VOIP GATEWAY

OR SERIES USER MANUAL

WAN WLAN LAN PHONE LINE

Operation Manual V1.1

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1. Introduction

Product Overview

The stand-alone OR SERIES 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. As a standard user agent, it is compatible to all well-known Soft Switches and VSP(Voice Service Provider)/ SIP proxy servers

OR SERIES can be seamlessly integrated to existing network by connecting to a phone set, 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 get the convenient of VoIP and reduce the cost of international and long distance calls.

In addition, the in-built router 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 OR SERIES reachable by its domain name where the ISP dynamically assigns the IP address.

OR SERIES can be assigned with a fixed IP address or by DHCP, PPPoE. It adopts the G.711, 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, OR SERIES can automatically divert the FXS end to the PSTN network on the PSTN 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 Connections and Description

The diagram shows how OR SERIES connects to other devices in your network.

OR SERIES

Front Panel
OR201LW

Power: Power LED. A steady light indicates a proper connection to a power source.

Prov./Alm.: A blinking light indicates the VoIP Gateway is attempting to connect with the Provisioning server. Once the service connects, the LED will turn off. The LED will light solid if the self-test or boot-up fails.

Reg.: The Register LED will turn on when the VoIP Gateway is connected to a VoIP service provider. The LED will turn off if not connected to a service provider.

WAN: When a connection is established the 10 or 100 LED will light up solid. The LED will blink to indicate activity. If the 10 or 100 LED does not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

WLAN: A steady light indicates a wireless connection. A blinking light indicates that the VoIP Gateway is receiving/transmitting from/to the wireless network.

LAN(L1-L4): When a connection is established the 10 or 100 LED will light up solid on the appropriate port. The LEDs will blink to indicate activity. If the 10 or 100 LED does not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

Phone: This LED displays the VoIP status and Hook/Ringing activity on the phone port that is used to connect your normal telephone(s). If a phone connected to a phone port is off the hook or in use, this LED will light solid. When a phone is ringing, the indicator will blink.

Line: Light on means the line is in use (off-hook), and vice versa.

Model Description

2S1LW: It includes 2FXS+1LifeLine+Wireless Network. FXS stands for Phone 1-2 which are connected to your analog telephone, and Life Line stand for Line port which is connected to your original telephone line on the wall jack with RJ-11 cable. Phone 1 will be relayed to Line port when the user enter the feature code (refer to **Force Calling Thru PSTN code** function) before FXS dials out via PSTN line or for emergency calls in the occasion of a power failure. With wireless function enabled, you can easily build a wireless network.

Rear Panel

Line: Connect to your original telephone line on the wall jack with RJ-11 cable.

Phone Port (1-2): Connect to your phones using standard phone cabling (RJ-11).

LAN: Connect to your Ethernet enabled computers using Ethernet cabling.

WAN: Connect to your broadband modem using an Ethernet cable.

Power Receptor: Receptor for the provided power adapter.

Ground: A conducting connection with the earth. Connect with the ground so as to make the earth a part of an electrical circuit using metal wire.

Antenna: Connect to a wireless network.

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.

Use Reset Button to restore factory default settings:

1. **Power on.**
2. **Press and hold the reset button for 5 seconds.**

Release the reset button. Factory settings will be restored.

2. Installation and Applications

The network interface is divided into 3 basic modes as described below:

- OR SERIES can be assigned with a Public IP Address
- OR SERIES can be built under the existing NAT
- OR SERIES can be assigned with a Public IP address and serves as a Bridge device

OR SERIES Assigned with a Public IP Address

OR SERIES 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).

OR SERIES IP Settings	Need to be set up as static IP, DHCP, or PPPoE
NAT/STUN Settings	Unnecessary (Disabled)
DDNS Settings	Unnecessary (Disabled)

OR SERIES in a NAT network

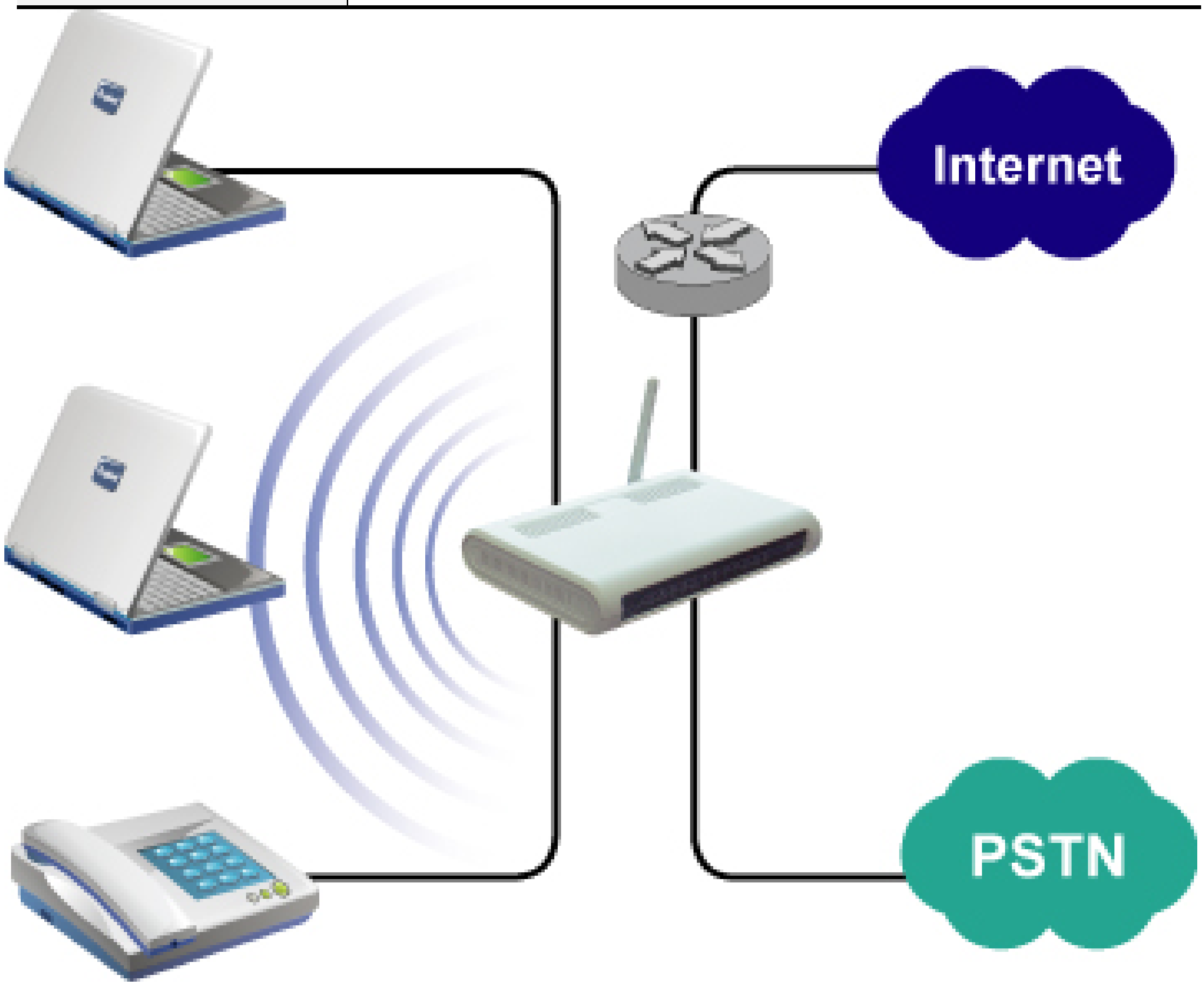
OR SERIES uses a virtual IP address and the IP sharing function of other systems to connect to the Internet.

LAN IP address of IP sharing	Please avoid IP address 192.168.8.1-192.168.8.254 (You may need to change the settings of IP sharing or change SIP series Gateway LAN Port IP address)
OR SERIES IP Settings	Set as static IP address, and assign the LAN IP address of the IP sharing to the Default Gateway.

OR SERIES assigned with a Public IP Address and serving as a Bridge

OR SERIES 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.

OR SERIES 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



3. Setting the OR SERIES through IVR

VoIP transmits voice data (packet) via the Internet to achieve telecommunications. This means that the telecommunication quality is closely related to the whole network environment. If any one of the telecommunicating parties has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, an excellent telecommunication can only be created when OR SERIES is connected to the Internet and when network environment is stable.

Preparation

- Install the OR SERIES according to instructions. Connect the power supply, telephone set, telephone cable, and network cable properly as described in Chapter 2.
- If a static IP is used, confirm the desired 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 users wish to build OR SERIES under the NAT, OR SERIES WAN Port IP address and LAN Port should not use the same range. This is to avoid network failures.

IVR (Interactive Voice Response)

OR SERIES 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 settings, make sure the new settings are saved. This is so that the new settings will take effect after OR SERIES is restarted.

Instructions

- FXS Port: When you have set the password in WEB-GUI with English character. To access OR SERIES IVR function is different. Instead of **[password]#. You should press ***[password]#. The character to number conversion can be acquired from **PPPoE Character Conversion Table**.

Example:

1. The factory default code is blank. Enter **<u>**#</u>**. You are now in IVR setting mode, enter the IVR function code. Please refer to **IVR Function Table** for IVR function code.
2. if the password is **1234**, then enter **<u>**1234#</u>**.
3. If your password is **abc123** then you access IVR by pressing <u>***414243010203#</u> .

- FXO Port: To use IVR functions, dial the phone number of FXO Port using an external line. You will hear the prompt "enter value", and then enter a PIN number. The factory default code is blank. Enter "**#" as above. You are now in IVR setting mode.
- Once the first setting or query has been completed, you will hear a dial tone. Then use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

Example:

1. Enter **<u>**#</u>** . You are now in IVR setting mode.
2. Enter **<u>101</u>** (to query IP address) . OR SERIES responds with an IP address.
3. You can continue with more settings or queries: enter **<u>111</u>** (to set IP address) →enter **<u>192*168*1*3</u>** (IP number).

Save Settings

After entering IVR mode, dial **<u>509</u>** (Save Settings). Wait for about 3 seconds and after hearing a confirmation tone "1", hang up the phone. Please reboot OR SERIES to enable the new settings.

To inquire about current OR SERIES's WAN Port IP address

After entering IVR mode, dial **<u>101</u>**. OR SERIES will repeat the current WAN Port IP address. If OR SERIES does not repeat the IP address, it indicates that OR SERIES is not currently connected to the Internet. Please check if the cable connection, account number, and password are correct.

IVR Functions Table:

Function Code	Description	Remark
111/101	Set/Query WAN Port IP address	Use in conjunction with function code **114**, select 1 for a Static IP function.
112/102	Set/Query WAN Port Subnet Mask	
113/103	Set/Query WAN Port Default OR SERIES	
114/104	Set/Query WAN Port IP Type (1: Static IP, 2.DHCP, 3.PPPoE)	
116/106	Set/Query Phone Book Manager Server IP address	Must use 116/106, 117/107 in conjunction with each other.
117/107	Set/Query whether or not to login Phone Book Manager (0: Disable 1:Enable)	
066	Querying the connection to Phone books manager	
118	Restart	
121	Setting PPPoE Account	Use in conjunction with function code **114,** select 3 for a PPPoE function
122	Setting PPPoE Password	
311/301	Set/Query LAN Port IP address	
131/132	Play/Record greeting message	**OR SERIES ONLY**
133	Saving greeting message	**OR SERIES ONLY**
215/205	Set/Query OR SERIES Telephone Number (Representative Number)	
216/206	Set/Query the extension number of Line 1.	
109	Restoring factory default setting of IP	The default of Static IP IP: 192.168.1.2 Mask: 255.255.255.0 Gateway: 192.168.1.254
409	Restoring factory default settings	
509	Save settings	
900	Set IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Soft Upgrade	

IP Configuration Settings—Setting IP Configuration of WAN Port

Static IP Settings

Note: Before setting Static IP, you must have IP address (111), Subnet Mask (112) and Default Gateway (113) provided by your local Internet Service Provider (ISP).

Function	Command
Select a Static IP	• After entering IVR mode, dial **114**. • After hearing "Enter value", dial **1** (select static IP)
IP address Settings	• After entering IVR mode, dial **111**. After hearing "Enter value", enter your IP address and **#** (speed up dialing). 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 and **#** (speed up dialing). Example: If the mask value is 255.255.255.0, dial 255*255*255*0#.
Default Gateway Setting	• After entering IVR mode, dial **113**. After hearing "Enter value", enter your default OR SERIES's IP address and **#** (speed up dialing). Example: If the Default Gateway is 192.168.1.1, dial 192*168*1*1#.
Save Settings and Restart	• Dial **509** to save settings. • Dial **118** to reboot OR SERIES. • Wait for about 40 seconds for restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, OR SERIES has not been successfully connected to the Internet, please check if the cable connection and IP address are correct.

Dynamic IP (DHCP) Settings

- After entering IVR mode, dial **114**.
- You will hear "Enter value",
- Dial **2** to select DHCP.
- Dial **509** to save settings.
- Dial **118** to reboot OR SERIES.

Wait for about 40 seconds for restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, OR SERIES has not been successfully connected to the Internet, please check if the cable connection is correct.

ADSL PPPoE Settings

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

Select a PPPoE

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

Set PPPoE account

- After entering IVR mode, dial **121**.
- 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 #.

Please note that 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 **122**.
- 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 **509** to save settings.
- Dial **118** to reboot OR SERIES.
- Wait for about 40 seconds for restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, OR SERIES has not been successfully connected to the Internet, please check if the cable connection, account, or password are correct.

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Recorded Voice File

- OR SERIES allows users to record their incoming call greeting messages, when calling via FXO.
- After entering IVR mode, dial **132**. After hearing "Enter value", record the incoming call greeting message. To end recording, simply hang up.
- After recording, to listen to the recorded message, press **131**. Press **133** to save the message.

