tone of both the PBX and the PSTN must be detected.

- Connect one of the PBX extension lines to the FXO Port (For MODEL 2S2O, connect the line to P3).

- Detect Channel: Enter 3 (The trunk line is connected to P3, and uses P3 for outgoing detection).

- Phone Number: Set the number of FXO line –to detect Reorder Tone.

- Dial Number: Enter the number of the end to be tested—307.

- Save detect value to CPT group: Enter the group that after testing to replace with.

- Finally, click the **Accept** button.

Filling in the CPT Table

Fill in the table after the detection is completed as below, where the values are the frequency and On-and-Off ratio detected. Please click the **Accept** button. If connecting gateway to a PBX extension line, please do not set the detected busy tone of the PBX and the PSTN in the same set, otherwise the value detected the first time will be overwritten.

Detect again	Detection Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

#2	Current Value					
Tone	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	400	0	3000	0	0	0
Congestion Tone	400	0	250	250	0	0
Busy Tone	400	0	250	250	0	0

If you sure overwrite Current Value, please click Accept

Accept

Save Settings

Tick the check box to save the new parameters and restart the gateway after the test is completed. Then click the **Accept** button. The gateway will use the new parameter to detect whether a call has ended.

System Operation	
Detect again	
☐ Save Settings	Save all configurations.
Be sure to save all settings before restart.	
☐ Restart	Restart the Gateway right away. All calls will be DROPPED when Restart.

Accept   Reset   Back

System Information

This page shows that the status of the gateway. There are Port Status, Server Registration Status, WAN Port Information, LAN Port Information and Hardware.

System Information

Port Status

No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	701	Idle	0		Disabled	
2	FXS	702	Idle	0		Disabled	
3	FXO	703	Idle	0		Disabled	
4	FXO	704	Idle	0		Disabled	
SIP Proxy Hunting Number Registration				FXS Disabled (02:40:07) , FXO Disabled (02:40:07)			

Server Registration Status

DDNS Registration	Disabled (02:40:07)
Phone Book Manager Registration	Disabled (02:40:07)
STUN Registration	Disabled (02:40:07)
UPnP Negotiation	Disabled (02:40:07)

WAN Port Information

Factory Default MAC Address	00 00 00 00 00 00
IP	192.168.1.2
Subnet Mask	255.255.255.0
Default Gateway	192.168.1.254
DNS	168.95.1.1

LAN Port Information

MAC Address	00 00 00 00 00 01
IP	192.168.8.254
Subnet Mask	255.255.255.0

DHCP Server

Enabled	Yes
IP Pool Range	192.168.8.1 - 192.168.8.250
Lease Time	1 hour(s)
DNS	168.95.1.1

Hardware

- Port Status: It includes if each port registers to Proxy successfully, the lasted dialed number, how many calls each port had since the gateway is start, etc.

- Server Registration Status: It shows the registration status of DDNS, Phone Book Manager, STUN and UPnP.

- WAN Port Information: It shows IP address, subnet mask, default gateway and DNS server. If you use PPPoE to obtain IP, you can know if the IP is obtained through this.

- LAN Port Information: It shows LAN port IP, subnet mask, and the status of DHCP server.

- Hardware: It shows the hardware platform.

RTP Packet Summary

Display the information of the last completed call. This report contains peer IP, peer port, packet sent, packet received and packet lost. Press Refresh button to get the latest RTP Packet Summary.

RTP Packet Summary						
Line 1	G.711 u-law 64kbps	Packet Sent	0	Packet Received	0	Packet Lost	0
The last packet's source IP				The last packet's source Port	0	
Line 2	G.711 u-law 64kbps	Packet Sent	0	Packet Received	0	Packet Lost	0
The last packet's source IP				The last packet's source Port	0	

Refresh

STUN Inquiry

Use STUN Inquiry to detect your IP sharing device's NAT type and communication between a STUN server and client.

NAT Type	Unknown
STUN Server IP / Domain	
STUN Server Port [1 - 65535]	3478

Ping Test

Use Ping to verify if a remote peer is reachable. Enter a remote IP address and click Test to ping the remote host.

Ping Destination	
Number of Ping [1 - 100]	4
Ping Packet Size [56 - 5600 bytes]	56

Test   Stop

NTP

This section works with Time server to provide NTP time synchronization. The gateway will get its local time from specific time servers after it connects to Internet.

- Time Zone: Set the Time Zone where the gateway resides.
- Time Server #1~#3: Set the Time Server where the gateway should sync up during start up. (via NTP protocol)

Backup/Restore

You can backup settings to a file on a PC and restore it back.

NOTE: You have to save settings and restart, and all settings will be restored.

- Configuration File: Click **Backup** to backup all settings.
- Configuration Template File: Click **Backup** the settings as a template file for editing. (not for end user).

