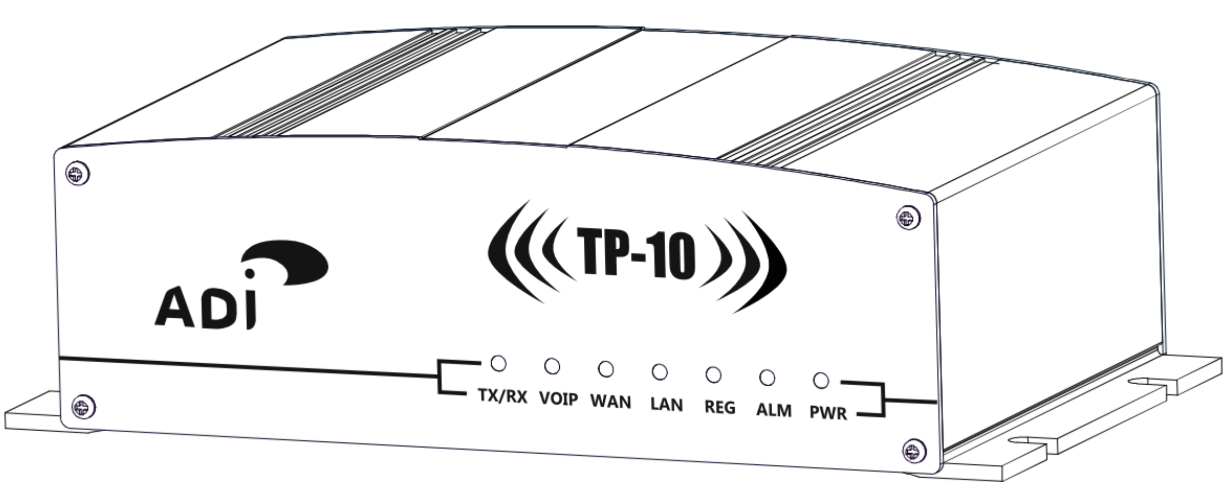
Version 1.0 (2016)

TP-10 RoIP Gateway

User Manual



Contents

Terms and Conditions

Notice:
Please follow the following instructions to avoid unnecessary damage to the product.

1. Please turn off the power before plugging in/out, and also during the setup of accessories.

2. When the Internet is congested, the quality of the calls will be influenced.

3. Please keep the product away from other electronic devices to avoid radio interference, such as wireless base station (AP), AC Power Converter, and etc.

4. “Push-to-Talk” pattern is also applied on Smart Phones calls.

5. Please use only the AC Power Converter included in the package.

6. Please do not open/dissect/fix the product by yourself. Please contact the Service Provider or the Original Dealer for more service.

7. Please keep the product away from hot, humid, or raining environment.

8. If you find any question about this product, please contact the Service Provider or the Original Dealer.

Item List
Please check if the package is damaged. Open the package carefully and check if the content is as the list below. Please contact the local dealer immediately if you find any accessory is missing or damaged.

Item	Quantity
Radio Gateway	1
Antenna	1
Power Converter	1
Ethernet Cable	1
User Manual	1
Guarantee Card	1

1. Introduction

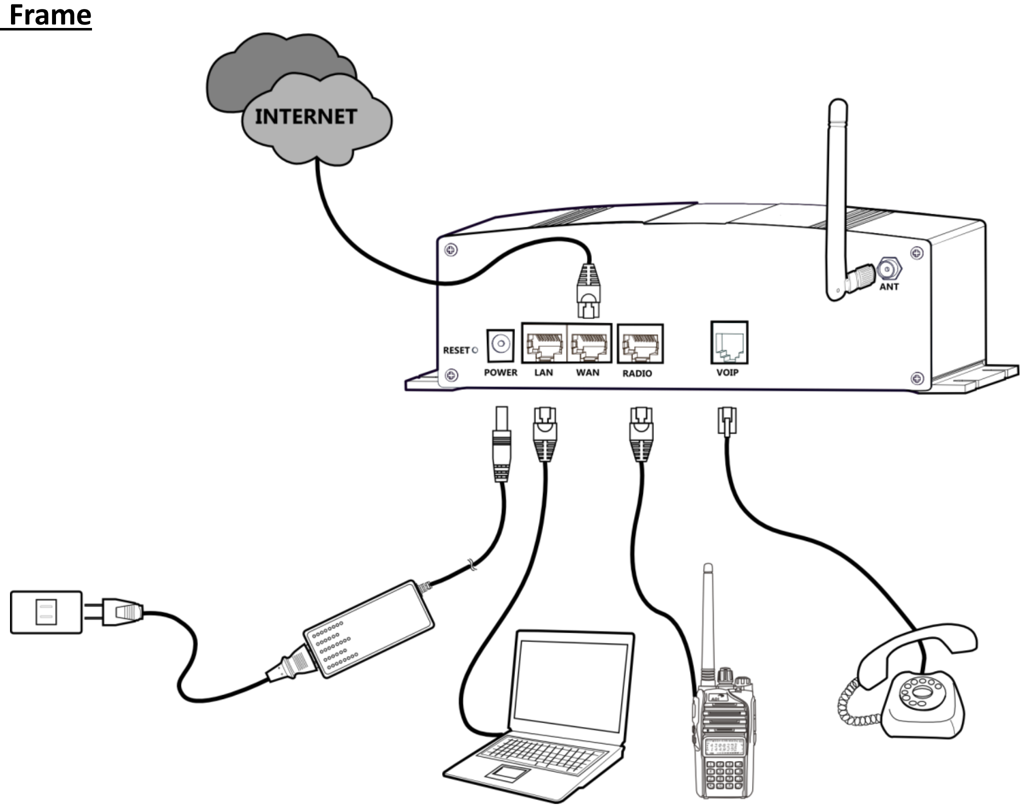
1-1 Product Overview

With Internet service becoming universal, the application of Internet of Things (IoT) steps into our daily lives as well, and VoIP (Voice over Internet Protocol) is one of them. Through Radio Gateway to make contacts directly with telephones and smart phones has become an innovative choice for transportation, restaurants, warehousing, industries and government departments..

ADI Communications combines the benefits of radio and Internet without changing the user pattern. Through DSP and Communication Protocol technology, the radio audio can be Encoded, Compressed and sent through Internet as Information Packet. Of course, the Information Packet on the Internet can be Decompressed, Decoded, and sent through Public Telecom Network as audio. This system is ADI Communications' Radio Gateway.

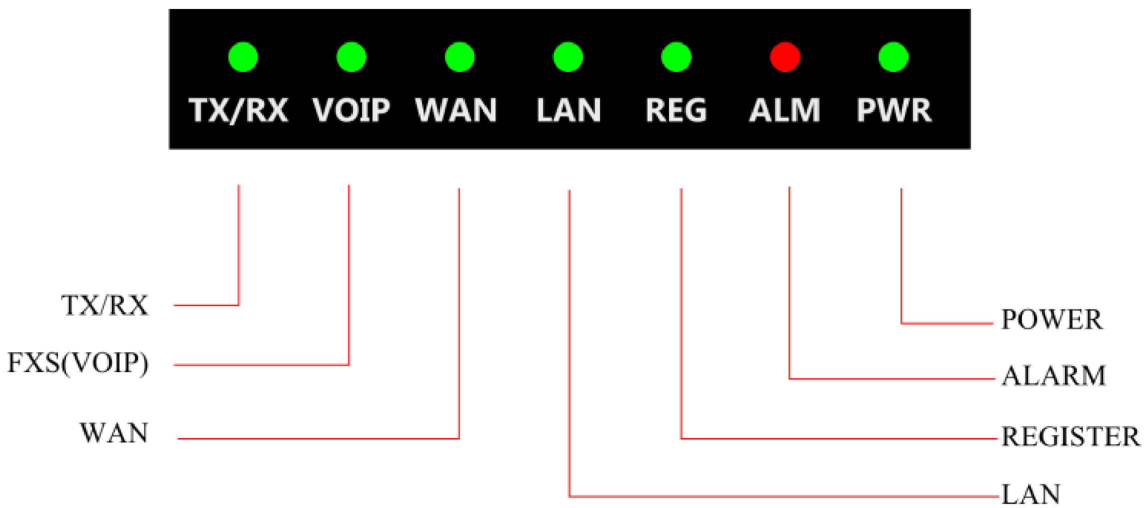
Interface RJ45 is applied to the Radio Gateway to connect to traditional radio equipments, and provides varied function according to purchased model. FXS Port connects to telephones, or connects to PBX for internal extensions to make VoIP calls. SIP is applied to transceive the signal, and support fixed IP/DHCP configuration. It adopts G.711 voice compression format to save network bandwidth while providing real-time, toll quality voice transmission and reception.

Connection Frame



1-2 Hardware Description

Front Panel



Power : Green light indicates normal power function.

Alarm: Red light indicates Automated Start Testing or VoIP Router abnormal function.

Register: Green light is always on when SIP Server connection is normal.

LAN : Green light indicates the Internet is connected.