PPPoE Character Conversion Table

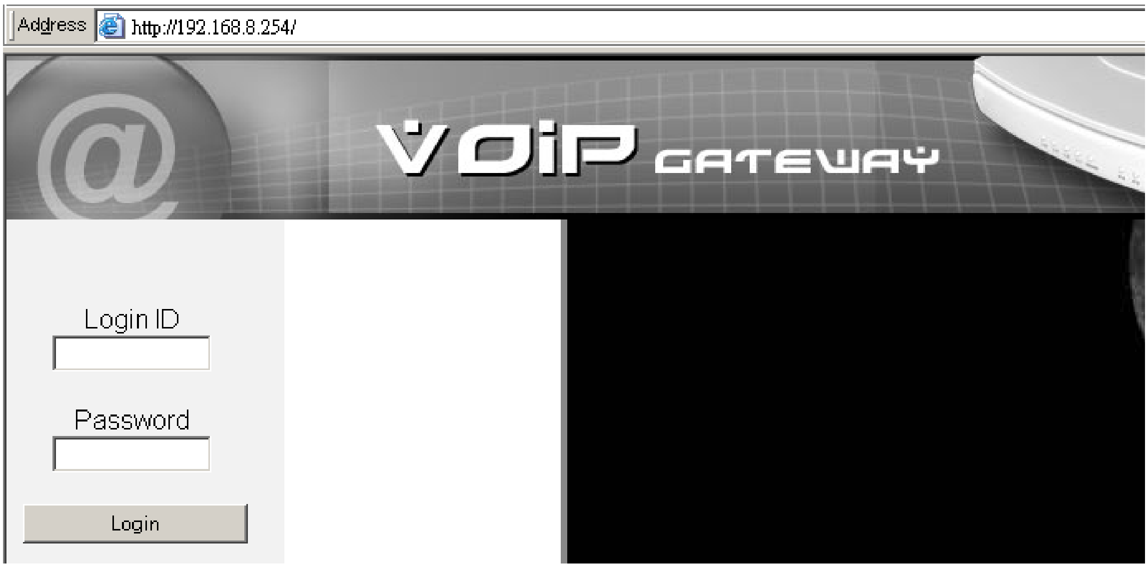
Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	Input Key
0	00	A	11	a	41	@	71
1	01	B	12	b	42	•	72
2	02	C	13	c	43	!	73
3	03	D	14	d	44	"	74
4	04	E	15	e	45	$	75
5	05	F	16	f	46	%	76
6	06	G	17	g	47	&	77
7	07	H	18	h	48	'	78
8	08	I	19	i	49	(	79
9	09	J	20	j	50	)	80
		K	21	k	51	+	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 OR SERIES with WEB Browser

OR SERIES allows users to make settings with a web browser. Activate your browser, and then enter OR SERIES's IP address (e.g. http：//192.168.8.254.) in the Location (for IE) or Address field and press Enter. And you will see the WEB page as following figure. You can also dial **101** on your phone's keypad to inquire the current WAN Port IP address. The factory default LAN Port IP address is 192.168.8.254.

Instructions

- Open a web browser.
- Enter OR SERIES's LAN Port IP address (Default is 192.168.8.254) in Address field (for IE) and make sure your PC is correctly connected to OR SERIES and IP addresses are also in the same network.
- The following registration screen will appear (The factory default settings for **Login ID** and **Password** are left blank).
- Change the default settings of Administrator's Name, Password and Web UI Login ID, Password in **Login Account**.
- After completing and confirming the settings, some of the settings will take effect immediately. But network related settings would take effect after OR SERIES is restarted. Please go to **System Operation** to save the settings before restarting OR SERIES.



For security concern, OR SERIES only accepts one user to login WEB UI for configuration at a time. Please remember to logout or restart OR SERIES before leaving.

Basic Network Settings

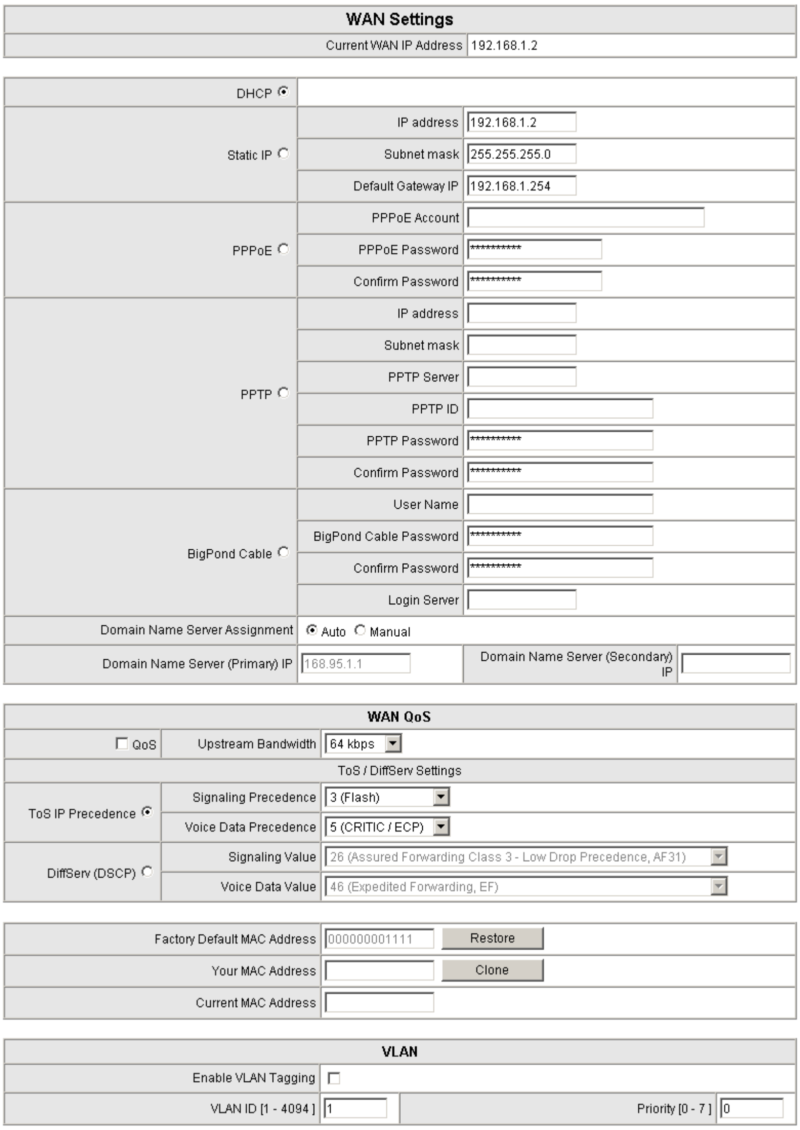
WAN, **SIP** and **Phone Book** are basic Network settings. You have to choose one of SIP and Phone Book for registration. It is recommended to use SIP if you're not sure which one to use. After completing these settings, OR SERIES will be able to make VoIP calls.

WAN

WAN Configuration includes the method of obtaining IP, the setting of DNS (Domain Name Server), etc.

Setup Hint:

1. Choose the correct access type that your ISP supports.
2. Set DNS (Domain Name Server) to Auto if you don't know the DNS server address.
3. WAN QoS, Clone MAC and VLAN are optional.



WAN Settings	
Current WAN IP Address	192.168.1.2

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

WAN QoS		
☐ QoS	Upstream Bandwidth	64 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)

Factory Default MAC Address	000000001111	Restore
Your MAC Address		Clone
Current MAC Address		

VLAN			
Enable VLAN Tagging	☐		
VLAN ID [1 - 4094]	1	Priority [0 - 7]	0

Current WAN IP Address	(N/A)

It is the IP address of WAN port.

When you use DHCP or PPPoE to obtain IP address, you can check the **Current WAN IP Address** field to know if OR SERIES has obtained IP address. N/A is no IP address.

IP Configuration

There are five methods of obtaining a WAN port IP address:

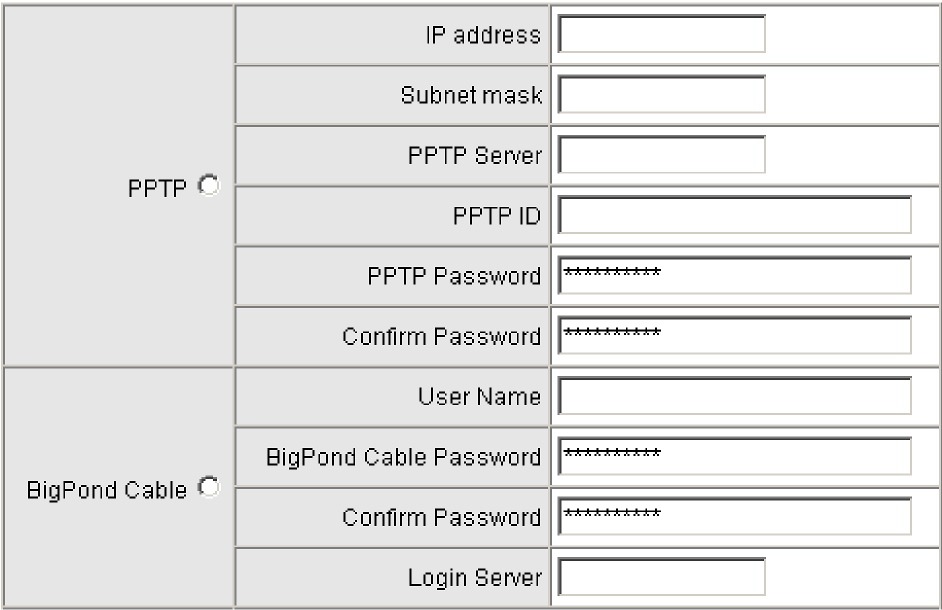
1. DHCP, means a Dynamic IP (Cable Modem)
2. Static IP
3. PPPoE (Dialup ADSL)
4. PPTP.
5. BigPond Cable

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

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	***********

Item	Description
DHCP	This is the default Internet access type. It will obtain IP address from DHCP server of ISP.
Static IP	If OR SERIES is connected to a router that request OR SERIES to have a static IP address, fill in the proper IP address, Subnet Mask and Default Gateway (IP address of the router).
PPPoE	Enter PPPoE account and password and make sure they are correct.

IP Configuration (continued)



Item	Description
PPTP	Enter IP address, Subnet mask, PPTP server address, PPTP ID and Password. It only obtains an IP address from PPTP server and does not provide VPN function.
BigPond Cable	Enter user name and password. Login Server is option.

Domain Name Server (DNS)

OR SERIES will look up the IP address from the DNS provided by ISP while it is accessing another VoIP devices or computer with a hostname. In most cases ISP servers will assign DNS information to OR SERIES automatically.

Note: Without correct DNS setting OR SERIES may not be able to provide services.

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

Item	Description
Domain Name Server Assignment	Auto : OR SERIES uses DNS IP automatically provide by ISP. Manual : Use it if OR SERIES has a static IP address
Domain Name Server IP	Enter correct DNS server address

VLAN

It is optional. It works with the Router or Switch that supports VLAN.

Note: Please do not change anything here unless requested by your ISP.

VLAN			
Enable VLAN Tagging	☐		
VLAN ID [1 - 4094]	1	Priority [0 - 7]	0

Item	Description
Enable VLAN Tagging	It is to tag the packets for VLAN Router or Switch identifying.
VLAN ID	It is to assign uniquely a user-defined ID to each packet.
Priority	It is the proprietary to VLAN Router or Switch.

WAN QoS

It is effective when OR SERIES is as a Bridge. Using QoS is able to ensure that voices have higher priority than data flow, and it also restricts upstream data flow.

WAN QoS		
☐ QoS	Upstream Bandwidth	64 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)

Item	Description
QoS	It is to set an external bandwidth to ensure sound quality during transmission (When this function is enabled, the voice packet has the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission).
ToS (Type of Service)/ DiffServ(DSCP)	The voice packet has the highest priority to ensure telecommunication quality, and the larger the value you set, the higher priority you will get.

Clone MAC

Some **I**nternet **S**ervice **P**roviders (ISP) assigns the IP via the MAC (Media Access Control) Address. Click the **Clone** button to copy the MAC address of the Ethernet Card installed in the computer used to configure the device. It is only necessary to fill in the field if required by your ISP.

Your MAC Address will be blank as you log in through WAN port.

Note: Please do not change anything here unless requested by your ISP.

Factory Default MAC Address	000000001111	Restore
Your MAC Address		Clone
Current MAC Address		

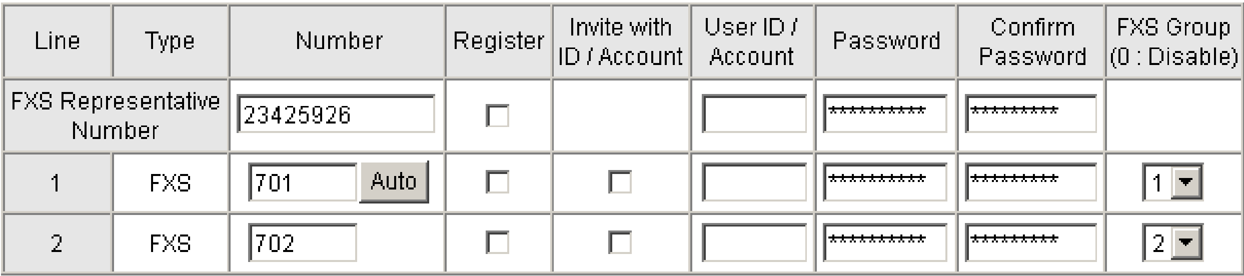
SIP

In this section, you should have one or more VoIP service accounts from **V**oice **S**ervice **P**rovider(VSP) and enter the related parameters of VSP.

Setup Hint:

1. Enter the SIP telephone number.
2. Tick register and invite with ID/Account.
3. Enter user ID/Account and password.
4. Enter the VSP IP address or URL (Uniform Resource Locator) and VSP listen port number.
5. Enter SIP domain if the VSP address is not IP.
6. OutBound Proxy is optional.

Accounts Settings



Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password	FXS Group (0 : Disable)
FXS Representative Number		23425926	☐			**********	*********	
1	FXS	701 Auto	☐	☐		**********	*********	1
2	FXS	702	☐	☐		**********	*********	2

Item	Description
Number	Enter the SIP telephone number assigned by your VSP
Register	Tick the check box to register the number before making calls.
Invite with ID / Account	Tick the check box if SIP server requests authentication.
User ID / Account Password	Authentication information required by VSP
FXS Group	Select group-hunting priority. When there is an incoming call, OR SERIES will automatically assign an unassigned call according to Hunting Priority. If Line 2 does not want to be set as an assigned line to receive any inbound calls, set it to “0”.