- Restore Default Configurations: Restore the device back to the factory default settings.

Provision Settings

This section sets parameters required by Auto Provisioning System. Typically, Provision Server is used to provision, configure, manage and maintain subscribers and network users. This gateway, acts as a part of subscribers, can be controlled by Provision Server. The gateway provides a simply way for users to connect and send request to Provision Server by enabling this setting. With this system, the Server can not only easily modify a configuration file to change gateway settings but to assign latest firmware for specific gateways to upgrade. Besides, Provision Server also reports the status of the gateway and all actions will be recorded in log file that offers users to trouble shouting effectively.

NOTE: Fill in the parameters needed by the Provision Server from your service provider. Please check with your service provider about the availability of these services.

Enable Auto Provisioning	☐
Provision Server Address	
Port [1 - 65535]	10101
Packet Format	Proprietary ▼
Connect Provision Server During Start Up	☑
Connect Provision Server Periodically	☑ Auto Provision Interval [60 - 604800 s] 10800 Random Offset [1 - 1800 s] 600
Provision Retry Times [0=always, 1 - 99] [0 - 99]	10 Retry Interval [30 - 120 s] 30
Suspend Service	☐

Binding Server for Trigger	☐
Binding Port [1 - 65535]	10103
Binding Interval [1 - 65535 s]	10

- Enable Auto Provisioning: Tick the check box to start provisioning.

- Provision Server Address: Enter the IP address/Domain of Provision Server required by your provider.

- Prot: Enter the port number of Provision Server used.

- Packet Format: Select the packet transmitting format required by provision server.

- Connect Provision Server During Start Up: The gateway will connect to Provision Server when it powers on or reboots.

- Connect Provision Server Periodically: Adjust the parameters for the gateway to connect to provision server periodically.

- Auto Provision Interval/Random Offset: Adjust the parameters for the gateway to do auto provision task.

- Provision Retry Times/Retry Interval: Adjust Retry times or interval.

- Suspend Service: When it is clicked, indicating the server has stopped providing provision and VoIP call service. Each FXO/FXS port is not available to make any call.

NOTE: Contact your server provider while necessary.

- Binding Server for Trigger: Tick this check box to trigger of a connection between server and gateway. Server will bind a port for the gateway to send provision request and tell the gateway to upload syslog onto the assigned Syslog Server.

- Binding Port: The binding port number of the server is used to tell the gateway the path of binding server.

- Binding Interval: Set the desired Interval at which the gateway will keep the binding.

System Operations

Some settings are effective only after **Restart**. Remember to save all settings using **Save Settings** before you restart.

- Save Settings: Click the **Save Settings** check box and the **Accept** button after completing changes. The new settings will take effect after the gateway is restarted.

- Restart: Click the **Restart** check box and click **Accept** button if it is necessary to restart the gateway.

Software Upgrade

The gateway provides a software upgrade function for a remote source. Please consult your service provider for information about the following details.

To Save Current Settings, Save Settings

Current Software Version No. [1.2.36.3-92-84]

Upgrade Server	⊙ TFTP ○ FTP ○ HTTP
Server IP Address	
Server Port [1 - 65535]	69
User Name	
Password	
Directory	

All calls will be DROPPED during upgrade.

Accept

- Upgrade Server: Choose the server type given by your service provider.
- Server IP Address: Enter the software server IP address.
- Server Port: Enter the port of that the server uses. TFTP is 69, FTP is 21,and HTTP is 80.
- User Name/ Password: The account information to access an FTP server.
- Directory: The directory path of the upgrade files for TFTP or FTP or HTTP.

Logout

The gateway only allows one user at a time to log-in, whenever a change is made, please save the settings and restart the gateway, or logout to avoid a situation where other users cannot long-in to change settings.

To save settings, click Here

5. IP Sharing Functions

All gateway series have a built-in IP sharing function. The settings and instructions at a PC end are described below:

Current Intranet only supports static IP mode, and the settings at the PC end are as follow:

Available IP address Range   : 192.168.8.1 – 192.168.8.253
(default address of gateway is 192.168.8.254)
Subnet Mask   : 255.255.255.0
Default Gateway   : 192.168.8.254

The above values vary with different LAN Port Settings.
Assume gateway's LAN settings are,
IP address   : 192.168.3.1
Subnet Mask   : 255.255.255.0

Then, the settings at PC end should be as follows:
Valid IP address range   : 192.168.3.2 – 192.168.3.254
Subnet Mask   : 255.255.255.0
Default Gateway   : 192.168.3.1