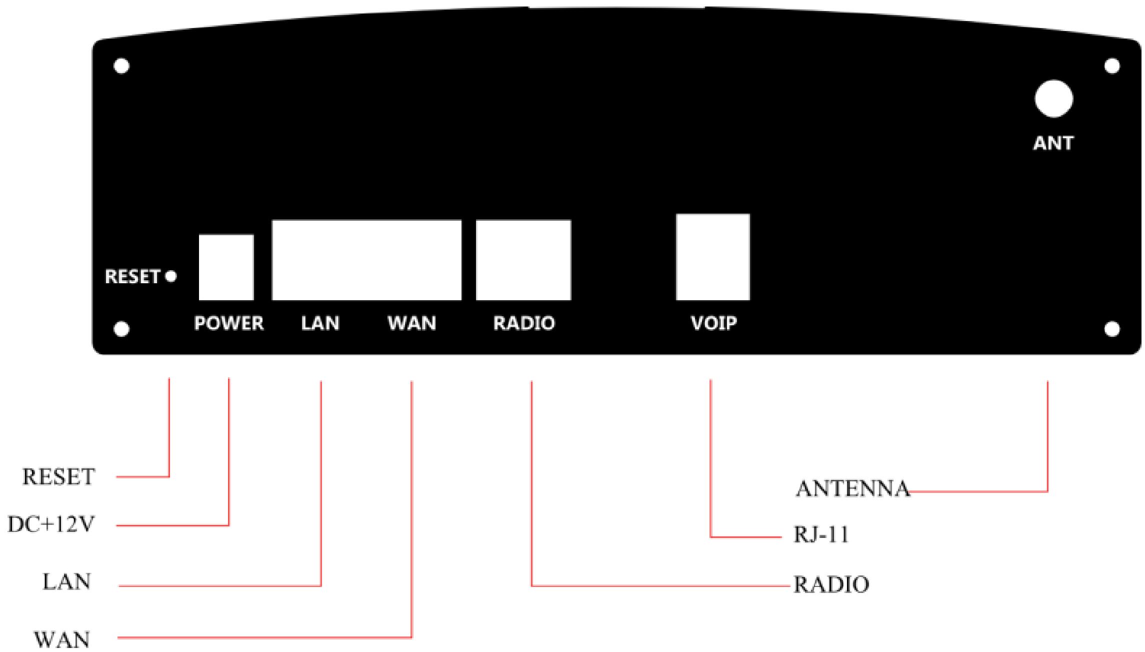
WAN: Green light indicates the Internet is connected.

VoIP: Green light is always on when FXS communication mode is being applied.

Radio: Green light indicates the radio is at receiving signal mode, and the red light is at transmitting signal mode.

The light of POWER, ALARM, REGISTER, RADIO will be lighted up when the Radio Gateway is turned on, the system will be functioning after around 30 seconds. When it’s functioning normal, the POWER light is always on, and the ALARM light is off; if the ALARM light keeps flickering, it indicates the Radio Gateway is connecting with ISP now, but hasn’t received the IP Address yet.

Back Panel



ANT : When the radio is built-in, VHF/UHF antenna can be setup.

VoIP: Through telephone the VoIP calls can be made.

RADIO : The custom-made external lead can be purchased additionally, and connect to your own radio system.

WAN : Use Ethernet cable to connect to Hub, ADSL, Cable Modem, or Router.

LAN : Use Ethernet cable to connect to PC. You can set up the web page through LAN Port.

DC+12V : Please use the Power Converter that comes along within the package.

Reset : When the power is on, press Reset button for around 6 seconds and stop when Alarm light is flickering, the factory defaults is now restored.

2. Radio Gateway Web Configuration

Steps :

1. Open a Web Browser (e.g., Explorer, Navigator, Opera, FireFox).

2. Enter the IP Adress of WAN Port, if you connect PC at LAN Port, then enter IP Address of LAN Port in the address filed and press Enter. Default LAN port IP address: **192.168.8.254.**

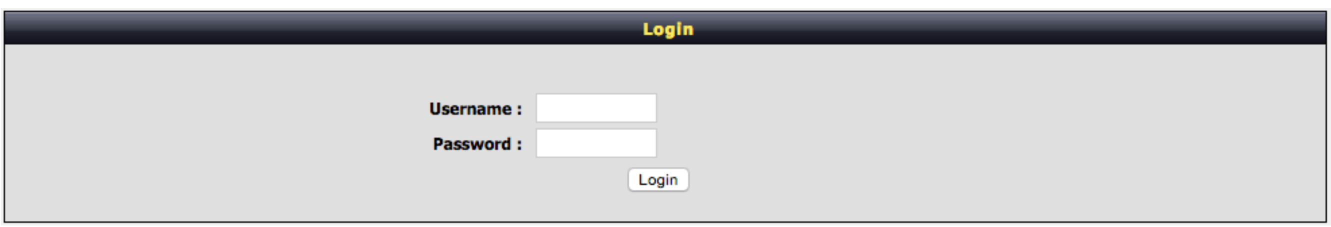
3. The log-in screen below will appear after the connection. The factory default settings

Username: admin/user

Password: blank(N/A)

4. After login, please go to System Settings → Login Account to reset Username and Password

5. All the settings will be effective immediately after clicking YES, but the Internet-related settings will be effective after Reboot. Save the settings: System Settings → Save/Reboot → Reboot then press YES



2-1 Status

2-1-1 Current Status

Status → Current Status

Current Status	
Refresh Time (2 - 30s) :	5

Port Status

Line	Type	Extension Number	Line Status	Calls	Number	Proxy Register
1	RoIP	701	Idle	0		Disabled (01:23:32)
2	FXS	702	Idle	0		Disabled (01:23:32)

SIP Proxy Hunting Number Registration :	FXS Disabled (01:23:32)

Server Registration Status

DDNS Registration :	Disabled (01:23:32)
Phone Book Manager Registration :	Disabled (01:23:32)
STUN Registration :	Disabled (01:23:32)

On this page it shows all the Type, Extension Number, Line Status, Calls, Proxy Registration Status, DDNS, and STUN status of each Line. You can see currently how the Radio Gateway is working.

You can also re-set Refresh Time on this page.

2-1-2 RTP Packet Summary

Status → RTP Packet Summary

RTP Packet Summary

Line

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

Display the audio packet transceive information of the last call of each Line, including IP Address, audio format, packet quantity and information. You can evaluate the Internet connection quality by this information: the portion between Packet Lost and Packet Sent.

Press Refresh you can see the latest RTP Packet summary.

2-1-3 System Information

Status → System Information

System Information

System Information

Time and Date :	2015/04/20 16:27:00
Firmware Version :	1.2.38.96-902

WAN Port Information

Factory Default MAC Address :	000C2A1004EA
Net Link :	Connected
IP address / Subnet mask :	192.168.2.104 / 255.255.0.0
Default Gateway :	192.168.2.1
Domain Name Server :	192.168.2.1

LAN Port Information

MAC Address :	000C2A1004EB
IP address :	192.168.8.254
Subnet mask :	255.255.255.0

Hardware

Hardware Platform :	RoIP
Driver :	1.4.2.230.501

IP address, subnet mask, default gateway and DNS server in column WAN Port Information. If you use PPPoE to obtain IP, you can know the obtained IP Address. If IP Address, subnet mask, default gateway is blank, it indicates the Radio Gateway does not obtain IP Address.

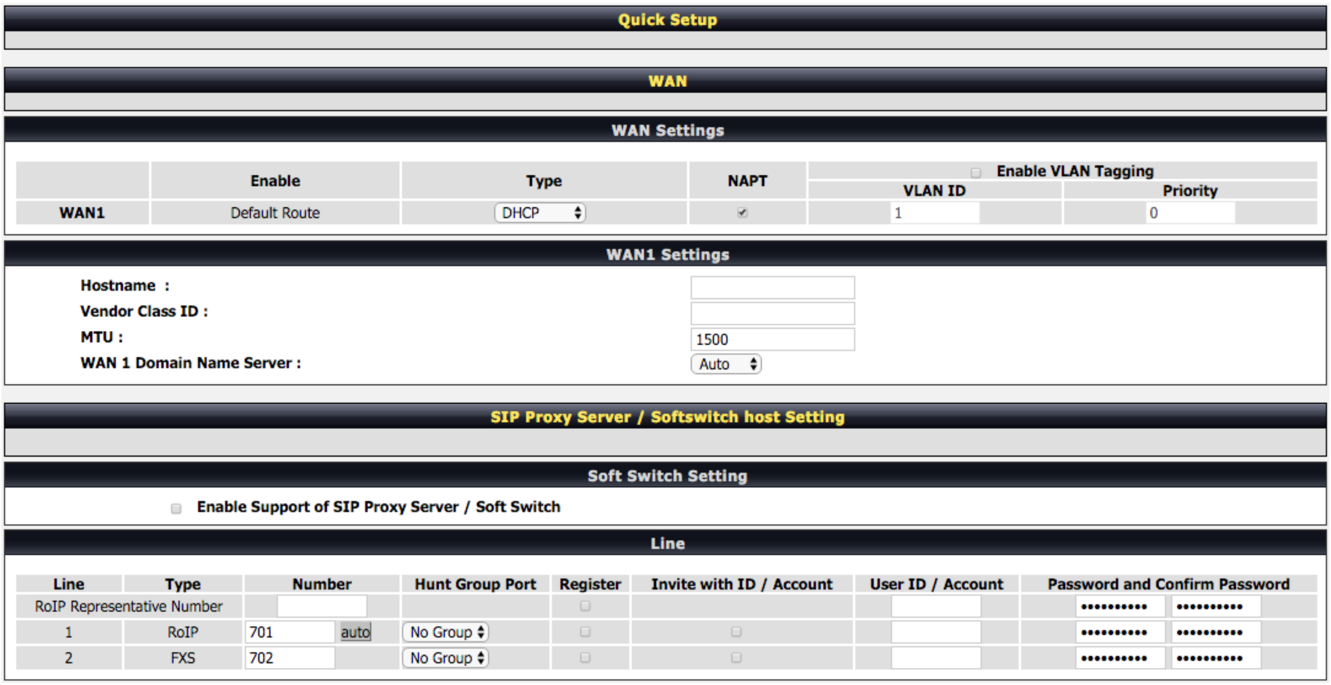
LAN port IP, subnet mask, and the status of DHCP server can be seen in column LAN Port Information

Hardware Platform and driver version are shown in column Hardware.