Note: There are two ways to register if you have one more accounts.

- Registration by each line: If your VoIP account and password are individual, the settings should be as below.

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password	FXS Group (0 : Disable)
FXS Representative Number		23425926	☐		nekol	**********	**********	
1	FXS	701 Auto	☑	☑	nekol1	**********	**********	1
2	FXS	702	☑	☑	nekol2	**********	**********	2

- Registration by FXS Representative Number: If you have one VoIP account and password, the settings should be as below.

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password	FXS Group (0 : Disable)
FXS Representative Number		23425926	☑		nekol	**********	**********	
1	FXS	701 Auto	☐	☐	nekol1	**********	**********	1
2	FXS	702	☐	☐	nekol2	**********	**********	2

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
1	FXS	701	☐	☐		**********	**********

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
1	FXS	701	☐	☐		**********	**********
2	FXO	702	☐	☐		**********	**********

Item	Description
Number	Enter the SIP telephone number assigned by your VSP
Register	It is to register this number before making calls
Invite with ID / Account	Tick the check box if SIP server request authentication.
User ID / Account Password	Authentication information required by VSP

VSP (Voice Service Provider) Settings

Note: If you fail to make a call, please contact your VSP.

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

Item	Description
Use DNS SRV	Tick the check box to make OR SERIES register to VSP.
DNS SRV Auto Prefix	The default is that OR SERIES will use ***_sip._udp.domain.com*** to query IP. If you untick the check box, OR SERIES will use ***domain.com*** to query IP.
Proxy Fallback Interval	Defines the time that OR SERIES registers to the main server if OR SERIES has registered to the secondary server.

☐ 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	☐

Item	Description
Enable Support of SIP Proxy Server / Soft Switch	Tick the check box to make OR SERIES register to VSP.
Enable SIP Proxy 1	SIP Proxy 1 is the main server. When SIP Proxy 1 and 2 are enabled, OR SERIES will register to SIP Proxy 2 which is a backup server after all lines are failed to register to SIP Proxy 1.

Item	Description
Proxy Server IP/Domain	Enter the SIP Server IP address or URL (Uniform Resource Locator)
Proxy Server Port	Enter the Proxy Server listen port number (The default value is 5060).
Proxy Sever Realm	Enter the correct registered Proxy Server Realm name to avoid registration failure. Set it by default if you are not sure.
TTL (Registration interval)	The interval that OR SERIES will report to the Proxy Server periodically. Set it by default if you are not sure.
SIP Domain	Enter SIP Domain (URI) if required by VSP(Voice Service Provider).
Use Domain to Register	Tick the check box to make OR SERIES register with SIP Domain; otherwise it will register with SIP Server IP address.

Outbound Proxy

This is optional. An outbound proxy server handles SIP call signaling as a standard VSP would. Furthermore, it receives and transmits phone conversation traffic(media) between two talking VoIP devices. This option tells OR SERIES to send and receive all SIP packets to the destined outbound proxy server rather than the remote VoIP device. This might help VoIP calls to pass through any NAT protected network without additional settings or techniques.

Note: Make sure your Voice Service Provider requires this feature before enable it. VSP gives parameters.

All Call through OutBound Proxy	☐		
OutBound Proxy IP / Domain		OutBound Proxy Port [1 - 65535]	5060

Item	Description
All Call through OutBound Proxy	Tick the check box to make OR SERIES register to OutBound Proxy Server / Soft Switch.
OutBound Proxy IP/Domain	Enter the OutBound Proxy IP address or URL (Uniform Resource Locator).

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 OR SERIES 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 IAD 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 IAD? 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].

Note: If you fail to make a call, please contact your VSP.

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

Item	Description
International Call Prefix Digit	Enter the International call prefix.
Country Code	Users please select the desired country code.
Long Distance Call Prefix Digit	The long-distance prefix digit for making a long-distance call.
Area Code	Enter the area code.

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

Item	Description
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. **(Only OR SERIES has this function)**.
ENUM Header Exception	Defines OR SERIES not to change the prefix..

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 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 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.

Phone Book

Some peer information needs to be added to this section before OR SERIES makes peer-to-peer calls.

Phone Book Manager: VoIP devices register to Phone Book Manager. When you make calls from OR SERIES to the peer VoIP device, it will get the number and IP from Phone Book Manager.
Phone Book: Some peer information is added to Phone Book. OR SERIES can set up and store 100 phone numbers into Phone Book and provide an IP address query when calling to other VoIP devices.

Using Phone Book Manager

Register to Phone Book Manager	☐	VoIP failure announcement	☐
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

Item	Description
Register to Phone Book Manager	Tick the check box to register to the Phone Book Manager.
VoIP failure announcement	If OR SERIES fails to register to the Phone Book Manager, it will play a voice announcement when FXS is off-hook.
Gateway Name for Phone Book Manager	The alias registered with the Phone Book Manager.
Phone Book Manager Login Password	Enter the registered password that is the same with Phone Book Manger.
Phone Book Manager IP / Domain	Enter the IP address for the Phone Book Manager. It supports URL (Uniform Resource Locator).
Phone Book Manager Listen Port	The protocol communication port for transmitting signals between the Phone Book Manager and OR SERIES.

Note: Make sure that Phone Book Manager Login Password and Phone Book Manager Listen Port are same as that of the Phone Book Manager.

Using Phone Book

OR SERIES can set up and store 100 phone numbers to a phone book. If there is no Phone Books Manager exiting in private network, all OR SERIESs in a group have to set up each gateway's number one by one to communicate with each other.

Note: If the VoIP peer is in a NAT network, the listen port may vary or unreachable depend on settings of that NAT router.

Phone Book 1 - 5   6 - 10

#	Gateway Name	Gateway Number	IP / Domain Name	Port
1				5060
2				5060
3				5060
4				5060
5				5060

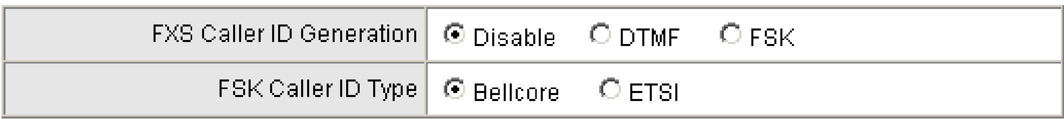
Item	Description
Gateway Name	Enter an easy-to-remember name to identify each VoIP device listed in the phone book. This parameter is optional.
Gateway Number	Enter the telephone number of other VoIP device.
IP/Domain Name	Enter the IP address or URL of other VoIP device.
Port	Enter the listen port of other VoIP devices.

Basic Voice Services

OR SERIES supports some voices such as display Caller ID, call forwarding, call hold, call transfer, call-waiting, three-way calling, Emergency No., etc.

Caller ID

In this section, it allows you to set Caller ID generation.. There are two type of FSK Caller ID. Choose the proper type for you.



Item	Description
FXS Caller ID Generation	Tick the check box to display the phone number of the calling party on your phone set when there is an incoming call.
FXO Caller ID Detection	Tick the check box to detect Caller ID delivered from PSTN port.
Detection Level	It is the gain volume that could be adjusted while detecting caller ID.
FSK Caller ID Type	In most cases, Bellcore is preferred in North America and ETSI in Europe.

Note: You have to enable "Hot Line->Wait for Caller ID before FXO / Trunk pick up" to ensure detect Caller ID correctly.

Transit In Caller ID Strip / Replace

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

Note: Available in OR SERIES only.

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

Item	Description
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 Caller ID while making calls to Internet by FXO. It will change two places of displaying the caller id. One is From-Header Display Name, and the other one is Remote Party ID Display Name.

Hot Line

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	PSTN Busy-Out With FXS Pick-Up [0=disable, 5 - 20 s]	VoIP Call Allow PSTN In	PSTN Call Allow VoIP In
1	☑	FXS	☐		0		☐	☐
2	☑	FXS	☐		0			
3		PSTN				9		

Wait for Caller ID before FXO / Trunk pick up	☑

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	PSTN Busy-Out With FXS Pick-Up [0=disable, 5 - 20 s]	VoIP Call Allow PSTN In	PSTN Call Allow VoIP In
1	☑	FXS	☐		0		☐	☐
2		PSTN				9		

Wait for Caller ID before FXO / Trunk pick up	☑

Item	Description
Enable	All lines are enabled by default. Untick the check box to disable it if the line is not in use (Pause Function).
Hot Line	While picking up the phone, OR SERIES will automatically dial the assigned Hot Line number. At the moment, dialing any number out is denied.
Hot Line No.	Enter the Hot Line number for an automatic dial.
Warm Line (Hot Line Delay)	A user can dial any number within the time. After the time expires, OR SERIES will divert incoming calls from an outside line to the Hot Line Number.
PSTN Busy-Out with FXS Pick-up	OR SERIES will reject a call from FXO while FXS is getting DTMF. If you would like to disable it, set the value as “0”.

Item	Description
VoIP Call Allow PSTN In	As making a VoIP call, a waiting call from PSTN is allowed. Before starting, do the following settings first: 1. Tick the check box to enable **VoIP Call Allow PSTN In**. 2. Tick the check box to enable **Call Hold**. (Calling Feature → Call Hold) 3. Set **PSTN Busy-Out With FXS Pick-Up** as 0.
PSTN Call Allow VoIP In	As making a PSTN call, a waiting call from VoIP is allowed. Before starting, do the following settings first: 1. Tick the check box to enable **PSTN Call Allow VoIP In**. 2. Tick the check box to enable **Call Hold** and **Call Waiting**. (Calling Feature → Call Hold and Call Waiting)
Wait for Caller ID before FXO / Trunk pick up	It is to detect caller ID from PSTN port.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]
1	☑	FXS	☐		0
2	☑	FXS	☐		0

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]
1	☑	FXS	☐		0

Item	Description
Enable	All lines are enabled by default. Untick the check box to disable it if the line is not in use (Pause Function).
Hot Line	While picking up the phone, dialing any number out is denied, since OR SERIES will automatically dial the assigned Hot Line number.
Hot Line No.	Enter the Hot Line number for an automatic dial.

Item	Description
Warm Line (Hot Line Delay)	A user can dial any number within the time. After the time expires, OR SERIES will divert incoming calls from an outside line to the Hot Line Number.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	Dial-Out Prefix	FXO Line Default Dial-Out
1	☑	FXS	☐		0		
2	☑	FXO	☐		0		

Item	Description
Enable	All lines are enabled by default. Untick the check box to disable it if the line is not in use (Pause Function).
Hot Line	While picking up the phone, dialing any number out is denied, since OR SERIES will automatically dial the assigned Hot Line number if set **Warm Line** to 0.
Hot Line No.	Enter the Hot Line number for an automatic dial.
Warm Line (Hot Line Delay)	A user can dial any number within the time. After the time expires, OR SERIES will divert incoming calls from an outside line to the Hot Line Number.
Dial-Out Prefix	It is the number dialed automatically by FXO port before the FXO interface diverts a VoIP call to PSTN.
FXO Line Default Dial-Out	Before starting to configure, you should set **FXO Line VoIP call in option** to **Default Dial-Out**. When FXO receives a call from VoIP, it will dial to PSTN with the default number.

FXO Line VoIP call in option	Caller Indicate Dial-Out		
Trunk Incoming Prompt Voice	◉ Default Greeting ○ Custom Greeting ○ Dial Tone		
Custom Greeting Upload / Backup	[ ] Browse... Upload Backup		
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		
Detect FXO Line Presence	☑		

Item	Description
FXO Hunting VoIP call in option	**Caller Indicate Dial-Out:** When there is a call from WAN interface to FXO port, it will dial to PSTN with the number assigned in SIP packet. **Default Dial-Out:** When there is a call from WAN interface to FXO port, it will dial to PSTN with the number filled in **FXO Line Default Dial-Out** field.
Trunk Incoming Prompt Voice	Select the greeting type. When FXO receives an inbound call, the caller can hear the greeting. (If you would like to record a voice file, you must use the IVR 132 function).
Custom Greeting Upload / Backup	It is to upload or backup the recorded voice file. The format must be G.723.1.
Enable FXO/Trunk Extension Number	When FXO is connected to different PBX or PSTN, or under special circumstances, the caller can choose one of them to call out. It MUST be ticked while registering to a Proxy.
Pick up Line by Dialing Extension Number	When there is a call from WAN interface and assigned FXO extension number, FXO goes off-hook and waits for the caller to dial the number to PSTN. It MUST be enabled while registering to a Proxy.
Wait for Caller ID before FXO / Trunk pick up	Detect caller ID from FXO port.
Transit in Busy Tone Limit	Define the duration of a busy tone before FXO hook-on. Notify the caller from PSTN that this call is finished.
Detect FXO Line Presence	Tick the check box to detect the line presence that FXO port is connected to PBX or a PSTN line. Untick the check box to disable this function if it mis-detect line presence on FXO port while ringing.

Calling Features

OR SERIES provides Call Forward, Call Hold, Call Transfer and Call Waiting.
OR SERIES also provides Three-Way Calling based on Nortel Soft Switch. It also works with the conference call supported by VSP.

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)
Line 1	FXS				After[10 - 60] 20 s				
Line 2	FXS				After[10 - 60] 20 s				

Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID
Line 1	FXS				After[10 - 60] 20 s				

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

Item	Description
Do Not Disturb	Tick the check box to reject all incoming calls from WAN interface. It allows only to make an outgoing call.
Unconditional Forward	All incoming calls will be forwarded to the **Forwarding Number** automatically. If the call is forwarded to FXO port, FXO is off-hook instead of dialing out.
Busy Forward	It is to forward the incoming call to **Forwarding Number** when the line is busy.
No Answer Forward	It is to forward the incoming call to **Forwarding Number** after the time expires without answer.
Call Hold	Tick the check box to enable call hold for specific FXS port.

Item	Description
Call Transfer	Tick the check box to enable call transfer for specific FXS port.
Call Waiting	Tick the check box to enable call-waiting for specific FXS port.
Three-Way Calling / Service ID	It is for conference all based on Nortel Soft Switch and must work with Proxy Server that supports Three-Way Calling service.