WAN Port
Connected to
Internet

LAN Port connected to
Intranet
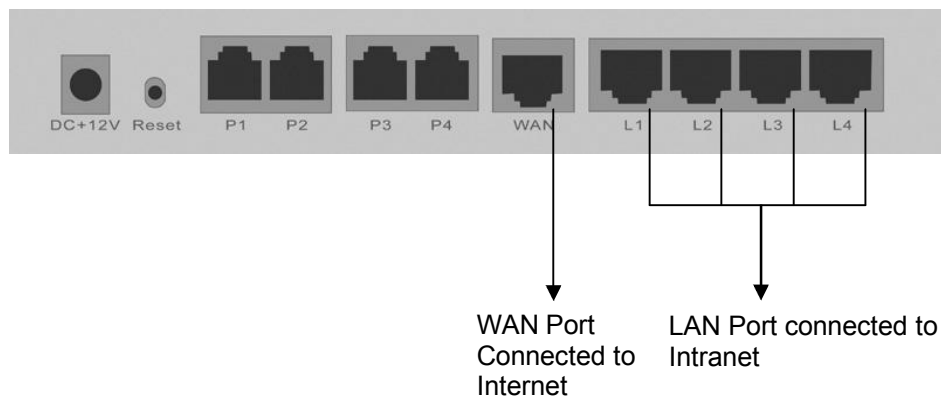
The IP settings on PC are as follows (using Windows 2000 for example)
Open Start->Settings->**Control Panel**
Open **Network and Dial-up Connection**
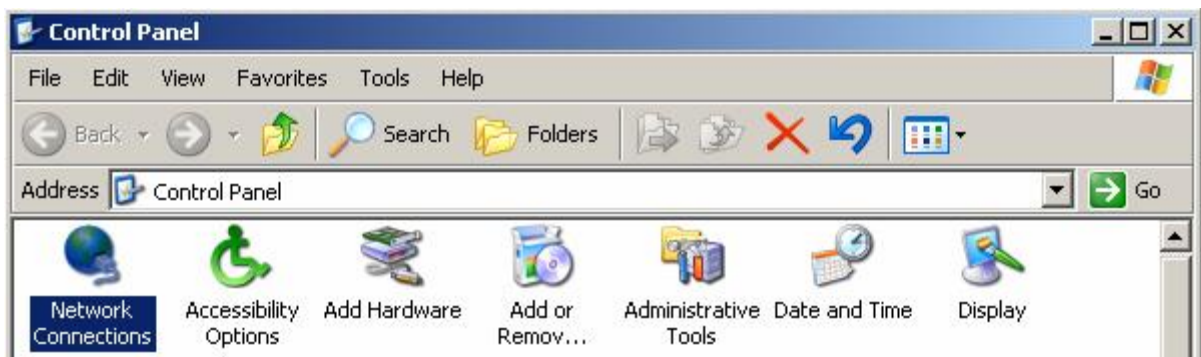
Open Local Area Connection
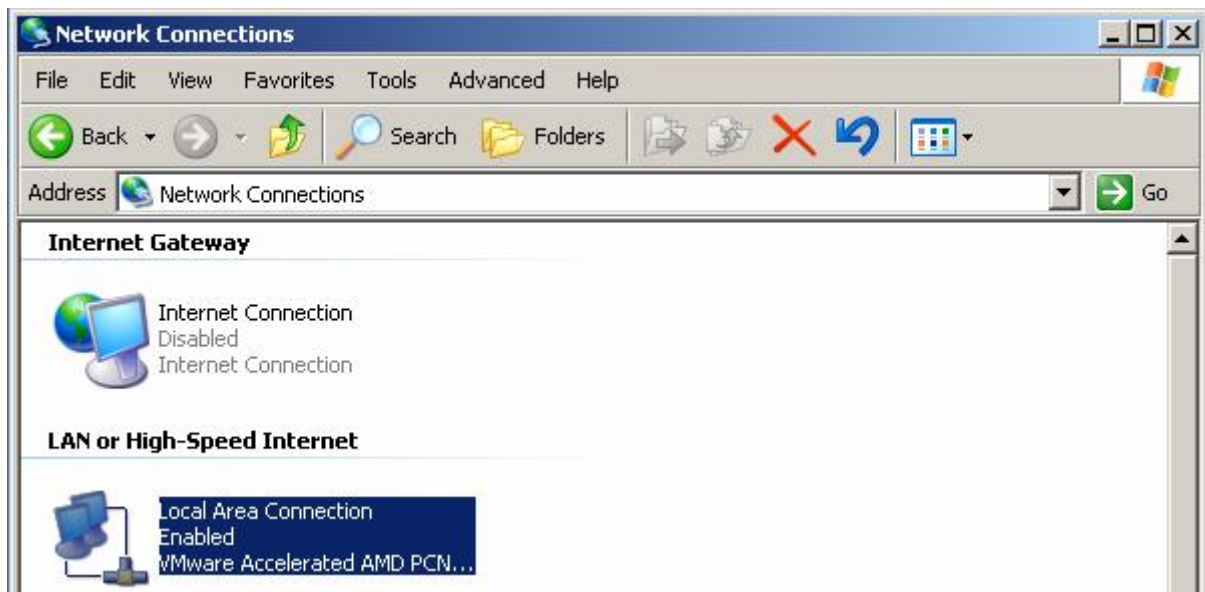
Click **Properties**
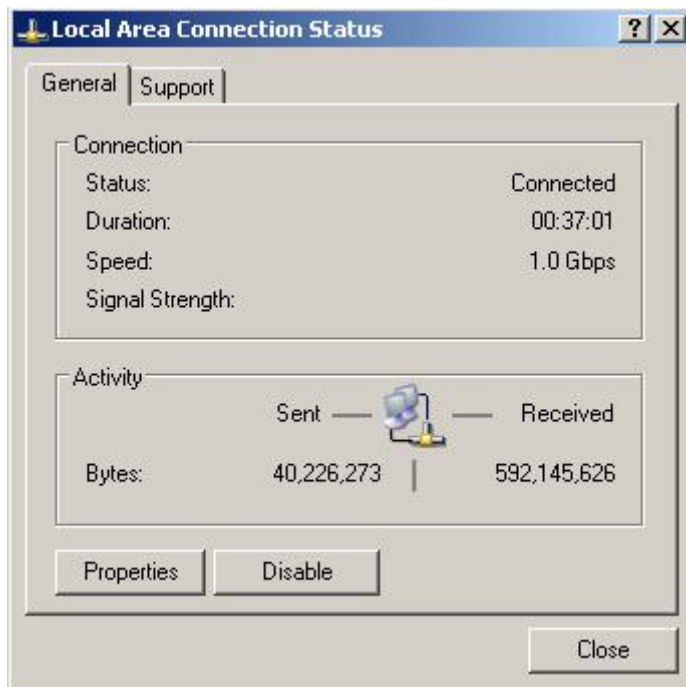
Select **TCP/IP**, and then click **Properties**.