2-2 General Settings

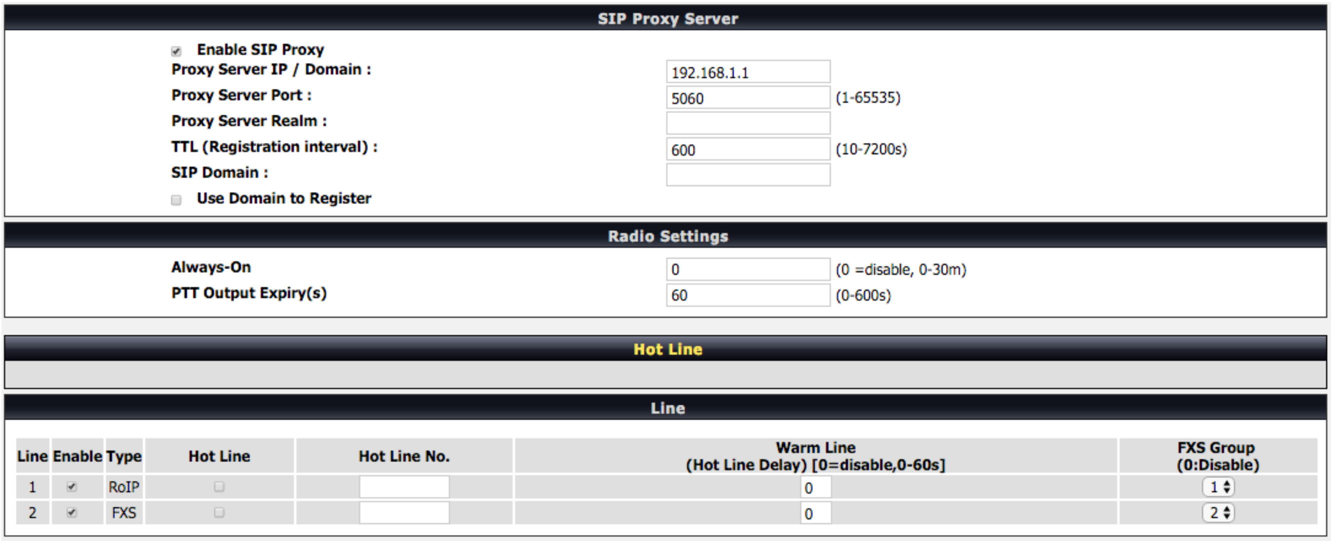
2-2-1 Quick Setup

This part includes WAN Settings, RoIP Accounts Settings, Hotlines, and Digit Map.



WAN Settings: Including DHCP, Fixed IP, PPPoE and etc.

RoIP Accounts Settings: Including Numbers, SIP Accounts, Password, and etc.



SIP Proxy Server: Including SIP Proxy and Port.

Radio Settings: Including Always-On Mode, PTT Output Expiry, and etc.

Hotlines: Including Hotline Numbers, Dial Waiting and etc.

Phone Book: 3 sets of frequently dialed number can be set and used by Calls Project.

Digit Map: 3 sets of frequently dialed number can be set and used by Phone Book.

2-2-2 WAN Settings

WAN (Wide Area Network) Settings is applied to connect to your ISP.

IP Configuration (Setting WAN Port)

There are four methods to obtain a WAN port IP address:

1. DHCP Dynamic IP
2. Static IP Fixed IP
3. PPPoE (dial-up ADSL)
4. PPTP

IP Address obtained by DHCP and PPPoE may change. If you are not familiar with creating a network connection, please contact your local ISP.

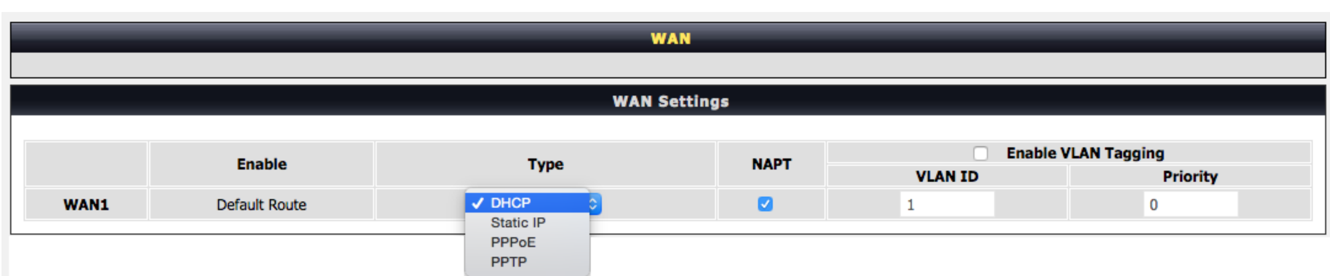
After selecting the suitable option, click **Accept** at the bottom of the screen to save the settings.

You need to save the changes and restart the Radio Gateway to activate the changes.

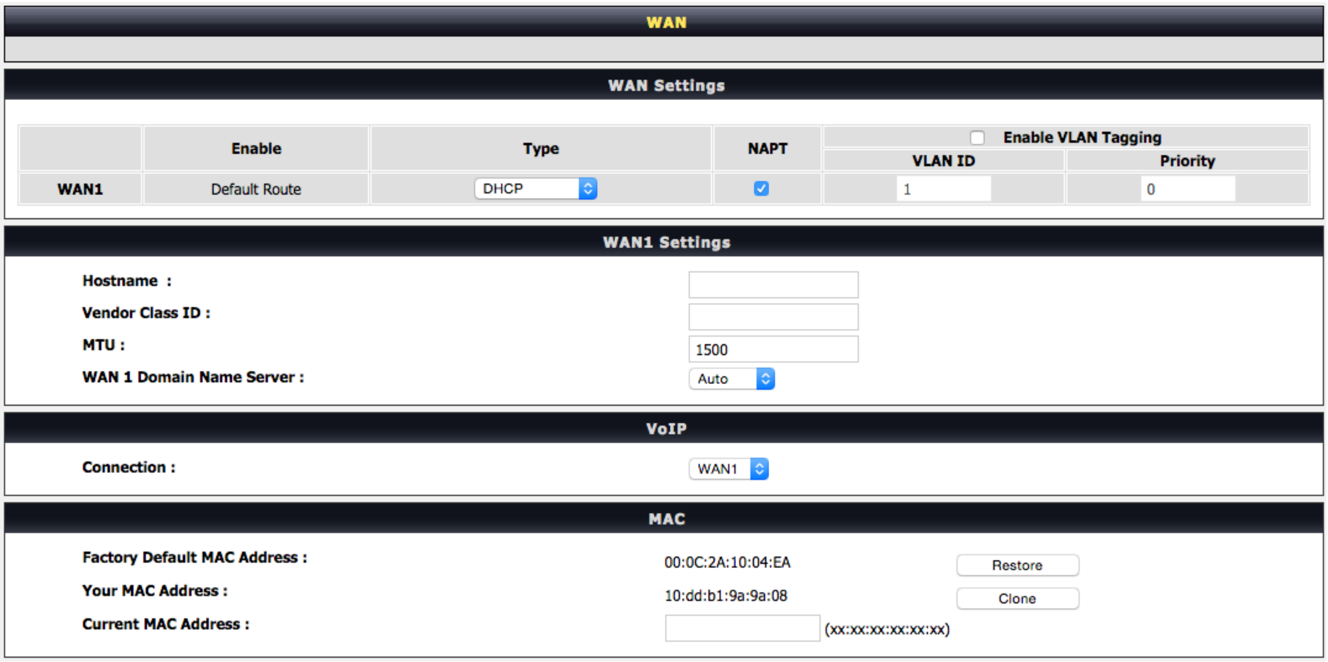
Enter System Settings → Save/ Restart page then **Save** and **Restart**. Wait for about 40 seconds before the Radio Gateway obtains an IP address by the method you selected.

Note: After the new IP address is obtained, the Web Configuration Screen can be entered by entering new IP address at the address bar if you're connecting through WAN Port. The same principle applies to the next two settings.