Calling Feature Instructions:

- Call Hold: While pressing FLASH button on the phone. The call is held.
- Call Transfer: Ongoing call will be put on hold after FLASH button pressed on local phone set. Meanwhile, the local user can dial out to another number after dial tone observed. After the handset is back on the 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 does not hang up the phone after new call sets up, press FLASH button to switch between the first call and the new call. If a phone set is connected directly to the FXS port of OR SERIES and not functioning to FLASH, please adjust the settings in **Flash Detect Time** in category "**Line Settings**".
- Call Waiting: When you are on the phone and a second call comes in, you will hear "Beep-Beep" tone to notify that there is another call. Press the FLASH button to hold the first call and take the second call. After finishing the second call, press the FLASH button again to take the first call.
- Example of a Three-Way calling:

1. Alex calls Bob, Bob answers the call.
2. Alex presses Flash and calls Coral (Bob is on hold), Coral answers the call.
3. Alex dials *61 and then presses Flash.
4. Thus the conference call is established.

Or

1. Alex calls Bob, Bob answers the call.
2. Coral calls Alex (Call Waiting), presses Flash and talks to Coral.
3. Alex dials *61 and then presses Flash.
4. Thus the conference call is established.

Calling Features - Advanced Setting

OR SERIES 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, OR SERIES will not dial again.

Function Code	Description
*40#	**Call Pickup:** The user can use the function of call pickup to answering others calls. When one of FXS is ringing and there is no one to answer the call. The user can use another FXS port to pick up the ringing call with this function code. For Example: If Alice calls Bob (9901701) who does not answer. Carol can pick up the call by dialing *40 9901701#.
*41#	Automatic Redial: The remote party 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 remote party is available. You are alerted and then pick up the phone to wait for the remote party answering.
*42#	It is to cancel the latest automatic redial function.
*43#	It is to query how long shall OR SERIES wait to redial (ms).
*44#	It is to adjust the duration of waiting for automatic redial. Method: Dial ***44 + Expiry Time#**
*45#	It is to query the duration of waiting for automatic redial (ms).

PSTN Control

Note: Available in OR SERIES only

This rule only applies to one-stage dialing. It is to replace the prefix number before diverting the number to PSTN dial out. It also restricts the number by checking the prefix number.

Example: If you transit out with 01907123456, OR SERIES will replace the number to 190601 907123456. If you transit out with 008621123456, OR SERIES will replace it with 190200 8621123456.

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

Item	Description
Trunk Dial Out Verify Trunk Dial Out Replace	Before the number is diverted to PSTN by FXO port, OR SERIES will verify the numbers in **Trunk Dial Out Verify** filed and replace them with the numbers in **Trunk Dial Out Replace** field.
Trunk Dial Out Deny	OR SERIES will deny the call with the leading number filled in this column.

Emergency No

Emergency numbers is defined here. You can call out to PSTN (Telco line) with the numbers that your VSP does not support (i.e. Toll free service numbers).

Note: Available in OR SERIES only

#	Enable	Scan Code	User Dial Length
1	☐		10
2	☐		10
3	☐		10

Item	Description
Enable	Tick the check box to make this entry effective.
Scan Code	Fill in the leading number for OR SERIES to scan or the full number.
User Dial Length	Set the total digit count of user dialed.

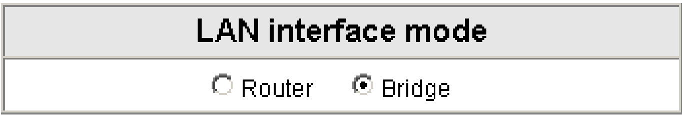
Advanced Network Settings

OR SERIES provides interface for advanced network settings to enhance your network security.

LAN

This is about LAN configuration. There are LAN interface mode that is to set OR SERIES as a router or a bridge, LAN IP and subnet mask, DHCP settings.

LAN interface mode



Item	Description
Router	OR SERIES serves as a router with NAT.
Bridge	OR SERIES serves as a bridge between WAN port and LAN port without NAT. (LAN default gateway will still be accessible for configuration).

Item	Description
Bridge Mode VLAN Tagging	It is to tag the packets for VLAN Router or Switch identifying when OR SERIES serves as a Bridge.
VLAN ID	It is to assign uniquely a user-defined ID to each packet.
Priority	It is the proprietary to Router or Switch.

LAN Settings

Note: OR SERIES LAN port IP address cannot be in the same section as the NAT LAN port IP address.

Example: If the LAN IP address of the Internet Sharing Device is 192.168.8.1, then OR SERIES'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.

LAN Settings			
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0

Item	Description
LAN IP/LAN default Gateway Subnet mask	LAN Port IP address and the subnet mask value. Please note that OR SERIES is built with NAT

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 ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

Item	Description
Enable DHCP Server	Tick the check box to enable DHCP server service of OR SERIES.
IP Pool Starting Address IP Pool Ending Address	The first IP address to be assigned to DHCP clients. 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 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	Auto : Assign DNS obtained from WAN port to the DHCP clients. Manual : Manually assign DNS for DHCP clients.
Domain Name Server IP	It is to manually assign DNS to DHCP client, a correct DNS IP address must be filled.

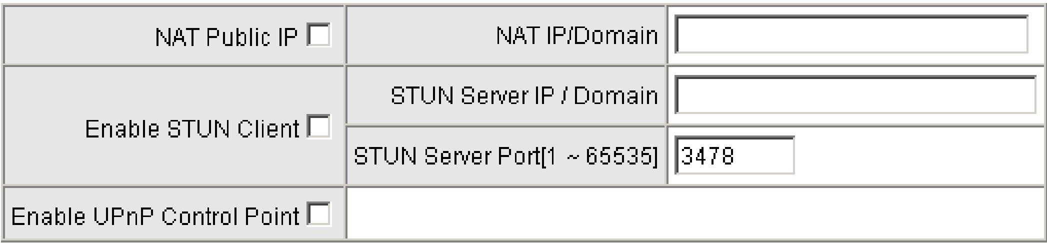
NAT Traversal

If OR SERIES is set up behind an IP sharing device or a router, you can select either the NAT or STUN protocol.

Note: NAT IP/Domain must be the same with **Hostname** (in **DDNS** page), if OR SERIES is behind a NAT Server that uses a dynamic IP and registers to DDNS.

The ports that need to set the Virtual Server Mapping in the NAT server are below.

1. Listen Port (UDP): 5060 is default.
2. RTP Port (UDP): 9000~9001. These ports are used for telecommunication.
3. Http Port (TCP): The default is 80.

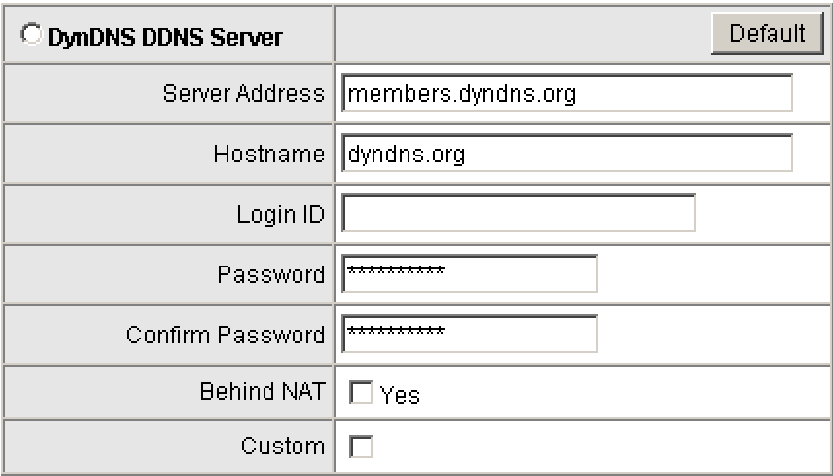


Item	Description
NAT Public IP	Tick the check box to enable NAT.
NAT IP/Domain	Enter the NAT Server IP address (Real External IP address of NAT Server) then fill in the URL (Uniform Resource Locator).
Enable STUN Client	Tick the check box to use STUN protocol prevents problems with setting the IP sharing function, but some NAT do not support this protocol.
STUN Server IP/Domain STUN Server Port	Enter the STUN server IP address and Listen Port number.
Enable UPnP Control Point	It only works when the NAT server supports UPnP. Tick the check box to enable OR SERIES to pass through the NAT server.

DDNS

These settings are only necessary when OR SERIES is set up behind a NAT that uses a dynamic IP address and do not support DDNS.
First of all, you need to apply an account from one of the servers. OR SERIES allows users to choose one of DynDNS, TZO, 3322.org, PeanutHull or a private server.

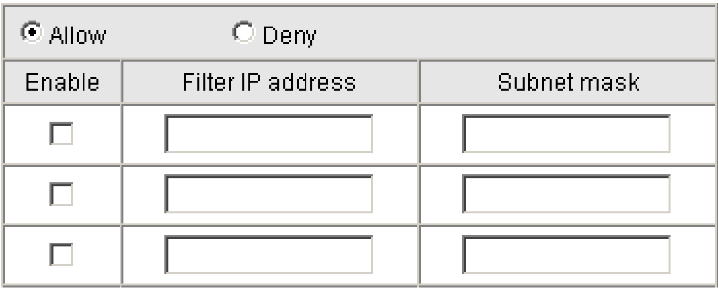
Item	Description
Register to DDNS	Tick the check box to enable DDNS and choose a DDNS Server as below to register.



Item	Description
Server address	Enter the IP address or URL (Uniform Resource Locator) of the DDNS Server.
Hostname	The URL of OR SERIES (or NAT) – provided by a domain name registration providers. (e.g. www.dyndns.org).
Login ID Password	The ID and password are used to login the DDNS server.
Behind NAT	Tick the check box to enable this function only when OR SERIES is set up behind a NAT.
Custom	Only DynDNS has. Tick the check box if you have a custom hostname in DynDNS.

Caller Filter

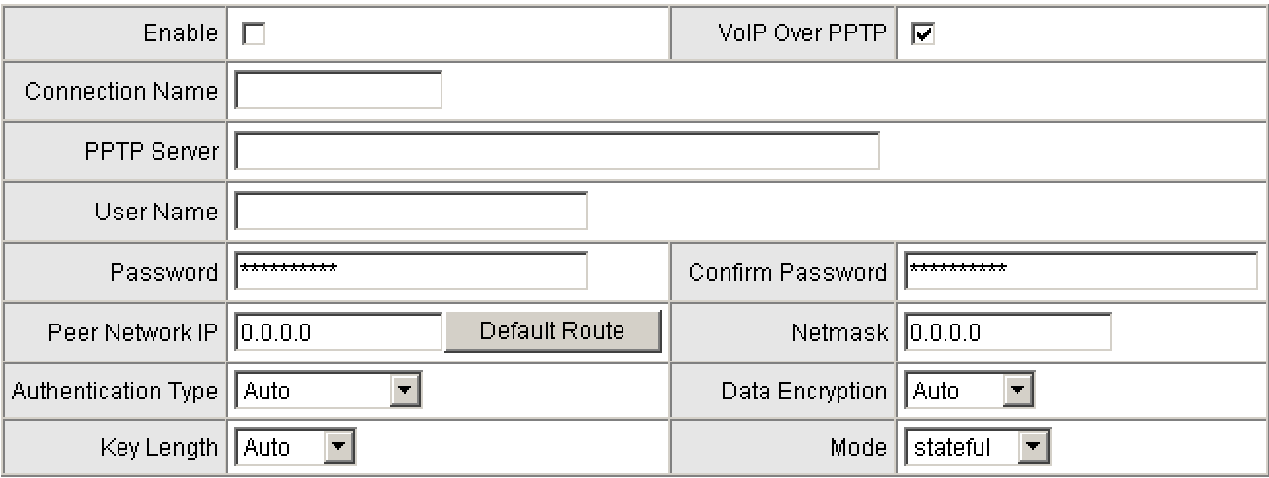
This function is used to allow or deny SIP Invite from the list. The IP address of VSP is allowed while registering to VSP.



Item	Description
Allow Deny	Choose the IP addresses in the table are allowed to call in or deny.
Enable	Tick the check box to make this effective.
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.

PPTP Client

This is optional. ISP gives all parameters.



SIP Advanced

In this section, you can set the listen port and RTP port of OR SERIES.
There are some parameters with VSP (Voice Service Provider).

Session Timer: It is to identify the connection of a session which is defined in RFC 4028.

SIP Timeout Adjustment: It is to set SIP message resend time and maximum response time.

Supplementary Features: Other features work with VSP (Voice Service Provider).

Listen Port UDP [1 - 65535]	5060	RTP Starting Port UDP [1 - 65500]	9000

Item	Description
Listen Port UDP	The listen port of OR SERIES.
RTP Starting Port UDP	The initial value of port number for transmitting voice data among OR SERIES(s). Each line requires 2 ports (RTP/RTCP). It is not necessary to change these. 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.

Session Timer

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

Item	Description
Session Expiration	It is to avoid the billing of abnormal dropping the call because of Internet. The default is disabled.
Session Refresh Request	The method of refreshing for Session Timer.
Session Refresher	The role OR SERIES plays in Session Timer. UAS is an originator, and UAC is a replier.

SIP Message Timeout Adjustment

SIP Message Resend Timer Base [s]	0.5
Max. Response Time for Invite [1 - 20]	8

Item	Description
SIP Message Resend Timer Base	SIP packet will resend if response dose not arrive in the base time set in this column. The max of resend time is 4 sec. It will send again at "base time" *2, and send again at "base time" *2 *2. Resend will stop/restart when total resend 20sec has reached.
Max. Response Time for Invite	If the remote party does not reply in the set time after the first invite, this call is failed.

SIP Proxy Server / Soft Switch Settings

SIP Proxy Server / Soft Switch Settings	
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

Item	Description
VoIP failure announcement	As soon as the registration to proxy server is failed, OR SERIES will drive IVR system to play out failure announcements for the user.
Bind Proxy Interval for NAT	OR SERIES will always send two packets in N seconds to VSP to bind the tunnel. The VSP can always send SIP packets to OR SERIES that is setup behind an NAT.
Initial Unregister	OR SERIES will send un-register packet to VSP as it is initialing.
Support Message Waiting Indication (MWI)	Tick the check box to enable voice mail function. OR SERIES will play a tone to notify user if there are messages in the voice mail.

Item	Description
MWI Subscribe Interval	The subscribe interval is for OR SERIES check of the voice mail.