Select "**Use the following IP Address**" and enter IP address, Subnet Mask, and Default Gateway. Please note that an IP address in the same domain cannot be reused. Then, enter the DNS server IP address (varies in different networks. consult your ISP's service for information). Click the "**OK**" button and after completing the settings, users can use both the VoIP and network services concurrently.
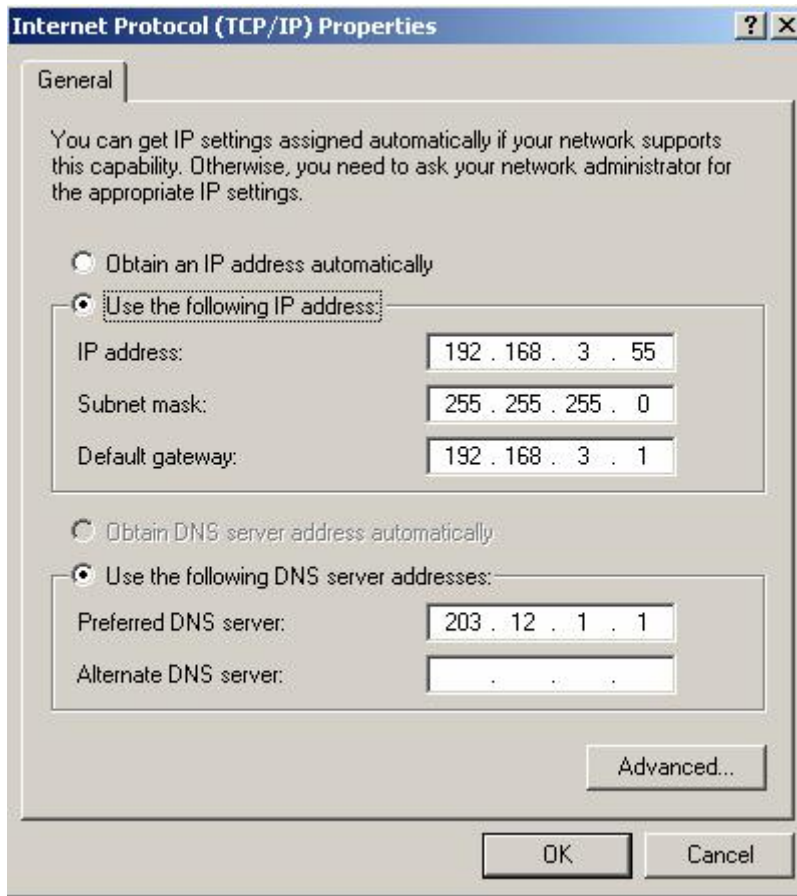
6. Coding Principle

Instruction

- Dial the phone number which you want to call and press # to call out immediately, or wait until the "Inter DTMF Timeout" expires (defined in **Advanced Options\Inter Digits Timeout**, default=4 seconds).