General Settings → WAN



General Settings → WAN Settings



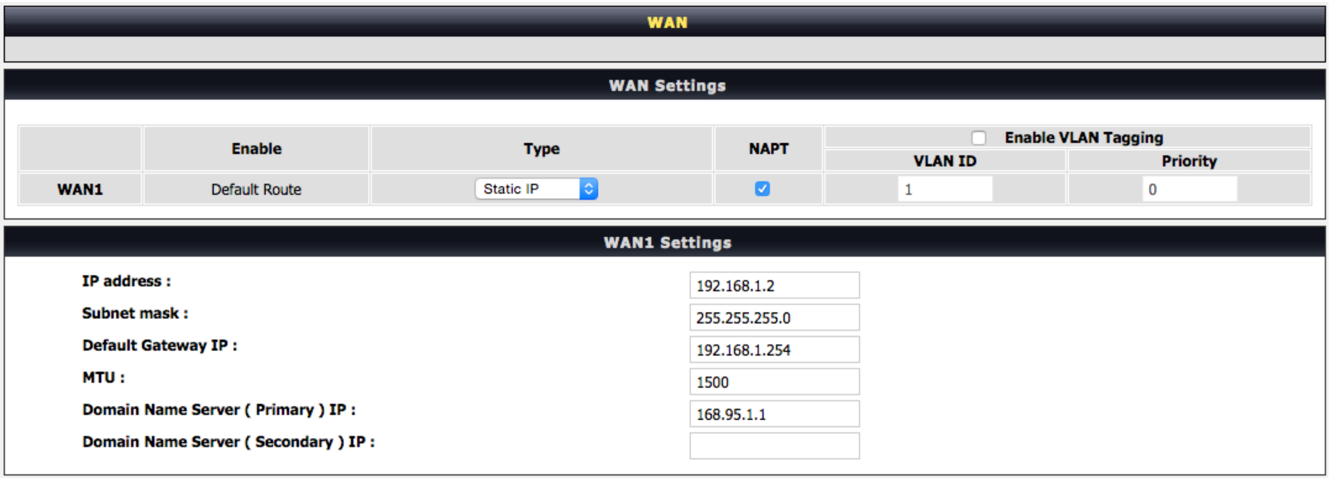
DHCP:

(i) Type → DHCP, then click YES at the bottom of the page

(ii) Some ISP don't require Hostname

(iii) System Settings → Save/Restart → Restart then click yes. After around 40 seconds the IP Address will be obtained from DHCP Server

General Settings → WAN

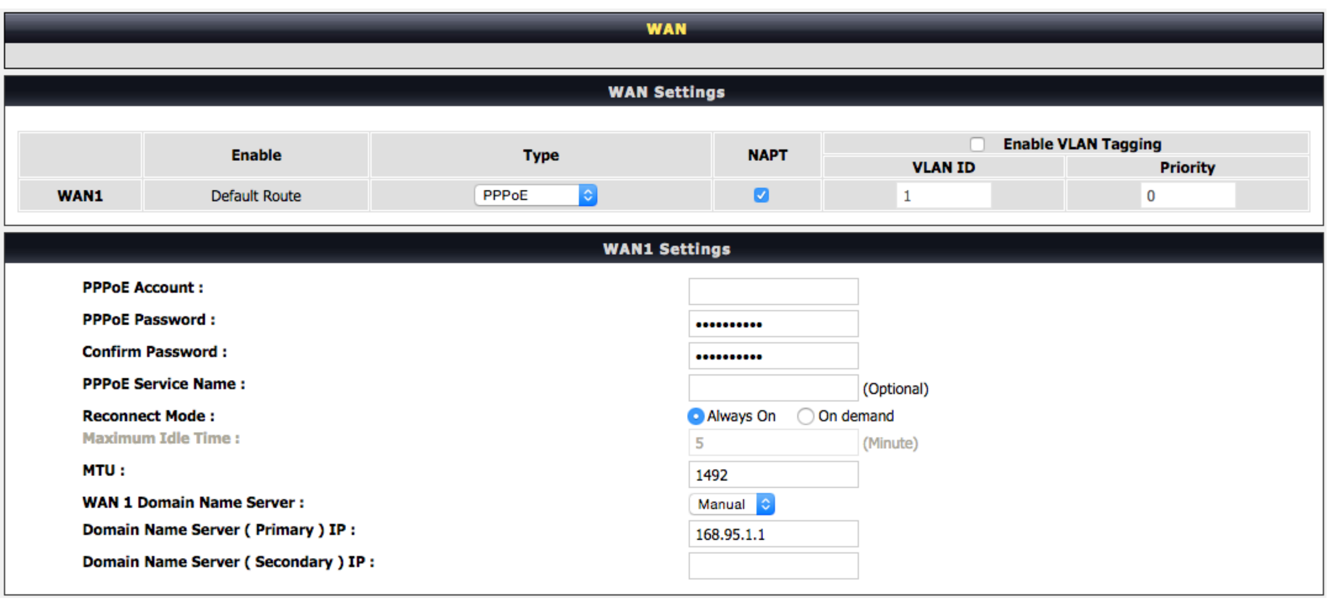


Static IP:

If a fixed IP address is used, please enter the IP Address, Subnet Mask, and Default Gateway provided by ISP.

If the radio gateway is set up under a broadband router, then enter an IP Address that can connect to Internet through this broadband router. If you're not familiar with the settings, then please choose DHCP.

General Settings → WAN

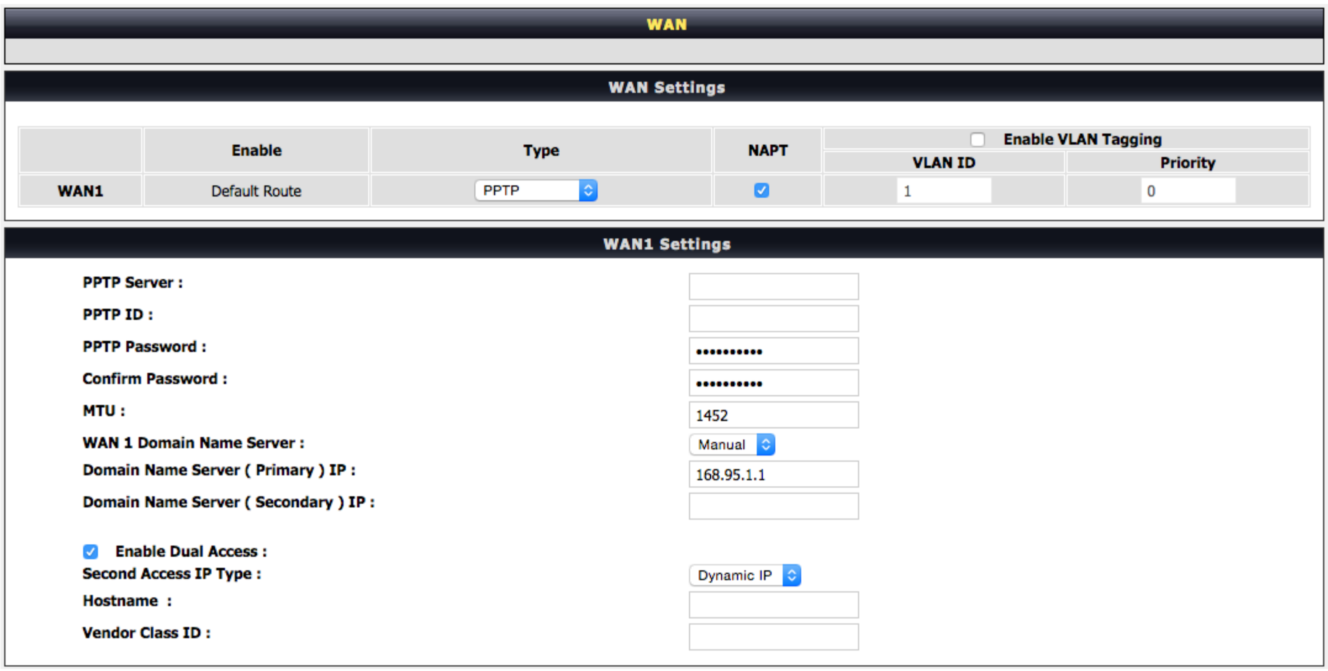


PPPoE (dial-up ADSL):

(i) Type → PPPoE, enter PPPoE Account, Password, Confirm Password, then click **YES** at the bottom of the page.

(ii) System Settings → Save/Restart → Restart then click yes. After around 40 seconds the IP Address will be obtained from Server.

General Settings → WAN

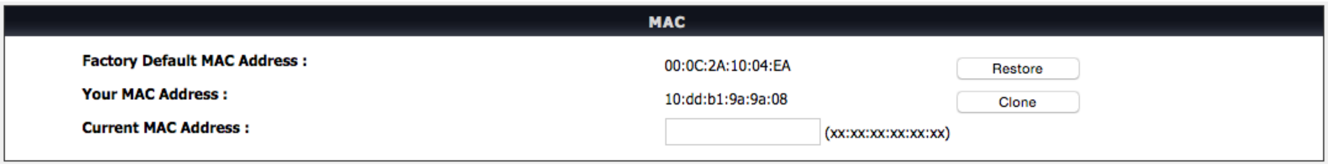


PPTP: Point-to-Point Tunneling Protocol (PPTP) is a WAN connection. Enter the **IP Address**, **Subnet mask**, **PPTP Server**, **PPTP ID** and **Password**.

General Settings → WAN

VoIP Connection Interface: Select a WAN interface for VoIP traffic as WAN is enabled.

General Settings → WAN



If your ISP determines if it's connectable with Internet by MAC (Media Access Control), then please copy Your MAC Address to Current MAC Address. If you want to restore to Radio Gateway's MAC Address, click Restore to get default settings. If you're not sure about this setting, please send inquiry to your ISP.

Note: The outward calls will be unavailable if the settings are incorrect.