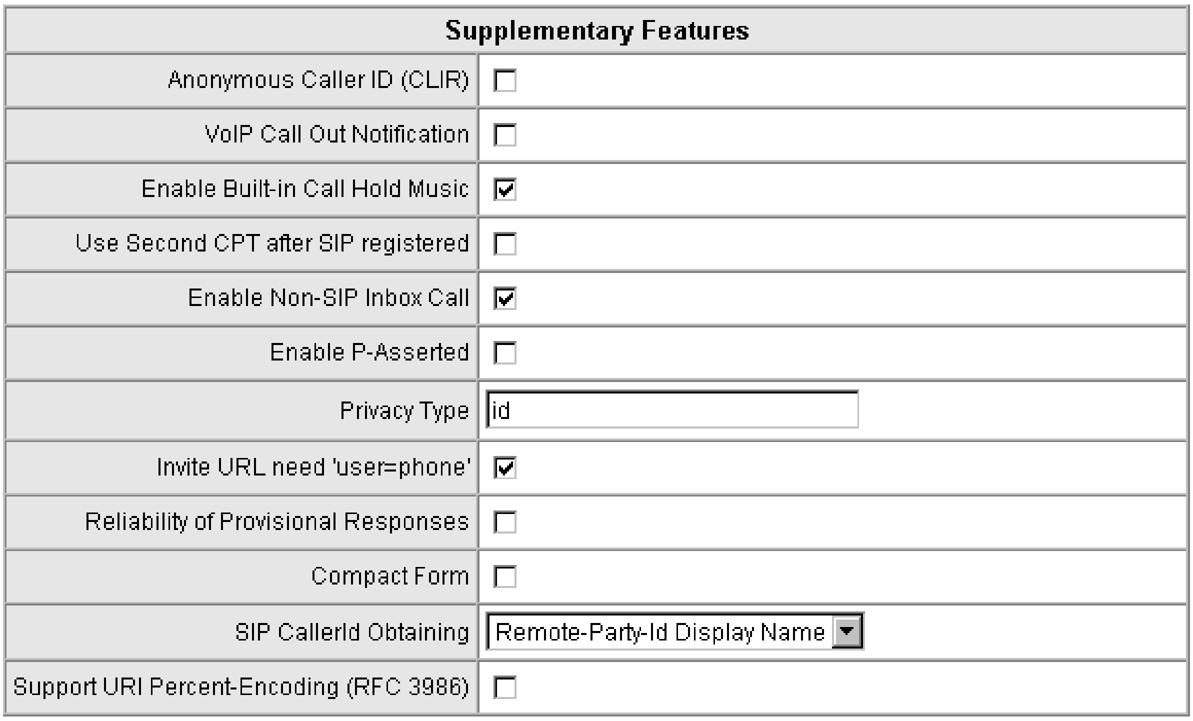
Supplementary Features

Supplementary Features	
Anonymous Caller ID (CLIR)	☐
VoIP Call Out Notification	☐
Enable Built-in Call Hold Music	☑
Use Second CPT after SIP registered	☐
Enable Non-SIP Inbox Call	☑
Delay PSTN Hangup Detection	☑
Enable P-Asserted	☐
Privacy Type	id
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)	☐

Note: Enable Anonymous Caller ID or Anonymous Transit in W/O Caller ID, you may be unable to make a call since OR SERIES doesn't send the number for authorization.

Item	Description
Anonymous Caller ID (CLIR)	Tick the check box to dial out with "anonymous" as caller identification by FXS. Sometimes it may require proxy server to identify by Caller ID, so disable it while the call is failed.
VoIP Call Out Notification	OR SERIES will play a tone to notify the call is through VoIP.
Enable Built-in Call Hold Music	The default setting is that when receiving a call hold request, OR SERIES will play music on hold. Untick the check box to disable the function.

Item	Description
Use Second CPT after SIP registered	This function is usually applied when the user set VoIP as the primary path for outgoing calls and PSTN as the backup. OR SERIES 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 PSTN tones instead of the second set CPT. (for CPT settings, refer **CPT Parameters Table**)
Enable Non-SIP Inbox Call	Tick the check box to disable Non-SIP inbox call if all calls need to go through VSP.
Delay PSTN Hangup Detection	The default is that OR SERIES detects dully if PSTN hangs up. Tick the check box to make OR SERIES detect PSTN status sensitively.
Enable P-Asserted	Tick the check box to use anonymous caller ID for protection if the SIP proxy has this function.
Privacy Type	Privacy requested for Third-Party Asserted.
Invite URL need 'user=phone'	It will contain "user=phone" in Invite Packet. Some Proxy Servers can't accept "user=phone", just disable it.
Reliability of Provisional Responses	Defines a type of SIP responses that provide information on the progress of the request processing. Tick the check box to achieve reliability for provisional responses.
Compact Form	Defines the header packet size will be shortened with signaling compression to enhance bandwidth. Tick the check box to enable this function.
SIP Caller ID 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 some letters as character triplet, consisting of the percent character "%" followed by the two hexadecimal digits representing that octet's numeric value. The unreserved characters that are not encoded are uppercase and lowercase letters, decimal digits, hyphen (or dash), period (or dot), underscore (or underline), exclamation, tilde, asterisk (star or multiplication), single quote, parenthesis, bracket, ampersand, equal, plus sign, dollar sign, comma, semicolon, question mark, slash, colon, at sign and back slash.



Note: Enable Anonymous Caller ID or Anonymous Transit in W/O Caller ID, you may be unable to make a call since OR SERIES doesn't send the number for authorization.

Item	Description
Anonymous Caller ID (CLIR)	Tick the check box to dial out with "anonymous" as caller identification by FXS. Sometimes it may require proxy server to identify by Caller ID, so disable it while the call is failed.
VoIP Call Out Notification	OR SERIES will play a tone to notify the call is through VoIP.
Enable Built-in Call Hold Music	The default setting is that when receiving a call hold request, OR SERIES will play music on hold. Untick the check box to disable the function.
Use Second CPT after SIP registered	This function is usually applied when the user set VoIP as the primary path for outgoing calls and PSTN as the backup. OR SERIES 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 PSTN tones instead of the second set CPT. (for CPT settings, refer **CPT Parameters Table**)
Enable Non-SIP Inbox Call	Tick the check box to disable Non-SIP inbox call if all calls need to go through VSP.

Item	Description
Enable P-Asserted	Tick the check box to use anonymous caller ID for protection if the SIP proxy has this function.
Privacy Type	Privacy requested for Third-Party Asserted.
Invite URL need ‘user=phone’	It will contain “user=phone” in Invite Packet. Some Proxy Servers can’t accept “user=phone”, just disable it.
Reliability of Provisional Responses	Defines a type of SIP responses that provide information on the progress of the request processing. Tick the check box to achieve reliability for provisional responses.
Compact Form	Defines the header packet size will be shortened with signaling compression to enhance bandwidth. Tick the check box to enable this function.
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 some letters as character triplet, consisting of the percent character "%" followed by the two hexadecimal digits representing that octet's numeric value. The unreserved characters that are not encoded are uppercase and lowercase letters, decimal digits, hyphen (or dash), period (or dot), underscore (or underline), exclamation, tilde, asterisk (star or multiplication), single quote, parenthesis, bracket, ampersand, equal, plus sign, dollar sign, comma, semicolon, question mark, slash, colon, at sign and back slash.

Supplementary Features	
Anonymous Caller ID (CLIR)	☐
CLIR At Transit In W/O Caller ID	☐
VoIP Call Out Notification	☐
Enable Built-in Call Hold Music	☑
Use Second CPT after SIP registered	☐
Enable Non-SIP Inbox Call	☑
Enable P-Asserted	☐
Privacy Type	id
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	☐

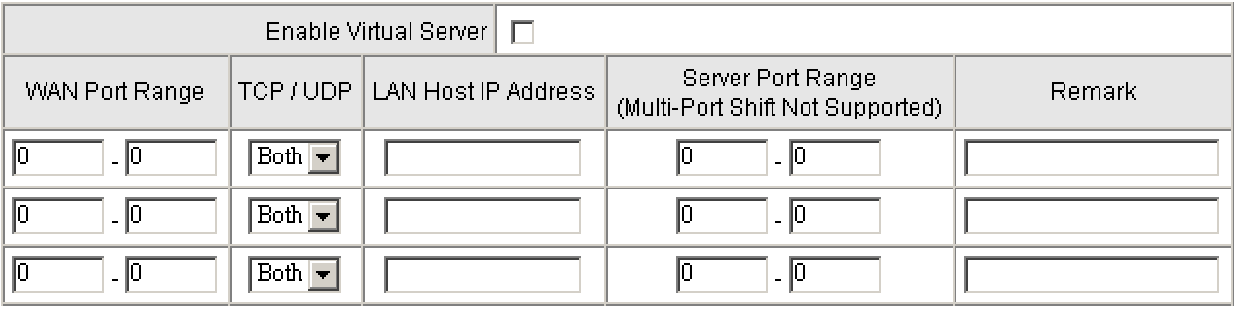
Note: Enable Anonymous Caller ID or Anonymous Transit in W/O Caller ID, you may be unable to make a call since OR SERIES does not send the number for authorization.

Item	Description
Anonymous Caller ID (CLIR)	Tick the check box to dial out with “anonymous” as caller identification by FXS. Sometimes it may require proxy server to identify by Caller ID, so disable it while the call is failed.
CLIR At Transit in W/O Caller ID	Disable it, if FXO detects caller ID from PSTN, OR SERIES will use the detected caller ID as caller identification; if FXO cannot detect caller ID from PSTN, OR SERIES will use “anonymous” as caller identification. When enabled, OR SERIES will always use “anonymous” as caller identification.
VoIP Call Out Notification	OR SERIES will play a tone to notify the call is through VoIP.
Enable Built-in Call Hold Music	The default setting is that when receiving a call hold request, OR SERIES will play music on hold. Untick the check box to disable the function.
Enable Non-SIP Inbox Call	Tick the check box to disable Non-SIP inbox call if all calls need to go through VSP.

Item	Description
Use Second CPT after SIP registered	This function is usually applied when the user set VoIP as the primary path for outgoing calls and PSTN as the backup. OR SERIES 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 PSTN tones instead of the second set CPT. (for CPT settings, refer **CPT Parameters Table**)
Enable P-Asserted	Tick the check box to use anonymous caller ID for protection if the SIP proxy has this function.
Privacy Type	Privacy requested for Third-Party Asserted.
Invite URL need 'user=phone'	It will contain "user=phone" in Invite Packet. Some Proxy Servers can't accept "user=phone", just disable it.
Reliability of Provisional Responses	Defines a type of SIP responses that provide information on the progress of the request processing. Tick the check box to achieve reliability for provisional responses.
Compact Form	It decreases the size of SIP header. Tick the check box to enable this function.
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 some letters as character triplet, consisting of the percent character "%" followed by the two hexadecimal digits representing that octet's numeric value. The unreserved characters that are not encoded are uppercase and lowercase letters, decimal digits, hyphen (or dash), period (or dot), underscore (or underline), exclamation, tilde, asterisk (star or multiplication), single quote, parenthesis, bracket, ampersand, equal, plus sign, dollar sign, comma, semicolon, question mark, slash, colon, at sign and back slash.
Compare SIP 'To' Header for Transit Out	When there is a call from WAN interface to FXO 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 VSP.

Virtual Server

Enable users on Internet to access the WWW, FTP and other services from your NAT. It is also known as port forwarding. When remote users are accessing Web or FTP servers through WAN IP address, it will be routed to the server with LAN IP address.



Item	Description
Enable Virtual Server	Tick the check box to enable virtual server function.
WAN Port Range	Enter the port on WAN.
TCP/UDP	Select the communication protocols used by the server—TCP or UDP.
LAN Host IP Address	Enter IP address that the server provides various services.
Server Port Range	Enter the port used by the server on LAN.
Remark	The space reserved for notations.

DMZ

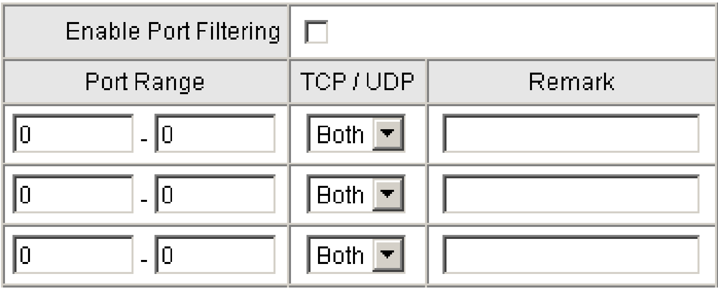
Demilitarized Zone lets the server on the LAN to be directly exposed to the Internet for accessing data. Either this function or the virtual server can be selected for use.

Item	Description
Enable DMZ	Tick the check box to enable this function.
DMZ Host IP Address	Enter the LAN host IP address.

Port Filtering

Port filtering enables you to control all data that can be transmitted in routers.

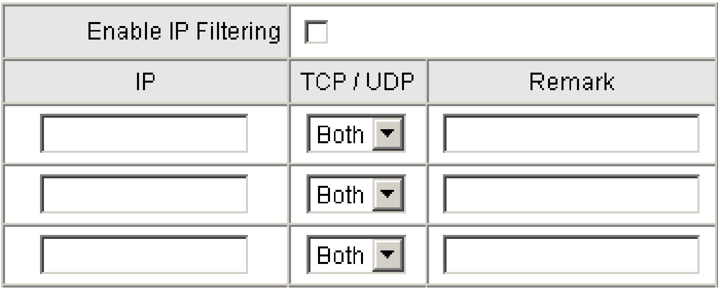
Note: When the port used at the source end is within the limited scope, it will be filtered without transmission.



Item	Description
Enable Port Filtering	Tick the check box to make this effective
Port Range	Set the range of port to be filtered. If set 80 and protocol is Both or TCP, all computers will be unable to use the services of http (port 80) — will be unable to browse normal WebPages.
TCP/UDP	Select the communication protocols used by the server—TCP or UDP.
Remark	The space reserved for notations.

IP Filtering

IP Filtering is to limit intranet users from accessing the Internet.