- If the phone number fits the setting of the Digit Map, the gateway dials out the phone number through the assigned interface automatically.

- The phone number should have at least 2 digits (not including * and #).

Dialed Number Processing Flow

To maintain maximum flexibility, the number dialed will be looked up in several tables defined by the gateway. If no match is found, it will look up the number form the registered Proxy Server. The number look up flow is shown below:

```
┌──────────┐   ┌──────────┐   ┌──────────┐   ┌──────────┐   ┌──────────┐
│ Digit Map│──▶│ Extension│──▶│  Phone   │──▶│Phone Book│──▶│   SIP    │
│  Table   │   │  Number  │   │   Book   │   │ Manager  │   │  Proxy   │
└──────────┘   └──────────┘   └──────────┘   └──────────┘   └──────────┘
```
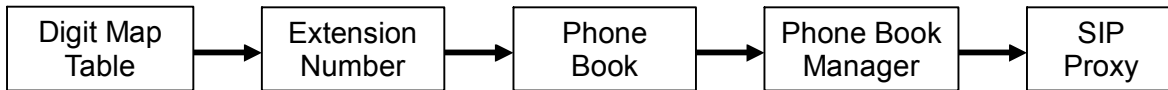
Example for Call Out via VoIP – Contents of Invite

The phone book settings:

#	Gateway Name	Gateway Number	IP / Domain Name	Port
1		88	88.77.66.55	5060
2				5060

- When dialing 88 or transferred to 88 with Digit Map, the gateway sends INVITE sip:88@88.77.66.55:5060

- When dialing 88123456 or transferred to 88123456 with Digit Map, the gateway sends INVITE sip:123456@88.77.66.55:5060

Example for Match phone numbers invited by callers

The table below is provided as a general reference expresses phone numbers dialed by the gateway instead of real phone numbers that callers dial.

Match	Scheme	Description
The same as "FXS Representative Number"	Ring FXS according to "FXS Group Hunting / Ring Priority" settings	
The same as "FXO Representative Number"	Off hook a FXO	It is not applied to registration with SIP Proxy
The same as "FXS Representative Number + Extension Number"	Ring or off hook the Extension	If Extension line is FXS, it should ring. If Extension line is FXO, it should off hook It is not applied to registration with SIP Proxy
The same as FXO Extension Number	Off hook the FXO	

A Prefix is the same as "FXS Representative Number +FXO Extension Number"	Eliminate a Prefix and use remaining digits to route calls via FXO	FXS Representative Number is 2252 and one of FXO Extension is 070123456 If callers dial 2252070123456 6371, the gateway dial 6371 via FXO Extension
A Prefix is the same as FXO Extension Number	Eliminate a Prefix and use remaining digits to route calls via FXO	One of FXO Extension is 070123456 If callers dial 070123456 6371, the gateway dial 6371 via FXO Extension
Differ from FXS/FXO numbers	Use these digits to route calls via FXO	If callers dial 6371, the gateway dial 6371 via one of FXO line
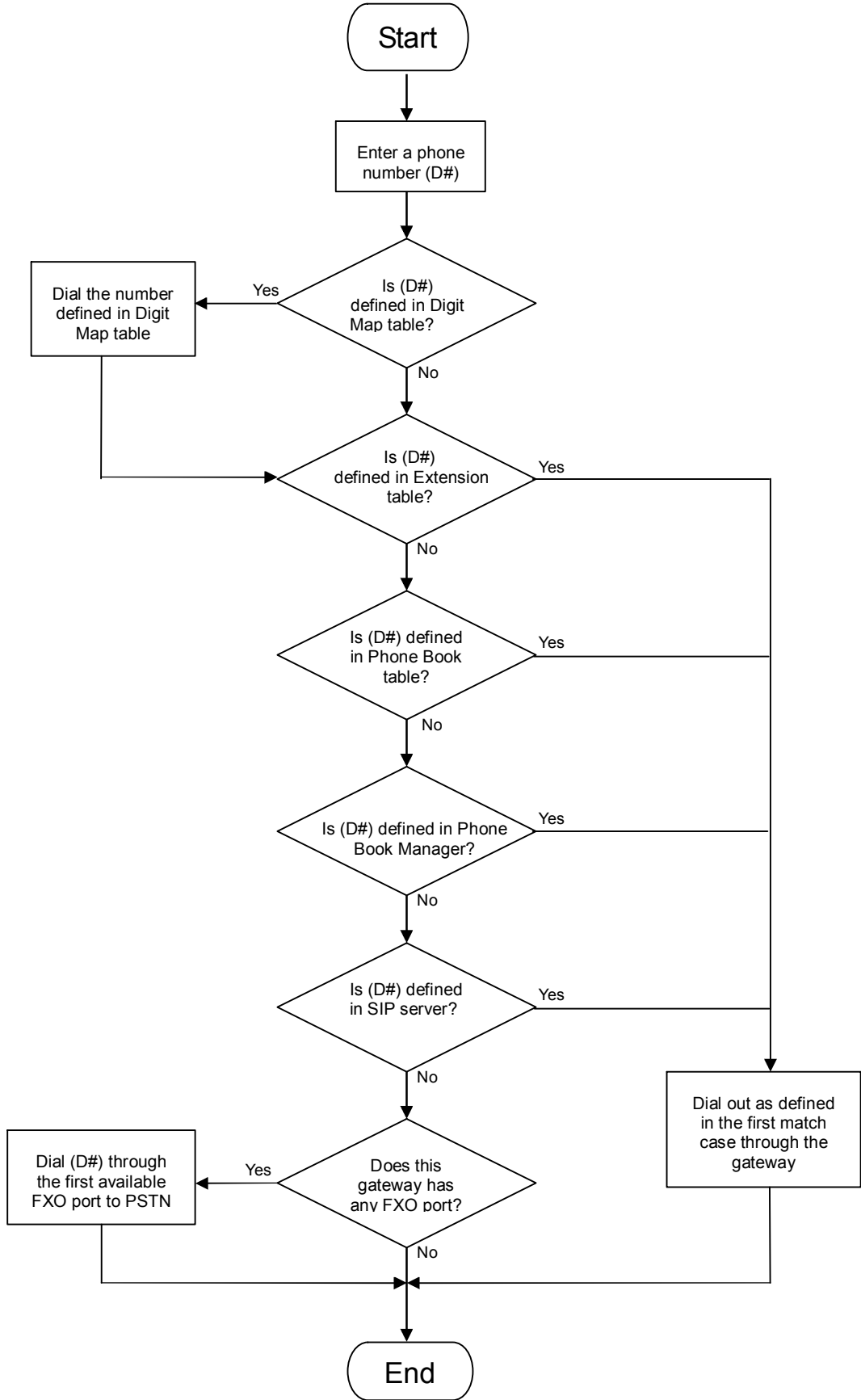
7. Advanced Feature