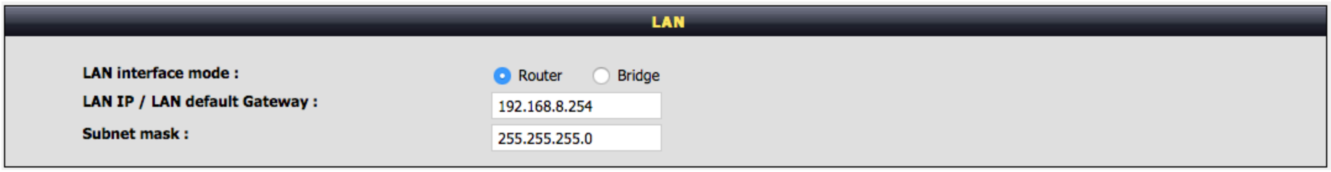
Factory Default MAC: Radio Gateway's MAC.

MAC Address: PC's MAC Address, only detectable when you use LAN to enter Radio Gateway's WEB Page.

Current MAC Address: If it's blank, it indicates currently it's the same with Your MAC Address.

2-2-3 LAN

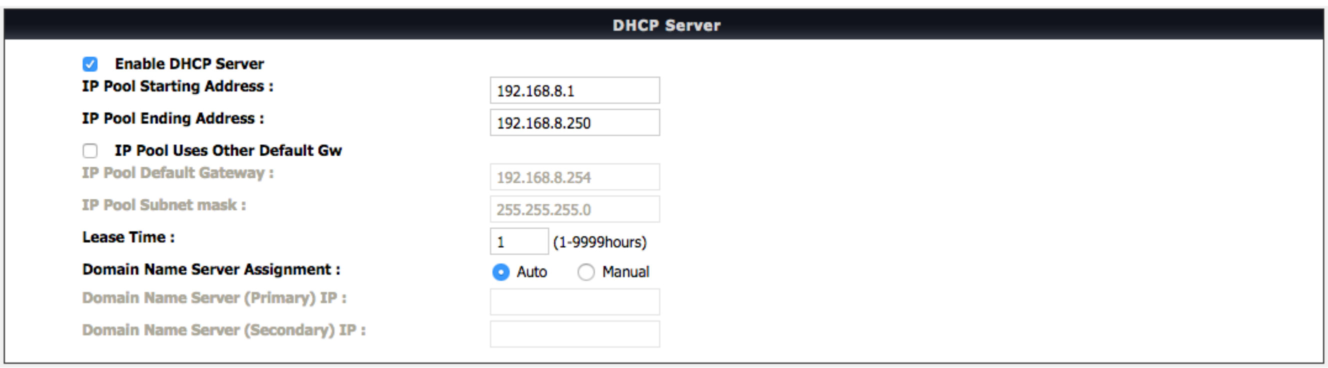
General Settings → LAN



LAN IP / LAN default Gateway : Enter the LAN IP address of the RoIP. It is also the default gateway for DHCP clients.

Subnet Mask: Enter the subnet mask for DHCP clients.

General Settings → LAN



Enable DHCP Server: This variable is to assign the IP address for the devices connected to LAN port of the RoIP.

IP Pool Starting Address/ IP Pool Ending Address: Enter the starting IP address for the DHCP server's IP assignment.

IP Pool Uses Other Default GW: Check the box to assign different default gateway for DHCP clients.

IP Pool Default Gateway: Enter the new default gateway that is different from LAN IP of the RoIP.

IP Pool Subnet mask: Enter the new subnet mask.

Lease Time: Enter the length of time for the IP lease.

Domain Name Server Assignment: Select **Auto** or **Manual** to get the IP address of Domain Name Server assigned by ISP or manually.

Domain Name Server IP: Enter the primary and secondary IP address of Domain Name Server if Domain Name Server Assignment is **Manual**. Otherwise, the RoIP will not be able to access hosts using hostnames instead of IPs.

General Settings → LAN

LAN Port Control					
Port	Enable Port	Incoming Rate Limit	Outgoing Rate Limit	Router/Bridge	VLAN ID
LAN Port 1	☑	Full 0 kbps	Full 0 kbps	Router	0

Enable Port: Tick the box to enable LAN Port.

Incoming Rate Limit: Use the drop-down menu to select the proper rate limit for the specific LAN port. The flow is from LAN to WAN, and the rate limit cannot exceed the real upstream bandwidth.

Outgoing Rate Limit: Use the drop-down menu to select the proper rate limit for the specific LAN port. The flow is from WAN to LAN, and the rate limit cannot exceed the real downstream bandwidth.

Router / Bridge:

Router: The Radio Gateway provides the IP sharing (Router) function.

Bridge: The Radio Gateway provides Bridge function.

The device connected with LAN Port on the Radio Gateway can connect to Internet or VPN via the WAN Port; if applied, server connected to LAN Port and other computer can still use IP of public internet or VPN, in addition the outward packets must be transferred via gateway, combined with QoS in order to ensure the audio quality.

2-2-4 RoIP

On this page, you can set up the internet telephone number of this gateway and IP Address from

Voice Service Provider.

Various SIP Proxy Servers are available.

General Settings → RoIP

Soft Switch Setting
☐ Enable Support of SIP Proxy Server / Soft Switch

Enable Support of SIP Proxy Server / Soft Switch: Check the box to register the RoIP with SIP proxy server or soft switch.

General Settings → RoIP

Line

Line	Type	Number	Hunt Group Port	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password
RoIP Representative Number				☐			•••••••••• ••••••••••
1	RoIP	701 auto	No Group	☐	☐		•••••••••• ••••••••••
2	FXS	702	No Group	☐	☐		•••••••••• ••••••••••

RoIP/FXS: Enter applied SIP number; RoIP and FXS use the same VoIP Frame, but the mode is different. Please do not operate cross-platform.

Hunt Group: Choose the group. “No Group” is single number. When Hunt Group number is the same with line number, that means this line number is the representative number of this line. If it’s different, it indicates as extension number of this Hunt Group. (VoIP Only)

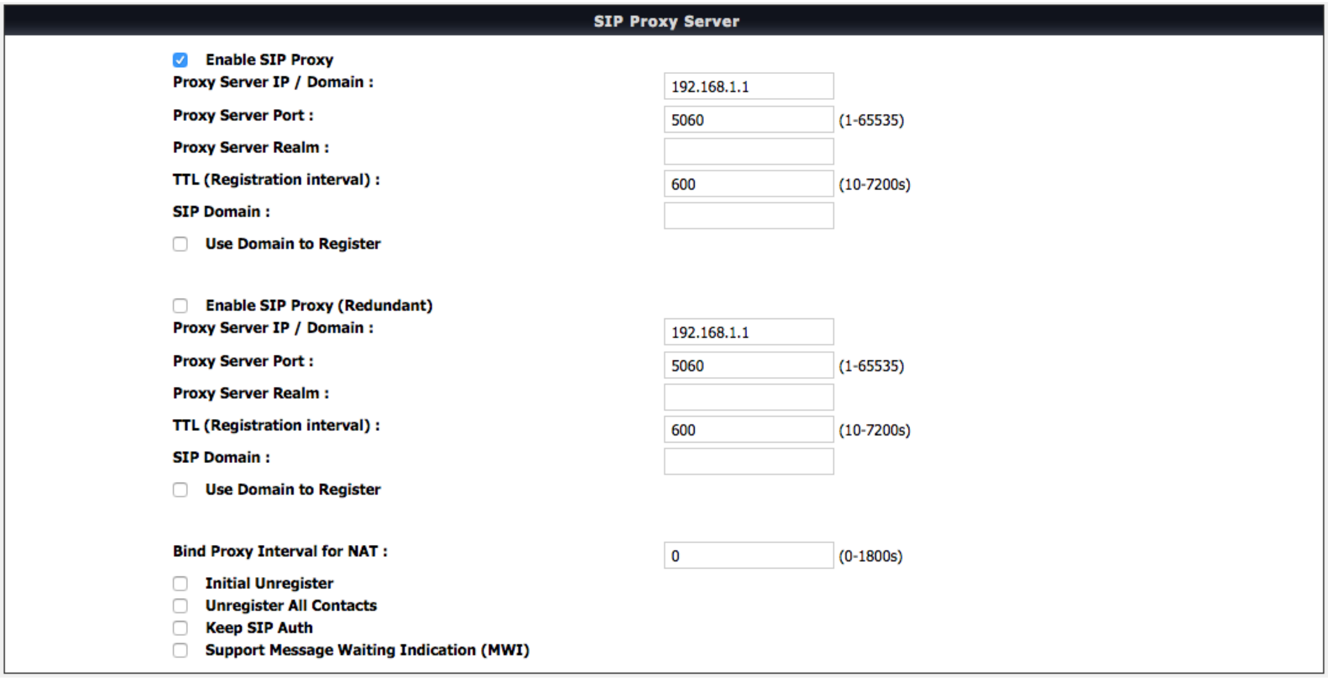
Register: Check the box to register with SIP proxy server.

Invite with ID / Account: The Radio Gateway provides dynamic dial function without registration so there’s no restriction to different Service Providers. Please consult with your system provider before applying this function.

SIP Registered ID / Password: Registered ID and password of this radio gateway.

Note: Please ensure if your VoIP Service Provider allows one account for multi-port using.

General Settings → RoIP



Proxy Server IP/Domain, Port: Enter the IP address or URL (Uniform Resource Locator) and Listen Port of SIP proxy server.