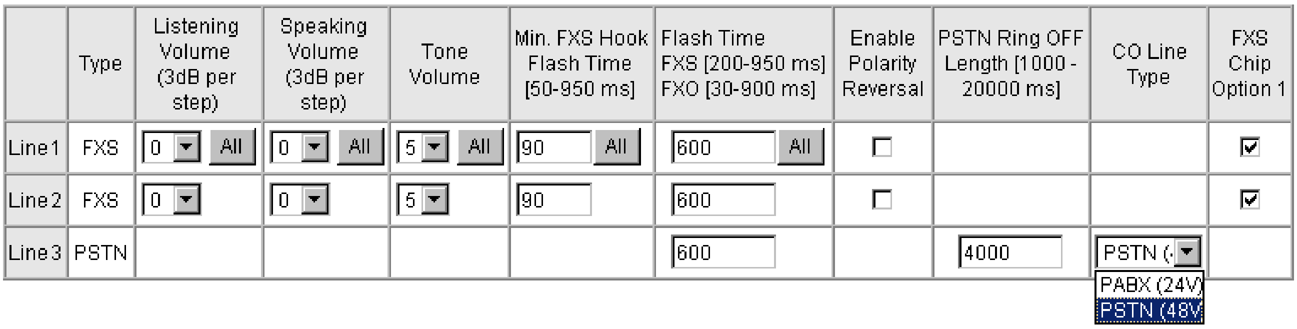
Item	Description
Enable IP Filtering	Tick the check box to make this effective
IP	Enter the IP address that you want to filter; the limited IP address will be unable to transmit the data to the Internet
TCP/UDP	Select the communication protocols used by the server—TCP or UDP.
Remark	The space reserved for notations.

Advanced Voice Services

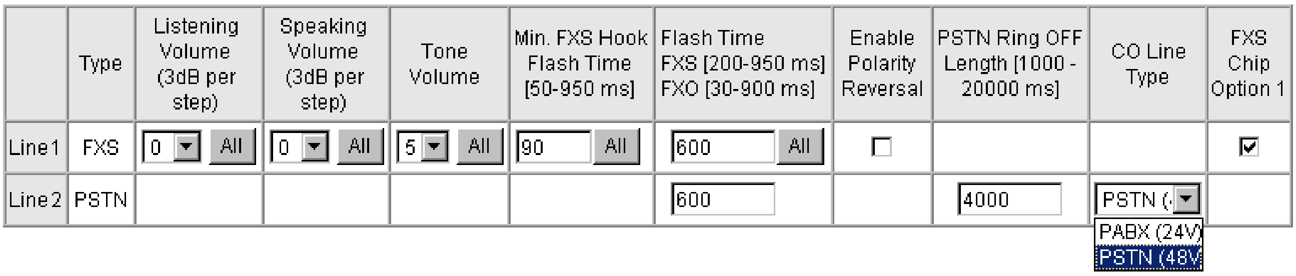
OR SERIES provides function for advanced voice settings, such as FAX, Codec, Speaking and Listening volume, etc.

Line Settings

You can adjust listening volume, speaking volume and tone volume here.



	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	Min. FXS Hook Flash Time [50-950 ms]	Flash Time FXS [200-950 ms] FXO [30-900 ms]	Enable Polarity Reversal	PSTN Ring OFF Length [1000 - 20000 ms]	CO Line Type	FXS Chip Option 1
Line1	FXS	0 All	0 All	5 All	90 All	600 All	☐			☑
Line2	FXS	0	0	5	90	600	☐			☑
Line3	PSTN					600		4000	PSTN (PABX (24V) PSTN (48V	



	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	Min. FXS Hook Flash Time [50-950 ms]	Flash Time FXS [200-950 ms] FXO [30-900 ms]	Enable Polarity Reversal	PSTN Ring OFF Length [1000 - 20000 ms]	CO Line Type	FXS Chip Option 1
Line1	FXS	0 All	0 All	5 All	90 All	600 All	☐			☑
Line2	PSTN					600		4000	PSTN (PABX (24V) PSTN (48V	

Item	Description
Listening Volume	It is to adjust the hearing volume.
Speaking Volume	It is to adjust the speaking volume.
Tone Volume	It is to adjust the tone volume. It will be applied to all tones volume generated by OR SERIES including Dial Tone, Busy Tone, and so on.
Min. FXS Hook Flash Time	It is to set the minimum flash time for FXS detecting.
Flash Time	FXS: Enter the maximum 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. PSTN: It is the time of PSTN port going on-hook. If on-hook time of PSTN is longer than the flash time of PSTN.
Enable Polarity Reversal	As the remote party answer this call, the polarity will be reversed.

Item	Description
PSTN Ring OFF Length	It is used to detect if the PSTN remoter party is on-hook through the ring length from PSTN by PSTN port. If the ring length form PSTN is larger than this setting, it is going on-hook by PSTN port, and it makes FXS stop ringing.
CO Line Type	Choose PSTN if the PSTN port is connected to PSTN line. Choose PABX if the PSTN port is connected to PABX line.
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.

Ring (Early Media) Time Limit [10 - 600 s]	90
Enable End of Digit Tone	☐
Force Calling Thru PSTN Code	
Early Media Treatment	☑
Loop Current Drop Trigger Time [0=disable, 3 - 30 s]	0
Loop Current Drop Duration [1 - 5 s]	2
Enable ROH	☐

Item	Description
Ring (Early Media) Time Limit	Specify the interval of ring time to cancel a call when no one answers a call.
Enable End of Digit Tone	OR SERIES will play a “Beep-Beep” tone to notify the call is in progress. It will play when invite packet is sent.
Force Calling Thru PSTN Code	Set the preferred code you set to force calling through PSTN. For example: If the code is set to *33 and you would like to dial “23456789” through PSTN, just dial “*33 23456789”.
Early Media Treatment	It refers to media that is delivered before call answer to inform the remote user about the session establishment. If you fail to make a call, please disable it.

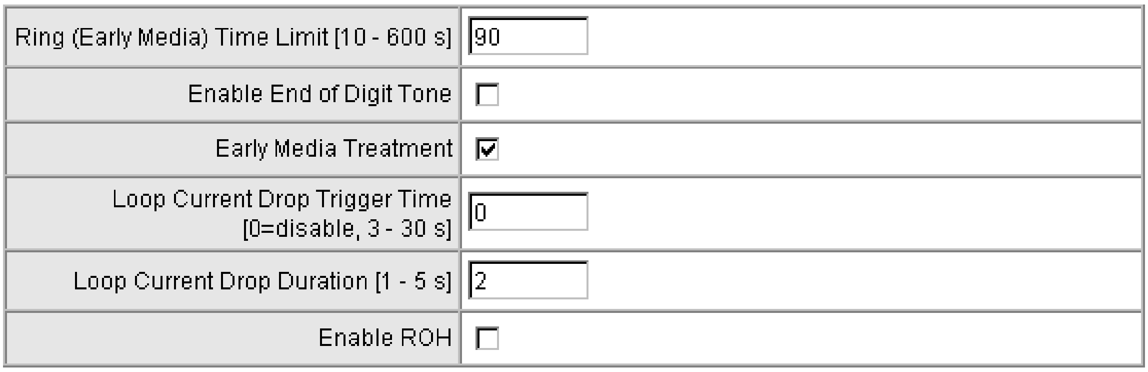
Item	Description
Loop Current Drop Trigger Time	It is to set the trigger time for dropping loop current by FXS port. A setting of zero is to disable this function. It is used to avoid the line engaged if FXS port is connected to PBX.
Loop Current Drop Duration	It is to set the drop duration.
Enable ROH	OR SERIES will play Receiver Off-Hook tone to notify user of hanging up the phone set.

	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	Min. FXS Hook Flash Time [50-950 ms]	Flash Time FXS [200-950 ms]	Enable Polarity Reversal	FXS Chip Option 1
Line 1	FXS	0 All	0 All	5 All	90 All	600 All	☐	☑
Line 2	FXS	0	0	5	90	600	☐	☑

	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	Min. FXS Hook Flash Time [50-950 ms]	Flash Time FXS [200-950 ms]	Enable Polarity Reversal	FXS Chip Option 1
Line 1	FXS	0 All	0 All	5 All	90 All	600 All	☐	☑

Item	Description
Listening Volume	It is to adjust the hearing volume.
Speaking Volume	It is to adjust the speaking volume.
Tone Volume	It is to adjust the tone volume. It will be applied to all tones volume generated by OR SERIES including Dial Tone, Busy Tone, and so on.
Min. FXS Hook Flash Time	It is to set the minimum flash time for FXS detecting.
Flash Time	It is to adjust the maximum 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 Detect Time" should fix this issue.
Enable Polarity Reversal	As the remote party answer this call, the polarity will be reversed.

Item	Description
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.



Item	Description
Ring (Early Media) Time Limit	Specify the interval of ring time to cancel a call when no one answers a call.
Enable End of Digit Tone	OR SERIES will play a “Beep-Beep” tone to notify the call is in progress. It will play when invite packet is sent.
Early Media Treatment	It refers to media that is delivered before call answer to inform the remote user about the session establishment. If you fail to make a call, please disable it.
Loop Current Drop Trigger Time	It is to set the trigger time for dropping loop current by FXS port. A setting of zero is to disable this function. It is used to avoid the line engaged if FXS port is connected to PBX.
Loop Current Drop Duration	It is to set the drop duration.
Enable ROH	OR SERIES will play Receiver Off-Hook tone to notify user of hanging up the phone set.

	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	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
Line 1	FXS	0 All	0 All	5 All	90 All	600 All	☐			☑
Line 2	FXO	0	0	5		600	☐	Disable	4000	

Item	Description
Listening Volume	It is to adjust the hearing volume.
Speaking Volume	It is to adjust the speaking volume.
Tone Volume	It is to adjust the tone volume. It will be applied to all tones volume generated by OR SERIES including Dial Tone, Busy Tone, and so on.
Flash Time	It is to adjust the maximum 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 Detect Time” should fix this issue.
Enable Polarity Reversal	As the remote party answer this call or FXS picks up, the polarity will be reversed.
PSTN Answer Detection	This is used for VSP only. When there is call from WAN interface to FXO port, it could identify if the called party of PSTN answers this call. After it dials to PSTN, it will send “183” to the calling party. After the called party of PSTN answers this call, it will send “200 ok” to another the calling party and the VSP starts to charge.
PSTN Ring OFF Length	It is used to detect if the PSTN remoter party is on-hook through the ring length from PSTN by PSTN port. If the ring length form PSTN is larger than this setting, it is going on-hook by PSTN port, and it makes FXS stop ringing.
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.

Ring (Early Media) Time Limit [10 - 600 s]	90
Enable End of Digit Tone	☐
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	☐

Item	Description
Ring (Early Media) Time Limit	Specify the interval of ring time to cancel a call when no one answers a call.
Enable End of Digit Tone	OR SERIES will play a “Beep-Beep” tone to notify the call is in progress.
Force Calling Thru PSTN Code	Set the preferred code you set to force calling through PSTN. For example: If the code is set to *33 and you would like to dial “23456789” through PSTN, just dial “*33 23456789”.
Trunk Early Media Options	Early Media refers to media that is generated prior to connection or answer of a call is established by the called party. 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. **Both Way Voice:** Use bidirectional early media to obtain information between caller and callee prior to the connection of a call. **One Way Voice:** Only the caller can hear early media from the callee prior to the connection of a call. **Ring Back:** Playing ring back tone for the caller, indicating that the callee is being alerted prior to the connection of a call.
Early Media Treatment	It refers to media that is delivered before the call is answered to inform the called party about the session establishment. If you fail to make a call, please disable it.
Loop Current Drop Trigger Time	It is to set the trigger time for dropping loop current by FXS port. A setting of zero is to disable this function. It is used to avoid the line engaged if FXS port is connected to PBX.

Item	Description
Loop Current Drop Duration	It is to set the drop duration.
Enable ROH	OR SERIES will play Receiver Off-Hook tone to notify user of hanging up the phone set.

Termination Impedance

Choose correct impedance in your country/area. The wrong impedance will cause voice failure.

FXO Impedance	Taiwan 600Ω
FXS Impedance	Taiwan 600Ω

Drop Inactive Call

This is used as a standard for FXS and FXO interface to determine whether or not to back to the idle state. OR SERIES will back to the idle state automatically to avoid keeping the line engaged while the time expires and the detected volume is lower than Silence Detection Threshold.

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

Item	Description
Silence Detection Threshold	Set the ceiling threshold of voice energy to be identified as silence.
Drop Silent Call Timeout	Set the silence period to wait for before dropping a call.

Voice Menu Options

This is used to enable or disable IVR function or Call Feature Code. When disabled, call pickup/repeat, dialing/unattend transfer will be disabled.

Enable	☑
Enable Call Feature Code	☑

Codec Settings

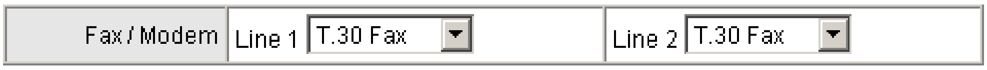
You can set the preferred codec, Jitter Buffer, Silence Detection/Suppression and Echo Cancellation in this section.

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

Item	Description
Preferred Codec Type	Since different voice codec have different compression ratios, so 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	It is to adjust the jitter to receive a packet. If the jitter range is too large, it will delay voice transmission.
Silence Detection/ Suppression	If one side of a connection is not speaking, OR SERIES will stop sending voice data (package) to decrease bandwidth usage.
Echo Cancellation	It is to prevent poor telecommunication quality caused by echo interference.
Packet Time	Defines how long OR SERIES sends a RTP packet (voice packet) to the remote party. The smaller the value, the more bandwidth usage. The larger the value, the more voice delay.
Approximate Bandwidth Require	The bandwidth required varies with codec format and packet time.

FAX Settings

The line will detect FAX automatically if you choose T.30 Fax, T.38 Fax, T.30/Modem or T.30 Only. Choose the type of FAX protocol and set the related settings.



Item	Description
Disable	The line do not detect FAX automatically.
T.30 Fax	OR SERIES uses T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the fax function will consume more network resources and will affect transmission quality.
T.38 Fax	OR SERIES uses 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/Modem	Choose T.30 Fax/Modem as the protocol for transmission if OR SERIES is connected to Modem.
T.30 Only	Choose T.30 as the protocol for transmission. OR SERIES only accept the fax protocol of T.30.

Item	Description
Enable High Quality	OR SERIES sends the same FAX frame twice to get a high quality of the FAX when the line is using T.38 Fax. It requires more bandwidth.
FAX Codec	OR SERIES provides G.711 and G.726 for T.30 fax transmission. It is recommended to use G.711 for T.30.
FAX Jitter Buffer	It is to adjusts the jitter to receive fax packets. If the jitter range is too large, it will delay fax transmission.