Static Route

Build static routes within an internal network. These routes will not apply to the Internet.

#	Route	Route Mask	Next Hop IP	Interface
1				▼
2				▼
3				▼
4				▼
5				▼

- Route: Enter the IP of the specified network.

- Route Mask: Enter the subnet mask to be used for the specified network.

- Next Hop IP: The next hop IP address to the specified network.

- Interface: The interface attached to this route.

RIP (Routing Information Protocol)

Establish dynamic routes within an internal network. RIP1, RIP2, or Both are supported. These routes will not apply to the Internet.

Enable RIP	☐
Send Version	1&2 ▼
Receive Version	1&2 ▼
Enable Authentication	☐
Authentication Password	**********
Update Timer	30
Timeout Timer	180
Garbage Timer	120

- Enable RIP: Enable or Disable dynamic routes on the gateway.

- Send Version\ Receive Version: Defines the version used for RIP. They are RIPv1, RIPv2 and RIPv1&2.

- Enable Authentication/Authentication Password: All the boxes in this RIP group should be

filled in the same password if ticked.

Port filtering

Port filtering enables you to control all data that can be transmitted over routers. When the port used at the source end is within the defined scope, it will be filtered without transmission.

Enable Port Filtering	☐

- • Enable Port Filtering: Tick the check box to enable this function.

Port Range	TCP / UDP	Remark
0 - 0	Both ▼	
0 - 0	Both ▼	
0 - 0	Both ▼	

- • Port Range: Set the range of the port to be filtered. If, for example, the port to be filtered is 80 and the selected protocol is Both or TCP, all computers will be unable to use HTTP services (port 80) and will be unable to browse normal web pages.
- • TCP/UDP: Choose to filter TCP, UDP, or Both.
- • Remark: Remark field, write comments for notations.

IP Filtering

IP Filtering is used to limit internal users from accessing the Internet.

Enable IP Filtering	☐	
IP	TCP / UDP	Remark
	Both ▼	
	Both ▼	
	Both ▼	

- • IP: Input the IP address that you want to filter. The listed IP address will be unable to transmit data to and from the Internet.
- • TCP/UDP: Choose to filter TCP, UDP, or Both.
- • Remark: Remark filed, write comments for notations.

MAC Filtering

MAC (Media Access Control) address filtering allows you to filter the transmission of data by network card physical address.

Enable MAC Filtering	☐
MAC	**Remark**

- MAC: Enter a MAC address to prevent the particular device from accessing the Internet.
- Remark: Remark field for this entry.

Virtual Server

Virtual Server allows you to enable users to access the Internet, FTP and other services from behind your NAT. When remote users are accessing web or FTP servers through WAN-end IP addresses, they will be routed to the server at the internal LAN end as appropriate in accordance with externally required services.