Proxy Server Realm: Most of the VSP aren't required to be set up, please consult with VSP if the call cannot be made.

TTL (Registration interval) [10-7200 s]: The interval for RoIP re-reports to SoftSwitch.

SIP Domain, Use Domain to Register: Area Network specified by VSP (Some VSP doesn't require extra settings. If the connection cannot be done by applying SIP Domain, then IP will be applied instead. Please contact the System Provider if the calls cannot be made.

Bind Proxy Interval for NAT: Check the box to keep the binding exist by sending packets when the RoIP is behind a NAT and SIP proxy server is not able to keep the binding.

Initial Unregister: Check the box to send an unregistered message initially by the RoIP and then it will perform a general register process.

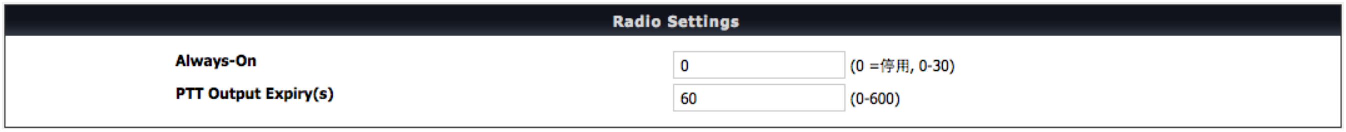
Unregister All Contacts: RoIP sends requests to un-register

Keep SIP Auth: RoIP keeps the last register SIP MD5 authentication information and re-use it for next register request.

Support Message Waiting Indication (MWI): It is used to enable/disable Message Waiting Indication. It is available only when Voice Mail Service is available from the VoIP Service Provider. (VoIP Only)

MWI Subscribe Interval: It is used to set the subscribe time for the RoIP to check the voice mail.

General Settings → RoIP (RoIP Only)

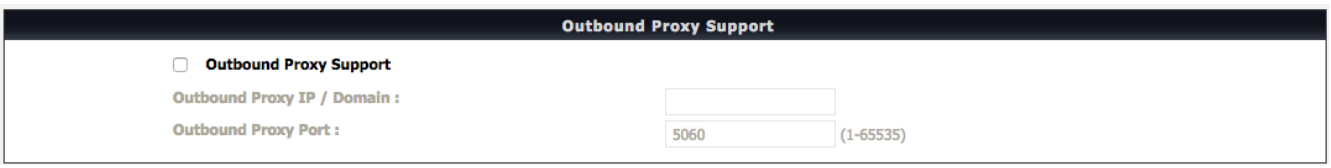


Always-On：Continuous call in time can reduce the verification time.

PTT Output Expiry(s)：Time limitation of RoIP Always-On.

Send “*” After connection: Call Delay Time, suggested default is 0 ms

General Settings → RoIP (VoIP Only)



Outbound Proxy Support: Optional, depends on different Internet Framework. SIP Protocol will be uploaded to specified IP if Outbound Proxy is applied. The applicability of Outbound Proxy depends on different Internet Framework; VSP regulations must be followed to functionalize it fully. Part of Outbound Proxy can pass through firewall. If the gateway is set up within the firewall, you can just enable Outbound Proxy without making extra settings.

Outbound Proxy IP/Domain: Enter the outbound proxy’s IP address or URL.

Outbound Proxy Port: Enter the outbound proxy’s listening port.

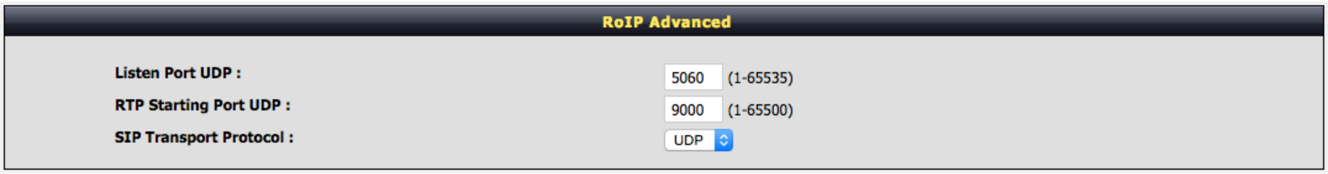
General Settings → RoIP

Enable P-Assert / Privacy Type: Checking the box indicates the dialing number and IP Address of the gateway will be anonymous when VSP transfers the packets of gateway

This function is defined by RFC 3325

2-2-5 SIP Advanced

General Settings → RoIP Advanced



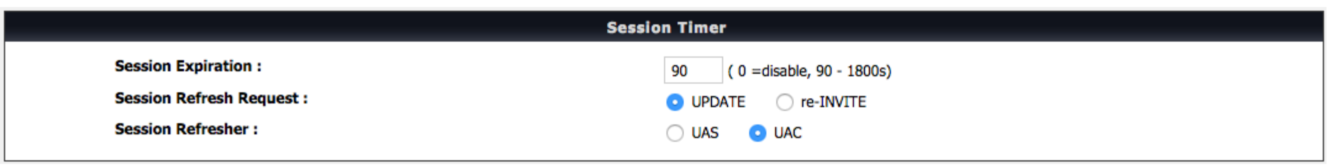
Listen Port UDP: Applied listen port UDP, this column is to set up the Listen Port of gateway to SIP signal

RTP Starting Port UDP: Enter the starting port number or transmitting voice data among VoIP devices. Each line requires 2 ports.

(Ex. If the Port starts at 9000, then Line 1 will occupy Ports 9000 & 9001, Line 2 will occupy 9002 & 9003, and so on.)

SIP Transport Protocol: UDP or TCP

General Settings → RoIP Advanced

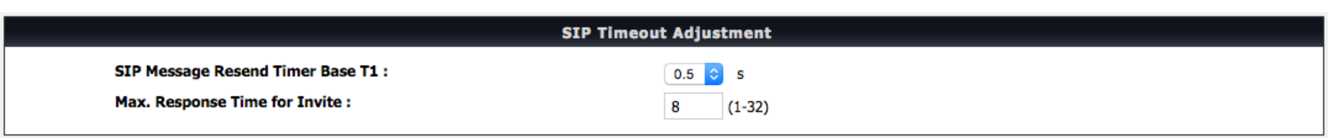


Session Expiration: This field will set the time that the RoIP will allow a SIP session to remain die (without traffic) before dropping it. Default 90.

Session Refresh Request: Select **UPDATE** or **re-INVITE** to send refresh requests to the Server.

Session Refresher: Choose the gateway as UAS or UAC.

General Settings → RoIP Advanced



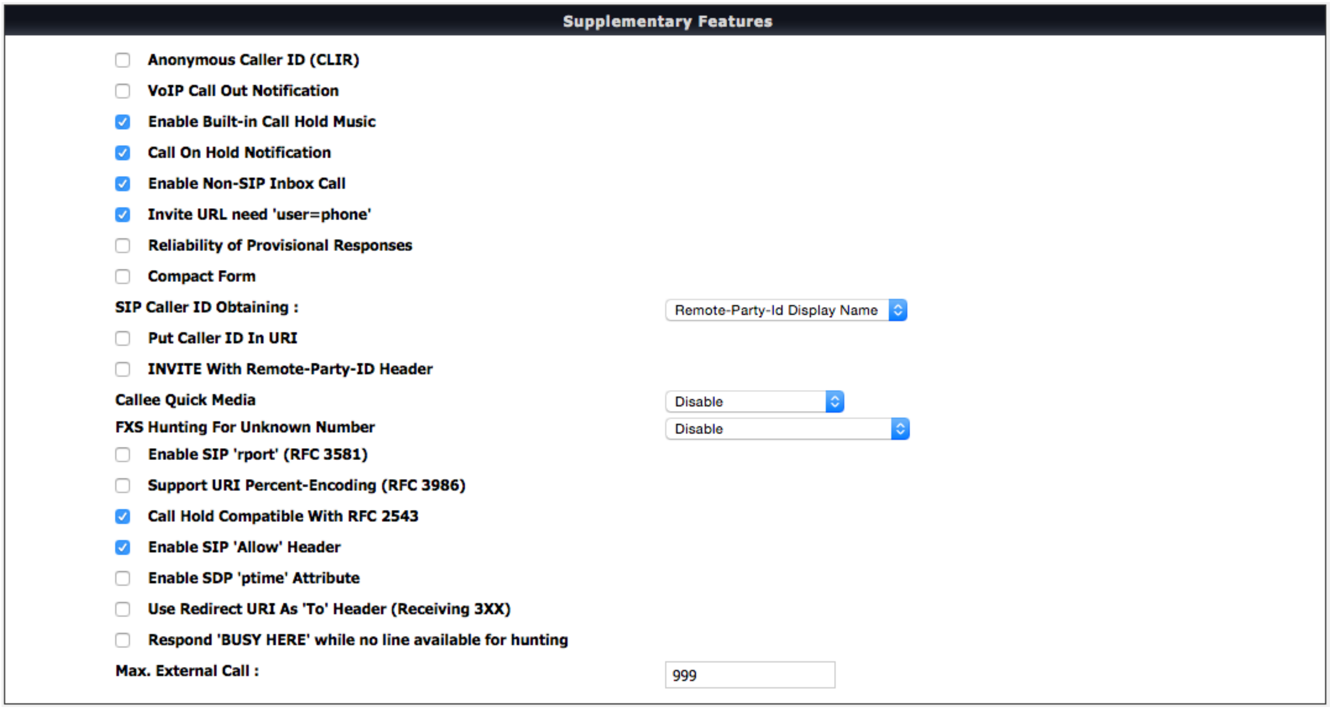
SIP Message Resend Timer Base: Select the resend timer base from the drop-down menu if response is not received within the base time. The sequence of sending is like "base time" * 2, and send again at "base time" *2 *2. The maximum resend time is four seconds. Resend action will stop when the total resend time has reached 20 seconds.