Other Settings

OR SERIES provides advanced settings to apply to various situations. Here are **Digit Map**, **DTMF & Pulse**, **CPT/Cadence Settings** and **Provision Settings**.

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)

Digit Map Table

#	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)

Item	Description
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	Define the default call route of OR SERIES. If Default Call Route is Deny, all numbers that are not match the Digit Map Table will be denied.
Enable	Tick the check box to make this entry effective.
Scan Code	Define the leading digits for OR SERIES to scan while the user is dialing.
VoIP Dial-out	Define the dialed number rule for OR SERIES calling through Internet.

Item	Description
PSTN Dial-out	Define the dialed number rule for the gateway to call through PSTN/FXO port.
User Dial Length	Define total number of digits that user dialed. A setting of zero tells the gateway scans digits only and disregards the total digit count.
Route	It is to determine the interface calls should go through if above conditions satisfied.

Enable Pound Key '#' Function	☑
Default Call Route	Auto (VoIP first)

Digit Map Table

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

Item	Description
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	Define the default call route of OR SERIES. If Default Call Route is Deny, all numbers that are not match the Digit Map Table will be denied.
Enable	Tick the check box to make this entry effective.
Scan Code	Define the leading digits for OR SERIES to scan while the user is dialing.
VoIP Dial-out	Define the dialed number rule for OR SERIES calling through Internet.
User Dial Length	Define total number of digits that user dialed. A setting of zero tells the gateway scans digits only and disregards the total digit count.

Item	Description
Route	Determine the interface calls should go through if above conditions satisfied.

Digit Map Testing

Digit Map Testing		
Test Dial No.		Run
Result		

Item	Description
Test Dial No.	You have to set some rules in Digit Map Setting first and enter the number for test.
Result	OR SERIES will show the number for VoIP Dial-out and PSTN Dial-out according to the Digit Map Table.

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 OR SERIES will dial 07021234567. You also can use Digit Map Testing to know that OR SERIES 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. OR SERIES will dial 351006301 and go through PSTN/FXO. You also can use Digit Map Testing to know that OR SERIES will dial 07021234567 and go through PSTN/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. OR SERIES will dial 0423456789 and go through Internet first. If the call is fail to Internet, OR SERIES will dial 1805423456789 and go through PSTN/FXO. You also can use Digit Map Testing to know that OR SERIES will dial 0423456789 to Internet and 1805423456789 to PSTN/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. OR SERIES will dial “test” and go through Internet. You also can use Digit Map Testing to know the dialing result.

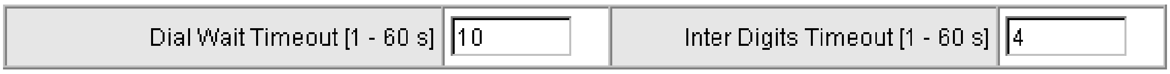
Digit Map Testing	
Test Dial No.	11 Run
Result	#1: VoIP=test PSTN=11

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

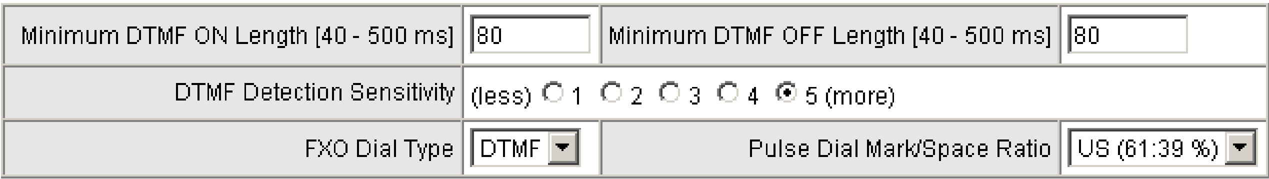
DTMF & Pulse

You can change these parameters if you have problems in dialing number.

DTMF Settings



Item	Description
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 Time Out	It is to set the waiting time between each key pressing. If the caller does not press the next number before the time expires, OR SERIES will play busy tone.



Item	Description
Minimum DTMF ON Length Minimum DTMF OFF Length	Define the length of diverting a call to another extension line. (Adjust length between Dail_on and Dail_off).
DTMF Detection Sensitivity	It is to adjust the sensitivity of detecting numbers for OR SERIES.
FXO Dial Type	Select dial type for FXO. There are DTMF and Pulse.
Pulse Dial Mark/Space Ratio	Duration and break of pulse dial ration.

Out-of-Band DTMF

Item	Description
Enable Out-of-Band DTMF	Tick the check box to send DTMF keys (0~9, *, #,) follow the RFC2833 rules or via SIP Info.
Enable Hook Flash Event	According to RFC2833 or SIP info, OR SERIES will deliver Hook Flash signal to the remote party.
Volume	Defines the DTMF volume of RFC 2833.
Payload Type	Payload type of RFC2833.

CPT/Cadence Settings

OR SERIES will generate the tones by the call process tone parameters table.

CPT parameters Table

The CPT has 2 sets of parameter tables. Please adjust the parameters based on local PSTN.

# 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

Busy Tone Cadence Measurement

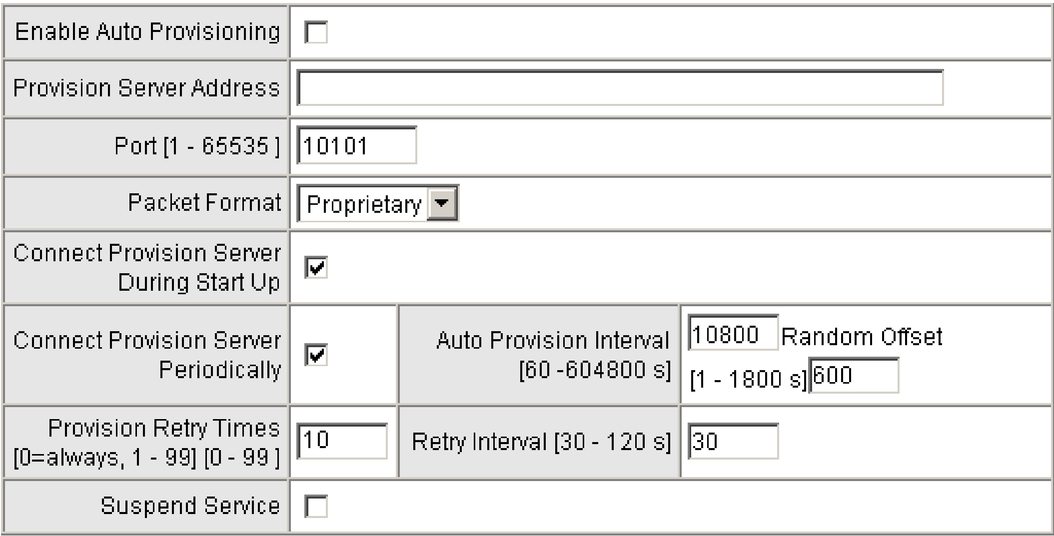
CPT/Cadence setting parameters serve as the basis of an FXO interface to determine whether or not a PSTN-call receiving party has hung up the phone. If the following parameters differ from the parameters of the actual assigned lines, it could cause the FXO to continue to engage a line.

BTC Enable	Busy Tone Cadence Measurement				
	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning
BTC # 1	250	250	0	0	Yes
BTC # 2	500	500	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)				
BTC Volume Threshold [20 - 70 dB]	25				

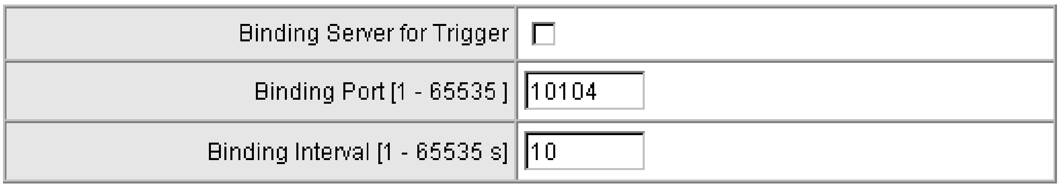
Item	Description
Auto Learning	It is to learn the busy tone automatically by FXO port.
BTC Detection Sensitivity	The more sensitivity, the more quickly it will cut off the call by FXO port. If it often cut off an un-finished call by FXO port, select less sensitivity.
BTC Volume Threshold	The detection level for BTC.

Provision Settings

Provision Server is used to provision, configure, manage and maintain subscribers and network users. OR SERIES, acts as a part of subscribers, can be controlled by Provision Server. OR SERIES 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 OR SERIES and all actions will be recorded in log file that offers users to trouble shouting effectively.



Item	Description
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.
Port	The port of Provision Server.
Packet Format	Select the packet transmitting format required by provision server.
Connect Provision Server During Start Up	OR SERIES will connect to Provision Server when it power on or reboot.
Connect Provision Server Periodically	It is to adjust the parameters for OR SERIES to connect to provision server periodically.
Suspend Service	It is to adjust the parameters for OR SERIES to do auto provision task.



Item	Description
Binding Server for Trigger	Tick the check box to trigger of a connection between server and OR SERIES. Server will bind a port for the gateway to send provision request.
Binding Port	The binding port number of the server is used to tell OR SERIES the path of binding server.
Binding Interval	It to set the desired Interval at which OR SERIES will keep the binding.

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**. Transit Call Control is active in one-stage dialing.

Inbound Call Control	☐	Outbound Call Control	☐

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

Item	Description
Inbound Call Control	Tick the inbound PIN code when users make phone calls from a PSTN to FXO and then using a VoIP — only effective for incoming calls calling from a PSTN trunk.
Outbound Call Control	Tick the outbound PIN code when users utilize FXO interface to divert to a PSTN — only effective for outgoing calls being diverted to a PSTN Trunk.
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, as below.

This table is used to prohibit dialing any numbers started with specified prefixes. Digit strings in this table are prefixes that the gateway will check on dialed numbers in transit out calls. It is 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. For example, 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.

#	0	1	2	3	4	5
1	0204					
2						
3						

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. It is 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.

#	0	1	2	3	4	5
1	020488988					
2						
3						

Status and Tools

This section shows the status of OR SERIES. There are **Current Status**, **RTP Packet Summary**, **System Information**, **Ping Test** and **STUN Inquiry**.

Current Status

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

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

Current Status

Port Status							
No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	701		0		Disabled	
2	FXS	702		0		Disabled	

Server Registration Status	
DDNS Registration	Disabled (00:01:53)
Phone Book Manager Registration	Disabled (00:01:53)
STUN Registration	Disabled (00:01:53)
UPnP Negotiation	Disabled (00:01:53)

RTP Packet Summary

Display the information of the final call. 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

System Information

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. If IP address, subnet mask, default gateway is blank, it means that OR SERIES does not obtain IP.

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

Hardware: It shows the hardware platform.

System Information	
WAN Port Information	
Factory Default MAC Address	88 69 36 89 70 89
IP Address	
Subnet Mask	
Default Gateway	
DNS	
LAN Port Information	
MAC Address	88 69 36 89 70 8A
IP Address	192.168.8.254
Subnet Mask	255.255.255.0
DHCP Server	
DHCP Server	Enabled
IP Pool Range	192.168.8.1 - 192.168.8.250
Lease Time	1 hour(s)
DNS	
Hardware	

Ping Test

Use **Ping** to identify if the remote peer is reachable. Fill in remote IP address and click **Test** will start the test.

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

STUN Inquiry

It is to know what NAT type of the router when OR SERIES is behind NAT.

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

System Settings

This section provides system settings such as **NTP**, **Login Account**, **Backup/Restore**, **System Operation**, **Software Upgrade** and **Logout**.

NTP

It is to set the **Time Zone** where OR SERIES resides. You can set the **Time Server** where OR SERIES should sync up during start up.

	Year	Month	Day	Hour	Minute	Second
Gateway Time	2000	1	1	8	0	0
Time Zone	+ 8 : 00					
#	Time Server					
1	ntp.ucsd.edu					
2	ntp.univ-lyon1.fr					
3	time.nuri.net					

Login Account

There are two sections in this page: Login Settings and Accessing Services.

Login Setting: There are two levels to enter Web. Administrator is able to change all settings. Web UI only changes some settings.

Access Services: It is to allow users to access OR SERIES not only from Web but also from Telnet.

Login Settings

Note: Enter new Login ID and password for two levels.

Administrator's Name			
Administrator's Password	***********	Confirm Password	***********
Web UI Login ID			
Web UI / IVR Password	***********	Confirm Password	***********

Access Services

Note: When "Enable Web UI" is unticked, you cannot access from Web.

Port of Web Access from WAN [0=disable, 1 - 65535]	80
Web UI auto logout [30 - 300 s]	60
Enable Web UI	☑
Enable Telnet Service	☑

Item	Description
Port of Web Access from WAN	Http port for WAN. To make this setting, the LAN Port must be used. It cannot be made using the WAN Port. Always use port 80 when connecting to LAN port. A setting of zero is to disable http port for WAN.
Web UI auto logout	If OR SERIES is inactive for the period defined in this filed, Web UI will auto logout to keep OR SERIES secure.
Enable Web UI	Untick the check box to disable WEB access from WAN or LAN while necessary.
Enable Telnet Service	Untick the check box to disable Telnet access from WAN or LAN while necessary.

Backup/Restore

You can backup settings to a file and restore settings from that file.

Backup Configurations

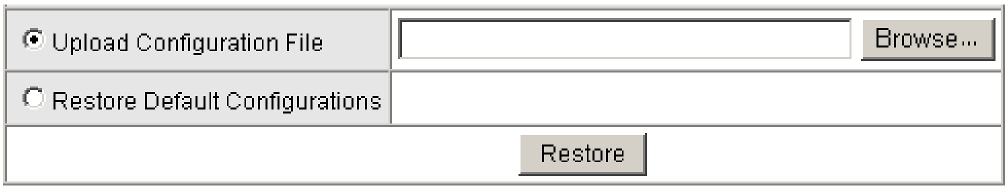
Configuration File	Backup
Configuration Template File	Backup

Item	Description
Configuration File	It is to backup the all settings.
Configuration Template File	It is to backup the settings as template file for editing.