Enable Virtual Server				☐
WAN Port Range	**TCP / UDP**	**LAN Host IP Address**	**Server Port Range**	**Remark**
0 - 0	Both ▼		0 - 0	
0 - 0	Both ▼		0 - 0	

- WAN Port Range: Input the port rang for the WAN side.
- TCP/UDP: Select the communication protocols used by the server, TCP, UDP or Both.
- LAN Host IP Address: Enter the IP address that provides various services.
- Server Port Range: Input the port used by the LAN host.

DMZ

DMZ allows the server on the LAN site to be directly exposed to the Internet for accessing data. Either this function or virtual server can be selected for use in accessing external services.

NOTE: Only one host in the LAN can be set as a DMZ host.

Enable DMZ	☐
DMZ Host IP Address	

URL Filter

URL filter is used to deny device on the LAN from accessing specific web sites. The gateway will block any URL that contains the strings listed.

Enable	URL string pattern to be blocked	IP Range	
☐		0.0.0.0	0.0.0.0
☐		0.0.0.0	0.0.0.0
☐		0.0.0.0	0.0.0.0

Special Applications

Provide multiple connections for special applications.

Enable	Name	Incoming Type	Incoming Port Range	Trigger Type	Trigger Port Range	
☐	MSN Gaming Zone	BOTH	2300-2400,28800-29000	TCP	47624	47624
☐	Quick Time	TCP	6970-6999	TCP	554	554
☐	ICU II	BOTH	2000-2038,2050-2051,2069,2085,3010	TCP	2019	2019
☐	PC-to-Phone	TCP	12120,12122,24150-24220	TCP	12053	12053
☐		TCP		TCP	0	0
☐		TCP		TCP	0	0
☐		TCP		TCP	0	0
☐		TCP		TCP	0	0

- Name: The name of the special application.

- Incoming Type: The protocol used to trigger the special application.

- Incoming Port range: Port range on the WAN side that will be used to access the application.

- Trigger Type: The protocol used to trigger the application.

- Trigger Port Range: Port range used to trigger the application.

DoS Prevention Settings

Enable DoS Prevention	☑		
Enable DoS Prevention on LAN	☑		
Whole System Flood	☑ SYN	50	(Packets/Second) [50 - 500]
	☑ FIN	50	(Packets/Second) [50 - 500]
	☐ UDP	68	(Packets/Second)
	☑ ICMP	50	(Packets/Second) [50 - 500]
Per-Source IP Flood	☑ SYN	30	(Packets/Second) [30 - 300]
	☑ FIN	30	(Packets/Second) [30 - 300]
	☐ UDP	68	(Packets/Second)
	☑ ICMP	30	(Packets/Second) [30 - 300]
TCP / UDP Port Scan	☐	TCP / UDP Port Scan Level	Low ▼
TCP Scan	☐		
TCP SYN with Data	☐		
TCP Land	☐		
UDP Echo Chargen	☐		
UDP Bomb	☐		
UDP Land	☐		
Ping of Death	☑		
ICMP Smurf	☑		
ICMP Land	☑		
IP Spoof	☐		
Tear Drop	☐		

- Enable DoS Prevention: To prevent DoS attacks from WAN.
- Enable DoS Prevention on LAN: To prevent DoS attacks from LAN.

Source Blocking	
Enable Source IP Blocking	☐
Blocking Time [2 - 600 s]	120

- Enable Source IP Blocking: Block a particular IP.

- Blocking Time: The time to block the IP.

8. VPN IPSEC ※

Virtual Privet Network lets two networks communicate securely when the only connection between them is over a third network which they do not trust (ex. Internet). For instance, two VPN enabled devices can establish a secure IP tunnel over Internet that allows traffic from both local network to communicate over this virtual dedicate line. IPSEC is one of the best technologies for building VPN tunnels, it provides high level encryption and authentication services at the IP level of the network protocol stack. Working at this level, IPSEC can be used in any public or private IP network to protect any traffic between two tunneled LAN.

This VPN enabled VoIP gateway can connect up to 8 IPSEC tunnels at the same time. Each of the tunnel use different settings to connect different remote node. Two encryption algorithms are supported, 3DES and AES. Authentication type can be MD5 or SHA, which is used for authentication header and data checksum. The encryption and authentication type must be correctly set up so that a tunnel can be built successfully.

To setup VPN tunnels and the VoIP Route, click **VPN IPSEC** from the menu.

IPSEC

VoIP Route | Internet ▾

Tunnel	Enable	Local Security Group	Local Security Group Mask	Remote Security Group	Remote Security Group Mask	Remote Security Gateway
1	No					
2	No					
3	No					
4	No					
5	No					
6	No					
7	No					
8	No					

[Accept] [Reset] [Default]

This is the main window of IPSEC VPN. On the top there is a pull down menu **VoIP Route** that specifies the route VoIP traffic should go through. By default VoIP goes through Internet and this should work in most cases. The "VoIP Route" menu will include only VPN tunnels that is Enabled.