Max. Response Time for Invite: Enter the maximum response time for INVITE packet. When the destination does not reply within the set time, the call is failed.

General Settings → RoIP Advanced (VoIP Only)

VoIP failure announcement: Check the box to play a voice announcement if the RoIP fails to register to the SIP proxy server while FXS is off-hook.

General Settings → RoIP Advanced (VoIP Only)



Anonymous Caller ID (CLIR): Check the box to lock the delivery of the Caller ID to the called party.

VoIP Call Out Notification: Check the box to enable the function of playing a tone to notify user that the call is through VoIP.

Enable Built-in Call Hold Music: Check the box to enable the function of playing music when receiving Call Hold request.

Call On Hold Notification: FXS will send alert to phone set as users hang up if there is a call still held in another line.

Enable Non-SIP Inbox Call: Check the box to make local calls. Local Call here means the call does not go through the Internet and if the dialed number is the extension of other line. You can un-check it to configure as all calls go through the Internet.

Invite URL need 'user=phone': Check the box to add 'user=phone' as a hint that the part left to the '@' sign is actually a phone number.

Reliability of Provisional Responses: Check the box to send back a PRACK request after receiving Packet 180 or Packet 18x with Require: 100 rel. This function is defined by RFC3262

Compact Form: Check the box to show common header field names in an abbreviated form to minimize the size of packets.

SIP Caller ID Obtaining: Select the part of the SIP packet from the RoIP to obtain Caller ID. There are several places where the Caller ID is located.

Remote-Party-ID Display Name - It is located at SIP → Remote-Party-ID → Before [<sip:]
Remote-Party-ID User Name - It is located at SIP → Remote-Party-ID → After [<sip:], Before [@]
From-Header Display Name - The standard way is in SIP → Message Header → From → SIP Display info.

From-Header User Name - It locates at SIP -> Message Header -> From -> SIP from address before [@].

Put Caller ID In URI: This feature is to put Caller ID in URL. The Caller ID is located in SIP → Message Header → After [From:], Before [<sip:] by default settings. It will be located in SIP → Message Header → After [<sip:], Before [@]if ticked.

INVITE With Remote-Party-ID Header: Check the box to comprise the information of Remote-Party-ID in the message header of INVITE. Different format of INVITE header might cause the call not to be connected. Please consult with your VoIP Service Provider before enabling it.

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

Call Hold Compatible With RFC 2543: It is used to set the procedure of Call Hold being compatible with RFC 2543.

Enable SIP 'Allow' Header: It is used to put "Allow" in SIP packets. The Allow header field lists the SIP requests supported by ITA when ticked.

Enable SDP 'ptime' Attribute: It is used to put "ptime" in SDP packets when ticked.

Use Redirect URI As 'To' Header (Receiving 3XX): It is used to change the content of 'To' header field when receiving 3XX.

Respond 'BUSY HERE' while no line available for hunting: It is used to reply 'BUSY HERE' to the calling party while no line is available for hunting.

General Settings → RoIP Advanced (RoIP Only)

Radio Settings		
VAD Open Threshold	15	(0-255)
VAD Shut Threshold	9	(0-255)
VAD Detect Times	20	(0-500)
Input Gain Detect	-12	(-12-33)
Output Gain Detect	-15	(-57-6)

Enable VAD: Check to activate software VAD

VAD Open Threshlod：Maximum detective sensitivity of radio audio

VAD Shut Threshold：Minimum detective sensitivity of radio audio

VAD Detect Times：Time of detective sensitivity of radio audio

Input Gain Detect：RoIP Codec input Gain

Output Gain Detect：RoIP Codec output Gain

2-2-6 Hot Line

General Settings → Hot Line

Hot Line

Line

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0=disable,0-60s]	FXS Group (0:Disable)
1	☑	RoIP	☐		0	1
2	☑	FXS	☐		0	2

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

Hot Line: Check to direct the call automatically to a pre-configured destination without any action when the FXS is off-hook. (i.e. as the user picks up the phone). When the FXS is under Hot Line mode, no other phone numbers can be dialed.

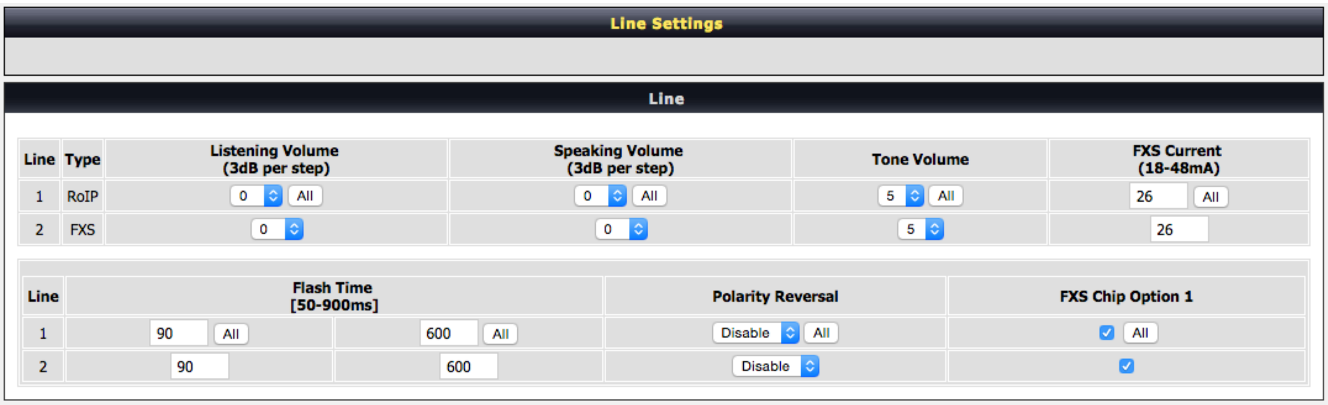
Hot Line No.: Enter hot line no. for automatic dialing. If there are more numbers than columns, you can use Speed Dial function of Digit Map.

Warm Line: Enter the time for the call to start with a pause, so the user can dial another number. The call will be automatically directed to the pre-configured destination within timeout period.

FXS Group: When there is an incoming call and the ATA will automatically assign an unassigned call according to the Hunting Priority. If Port 2 does not want to be set as an assigned line to receive any inbound calls, the function can be disabled.

2-2-7 Line settings

General Settings→ Line settings



Listening Volume: Use the drop-down menu to adjust the hearing (listening) volume.

Speaking Volume: Use the drop-down menu to adjust the speaking volume.

Tone Volume: Use the drop-down menu to adjust the tone volume. It will apply to all tones generated by the RoIP/FXS including Dial Tone, Ring Back Tone and Busy Tone.

FXS Current: Set the output D.C. current of FXS port.

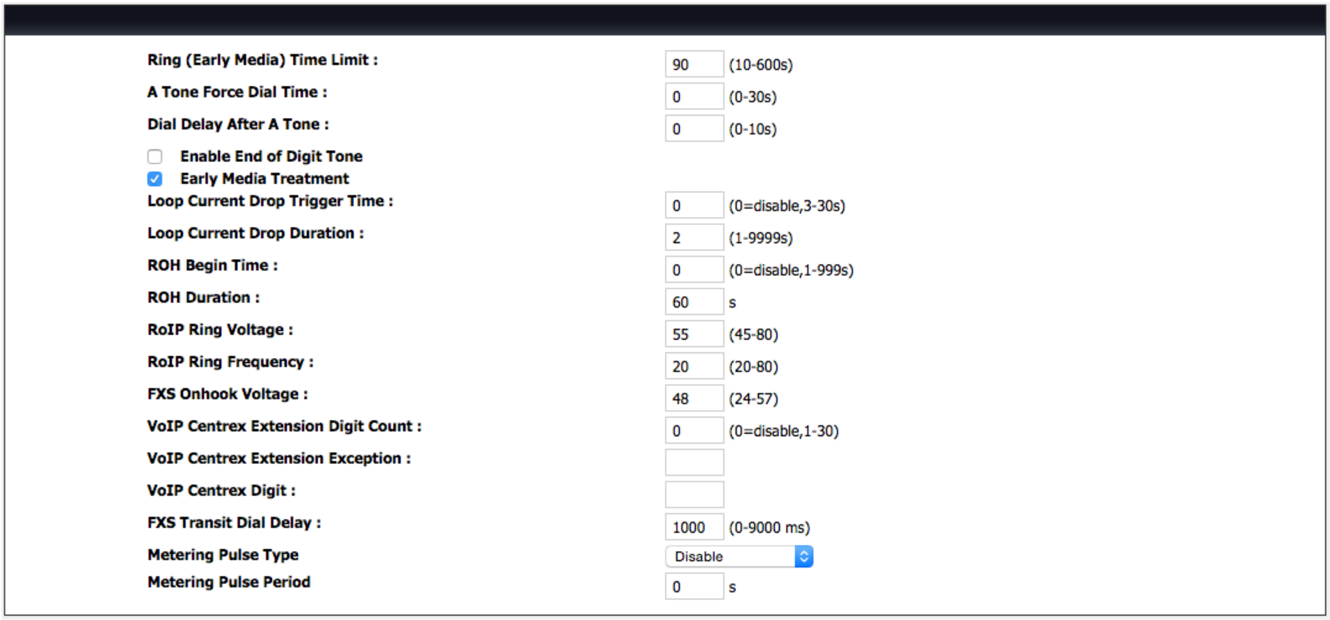
Min. FXS Hook Flash Time: Enter the minimum flash time for FXS detecting. When the flash signal generated by the phone set is shorter than Min. FXS Hook Flash Time, FXS port will be on-hook.