Restore Configurations

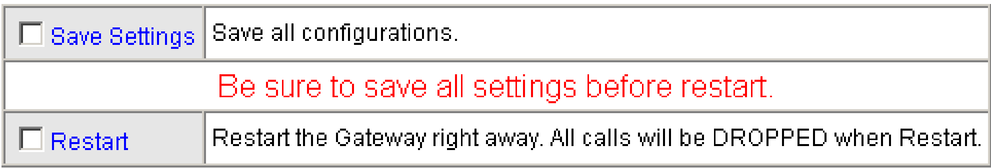
You can backup settings to a file and restore settings from that file. You also can restore all settings back to default by selecting **Restore Default Configurations** and click **Restore**.

Note: You have to save settings and restart, and all settings will take effect.



System Operations

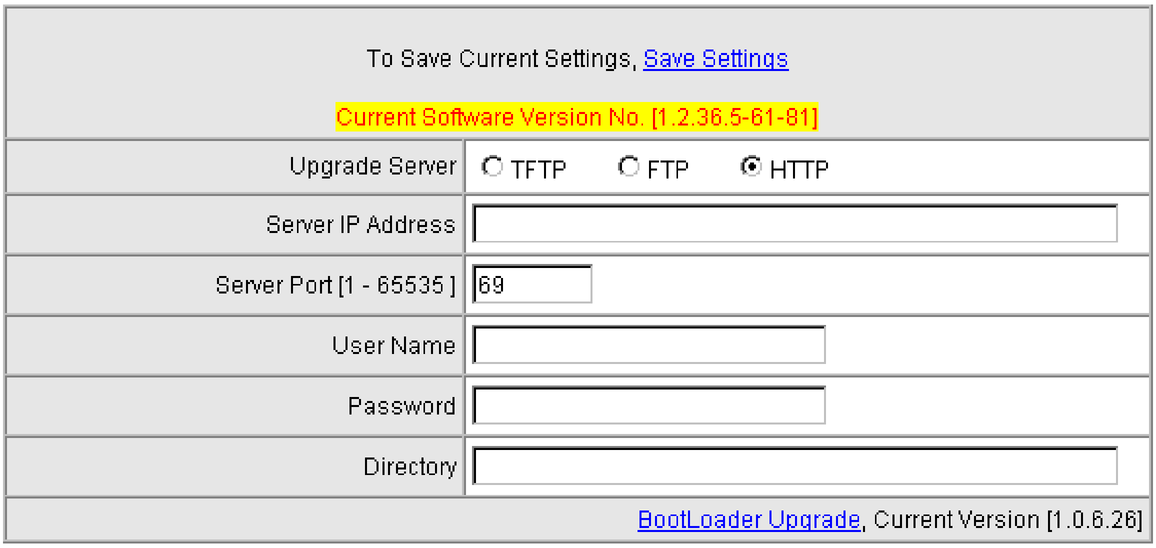
Some settings are effective by **Restart**. Remember to save all settings by **Save Settings** before to restart.



Item	Description
Save Settings	Save settings after completing configuration.
Restart	The new settings will take effect after OR SERIES is restarted. Please select it and click the **Accept** button.

Software Upgrade

OR SERIES provides software upgrade function for a remote end.
Your provider gives all parameters.



Item	Description
Upgrade Server	Choose the server type of your provider.
Software Upgrade Server IP	Enter the software upgrade server IP address.
Software Upgrade Server Port	Enter the port that server uses. TFTP is 69, and FTP is 21.
User Name/ Password	The account/password is to login the upgrade server.
Directory	The location of Directory for Upgrade Server.

Logout

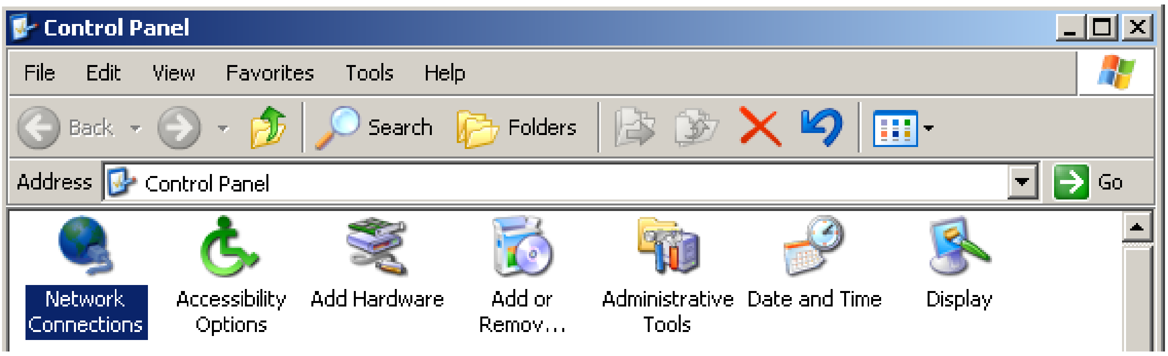
OR SERIES only allows one user to login at a time, so whenever a change is made, please save the settings, restart OR SERIES, or logout to avoid the situation where other users cannot login to change settings.

5. TCP/IP Setting

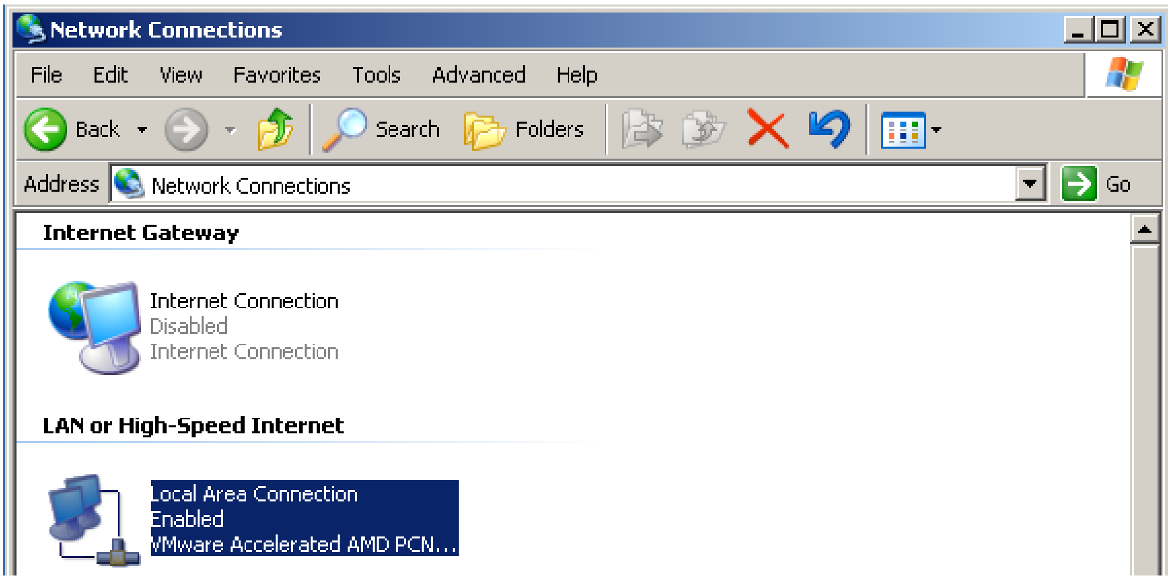
Follow the description if you have problems in how to assign a static IP Address in your PC.

Using Windows XP for example

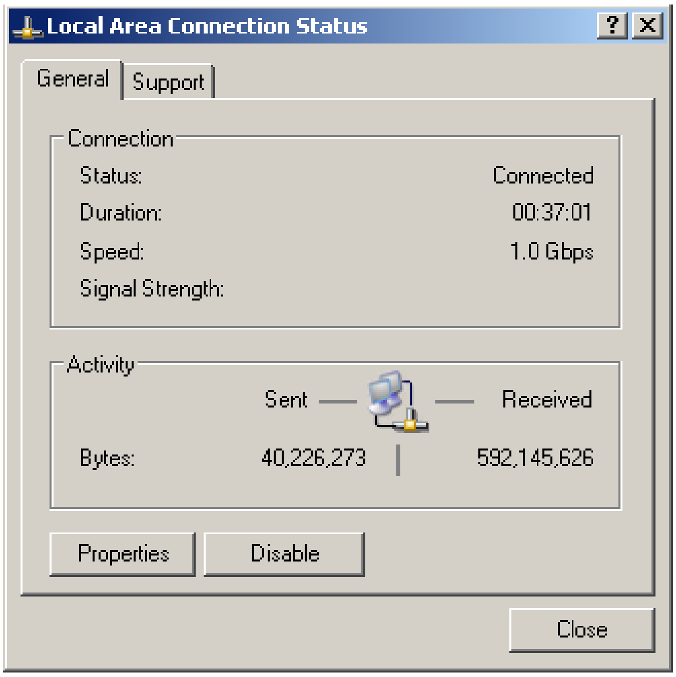
Go to **Start** -> Click on **Control Panel** -> **Double-click on Network and Dial-up Connection** ->



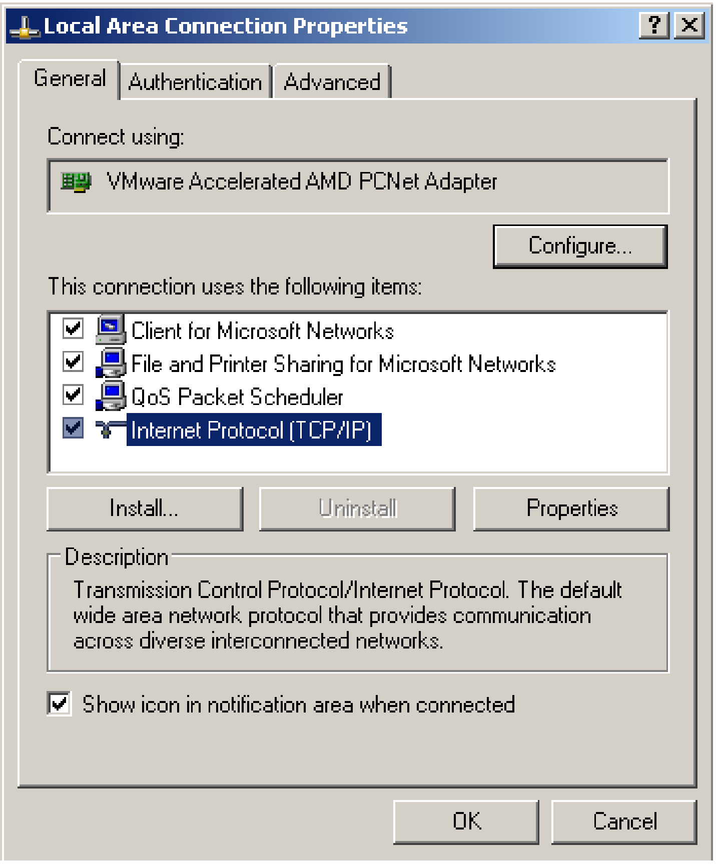
Click on **Open Local Area Connection** ->



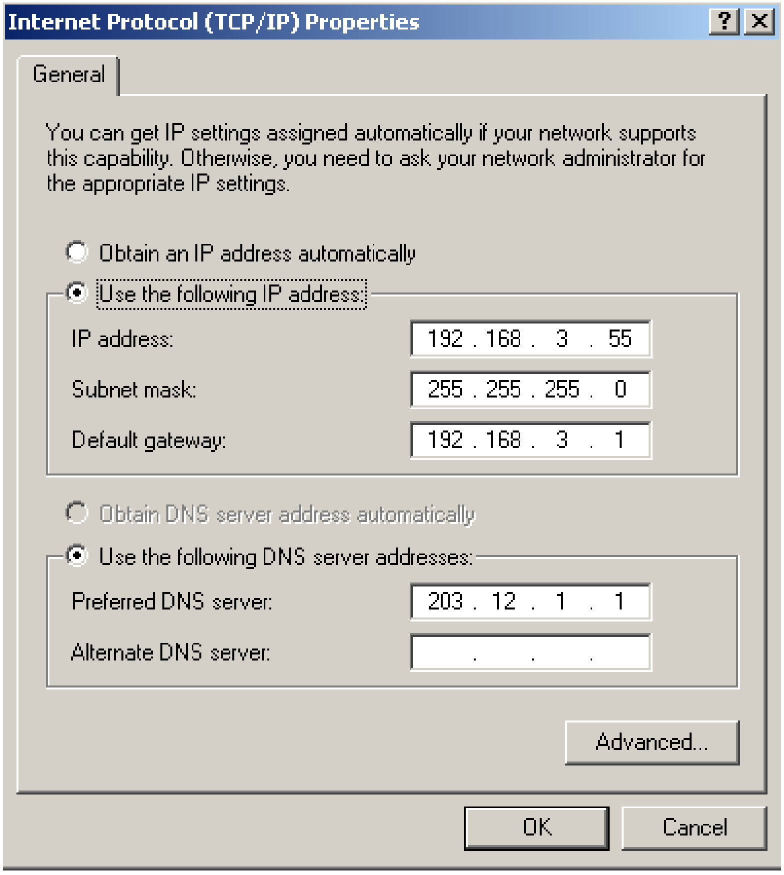
Click **Properties**.



Highlight **Internet Protocol (TCP/IP)** and then click **Properties**.



Select **Use the following IP Address**. Set **IP address**, **Subnet mask** and **Default gateway**. The IP Address must be within the same range as OR SERIES (If the IP Address of OR SERIES is 192.168.8.254. You can assign 192.168.8.100 for your PC). Then, enter the DNS server IP address (varies in different networks. consult your ISP's service for information). Click on the **OK** button to make settings take effect.



FEDERAL COMMUNICATIONS COMMISSION INTERFERENCE STATEMENT

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

– Reorient or relocate the receiving antenna.
– Increase the separation between the equipment and receiver.
– Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
– Consult the dealer or an experienced radio/TV technician for help.

CAUTION:

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference and
(2) This device must accept any interference received, including interference that may cause undesired operation.

RF exposure warning ·

This equipment must be installed and operated in accordance with provided instructions and the antenna(s) used for this transmitter must be installed to provide a separation distance of at least 20 cm from all persons and must not be co-located or operating in conjunction with any other antenna or transmitter. End-users and installers must be provide with antenna installation instructions and transmitter operating conditions for satisfying RF exposure compliance.