To set or view VPN tunnel parameters, tick Tunnel number link.

IPSEC configurable parameters

VPN IPSEC -> Configuration

- Enable IPSEC: Enable or disable this VPN tunnel.
- Local Security Group: The IP segment of local LAN port.
- Local Security Group Mask: The subnet mask of local LAN port.
- Remote Security Group: The IP segment of remote VPN device LAN port.
- Remote Security Group Mask   : The subnet mask of remote VPN device LAN port.

- Remote Security Gateway: The IP address or FQDN of remote VPN device.

- Encryption: Tunnel data encryption.

- Authentication   : Authentication header and checksum.

- Local ID (optional): ID field to be identified by remote VPN device (ie. "@local-domain.com").

- Remote ID (optional): ID field to authenticate remote VPN device (ie. "@remote-domain.com").

- Pre-shared Key: Password for VPN connection.

- Perfect Forward Secrecy: Key exchange which uses a long-term key (such as the shared secret in IKE) and generates short-term keys as required.

- Key Lifetime: The next exchange key time.

- ISA/KMP Key Lifetime: The keying channel connection life time.

Three steps to setup a VPN connection between two VPN VoIP gateways

Make sure in Network Setting, the LAN segment is different from the remote LAN segment. A VPN tunnel can not be built successfully if LAN segment of two VPN end points are the same or overlap. A static IP address for the WAN port is also necessary.

This is a sample of gateway1:

Network Settings (LAN)			
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0
DHCP Server			
Enable DHCP Server	☑		
IP Pool Starting Address	192.168.8.1	IP Pool Ending Address	192.168.8.250
Lease Time [1 - 9999hours]	1		
Domain Name Server Assignment	⦿ Auto ◯ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

This is a sample of gateway2:

Network Settings (LAN)			
LAN IP / LAN default Gateway	192.168.9.254	Subnet mask	255.255.255.0
DHCP Server			
Enable DHCP Server	☐		
IP Pool Starting Address	192.168.9.1	IP Pool Ending Address	192.168.9.250
Lease Time [1 - 9999hours]	1		
Domain Name Server Assignment	⦿ Auto ◯ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

1. In both gateways you will need to enable IPSEC first, fill in gateway1 WAN IP address, gateway1 LAN segment, gateway2 LAN segment, gateway2 WAN IP address, etc.

This is a sample of gateway1:

In **Local Security Group** and **Local Security Group Mask** fields, put in your local LAN IP segment information. **Remote Security Group** and **Remote Security Group Mask** are to set remote LAN IP segment information. The **Remote Security Gateway** is the WAN IP address of remote VPN device (in this case, gateway2). All other setting should set identical between VPN peers.

This is a sample of gateway2:

<table>
<tr><td colspan="2" align="center">**IPSEC Settings**</td></tr>
</table>

	IPSEC Tunnel 1 Next
Enable IPSEC	☑
Local Security Group	192.168.9.0
Local Security Group Mask	255.255.255.0
Remote Security Group	192.168.8.0
Remote Security Group Mask	255.255.255.0
Remote Security Gateway	2.2.2.2
Encryption	3DES
Authentication	MD5
Local ID	remote-domain.com
Remote ID	local-domain.com
	Key Management
Pre-shared Key	•••••
Confirm Password	•••••
Perfect Forward Secrecy	☑
Key Lifetime [0 - 86400 s]	86400
ISA/KMP Key Lifetime [0 - 86400 s]	86400
	Operation
	[Connect] [Disconnect]

[Accept] [Reset] [Default]

2. Save and restart both gateways then these two gateways should be connected with IPSEC VPN.

System Operation	
☑ Save Settings	Save current settings to the permanent storage of Gateway.
	Without save settings, all current settings would be lost when Gateway is restart ,shutdown or the power is cut off.
☑ Restart	Restart the Gateway right away.
	[Accept]

Notice

※

These functions are only supported for the special hardware. The special hardware supports VPN IPSEC function.

General

"IMPORTANT SAFETY INSTRUCTIONS - When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock and injury to persons, including the following:
-Do not use this product near water for example, near a bathtub, washbowl, and kitchen sink or laundry tub, in a wet basement or near a swimming pool.
-Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
-Do not use the telephone to report a gas leak in the vicinity of the leak.
-Use only the power cord and batteries indicated in this manual. Do not dispose of batteries in a fire. They may explode. Check with local codes for possible special disposal instructions.
SAVE THESE INSTRUCTIONS"

Telephone line cord

"CAUTION: To reduce the risk of fire, use only No. 26 AWG or larger UL Listed or CSA Certified Telecommunication Line Cord"

<center>END OF THIS DOCUMENT</center>