Flash Time: Enter the maximum flash time for FXS detecting. When the flash signal generated by the phone set is longer than the Flash Time, FXS port will be on-hook.

Enable Polarity Reversal: Check the box to activate the generation of polarity reversal from FXS.

FXS Chip Option 1: Check the box to avoid mis-detecting the loop state of a subscriber line or PBX user loop from FXS interface. In some cases, the off-hook voltage might cause the FXS interface mis-detect the idle and the active state, in order to avoid this situation, un-check this feature.

General Settings→ Line settings (VoIP Only)



Ring (Early Media) Time Limit: Enter the timeout to cancel a call if no one answers the phone.

A Tone Force Dial Time： Waiting time for A Tone after dialing, the number will be dialed forcibly after set up time

Enable End of Digit Tone: Check the box to activate the function of playing a “Beep-Beep” tone to notify the user that the call is in progress.

Loop Current Drop Trigger Time: Enter the time to avoid the line being engaged when FXS port is connected to PBX. It stops the loop current from FXS port when FXS port is playing busy tone. The setting “0” zero is to disable this function.

Loop Current Drop Duration: Enter the drop duration for loop current.

ROH Begin Time: As users forget hang up phone set it makes FXS play loud Howler Tone to notify users put hand set correctly. If this timer is set to be 20 seconds, that FXS play busy tone for 20 seconds then play ROH.

ROH Duration: The remind audio will be displayed for 60 seconds to remind the user to hang up. It will stop displaying automatically when the time is up.

FXS Ring Voltage: It is to set the Ring Voltage of FXS.

FXS Onhook Voltage: It is to set the Onhook Voltage of FXS.

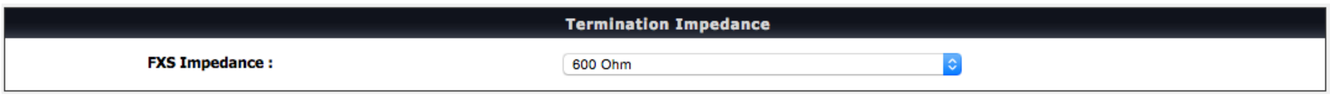
Attention: After disabling this function, FXO cannot recognize available lines. The Internet incoming calls may not be available through FXO. It is suggested to disable the unconnected lines to avoid this situation.

VoIP Centrex Extension Digit Count: This feature is to enable and set the digit count of VoIP Centrex. The setting “0” zero is to disable this function.

VoIP Centrex Digit: Enter the digit for VoIP call. If you dial VoIP Centrex Digit first, the dialing plan is according to the Digit Map; otherwise the FXS will send the number which digit count is the same as VoIP Centrex Extension Digit Count.

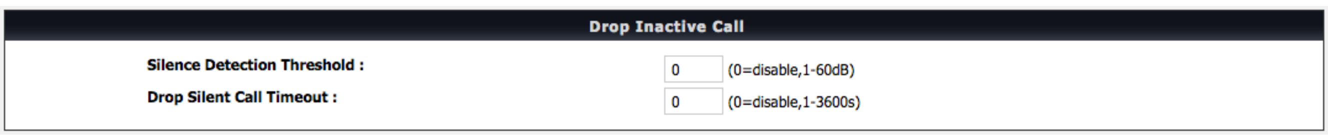
Metering Pulse Type/ Metering Pulse Period: It is used for telephony device which connected to FXS port for billing purpose. **FXS provide Polarity Reversal、 12k Hz and 16k Hz metering capacity. The fully support for detail Metering Pulse Period is not free charge, please contact with your vendor.**

General Settings→ Line settings (VoIP Only)



FXS Impedance: Select different impedance from the drop-down menu.

General Settings→ Line settings (VoIP Only)



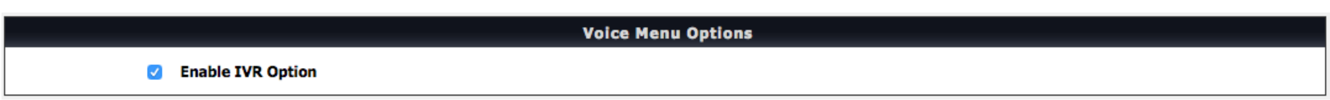
This feature is a call drop standard for a FXS to determine whether or not to hang up the phone. The RoIP will disconnect the call automatically to avoid keeping the line engaged if the detected volume is below the **Silence Detection Threshold** or the time exceeds the **Drop Silent Call Timeout**.

Silence Detection Threshold: When the detected audio is lower than the set-up threshold, the connection will be hung up automatically.

Drop Silent Call Timeout: The system will hang up automatically when the detected audio is lower then threshold and longer than timeout duration to avoid occupied lines.

Note: Improper values for above settings might cause unexpected automatic disconnection of a call. Default values are recommended.

General Settings→ Line settings (VoIP Only)

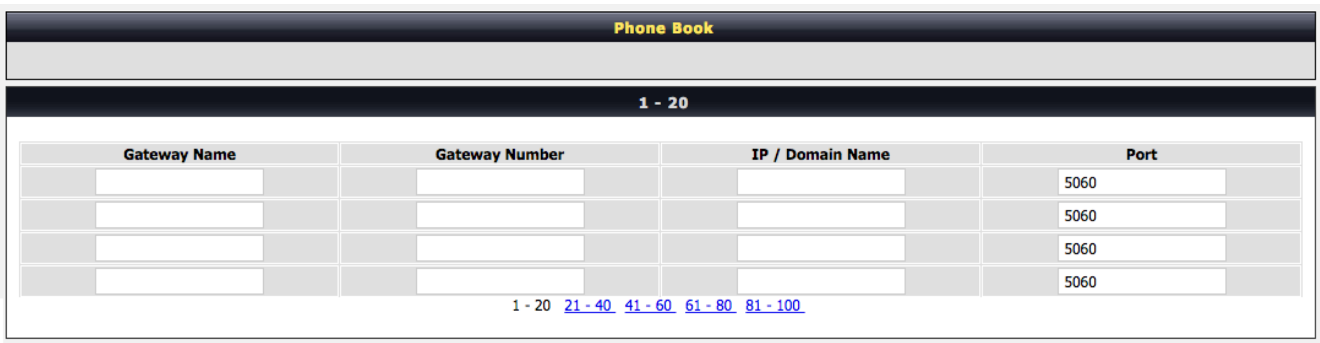


Enable IVR Option: Check the box to enable IVR function. Once disabled, it's unavailable to use **password# to acquire or modify settings. (It is suggested not to use IVR and Radio together to avoid anomaly.)

2-2-8 Phone Book

Phone Book: 100 sets of built-in contact lists to look for others' IP Addresses when dialing other VoIP equipments. If VSP is not applied to one of the groups, then all VoIP equipments have to set up contact information of one another to make the calls available.

General Settings → Phone Book



Gateway Name: Enter the alias of the remote peer.

Gateway Number: Enter the phone number of the remote peer.

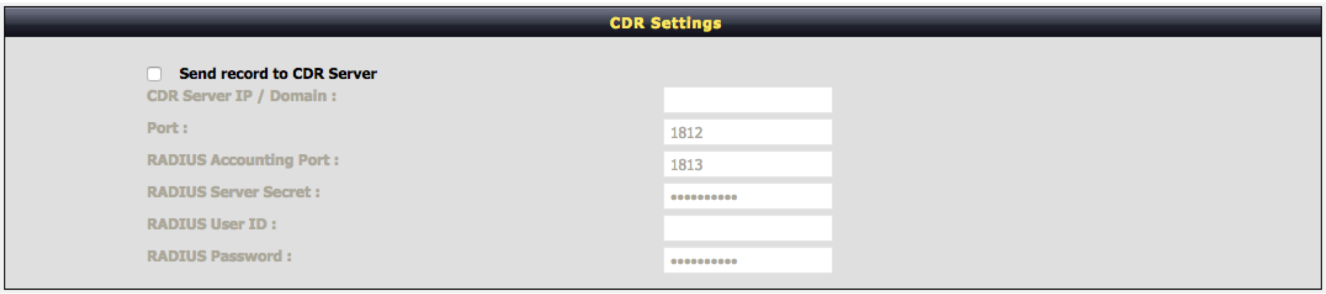
IP / Domain Name: Enter the IP address or URL (Uniform Resource Locator) of the remote peer.

Port: Enter the listen port of the remote peer.

2-2-9 CDR Settings

CDR provides call details records in TXT form. You can import the TXT file and write analysis based on requirement.

General Settings → CDR



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

CDR Server IP / Domain: Enter the IP address or domain name of the CDR server.

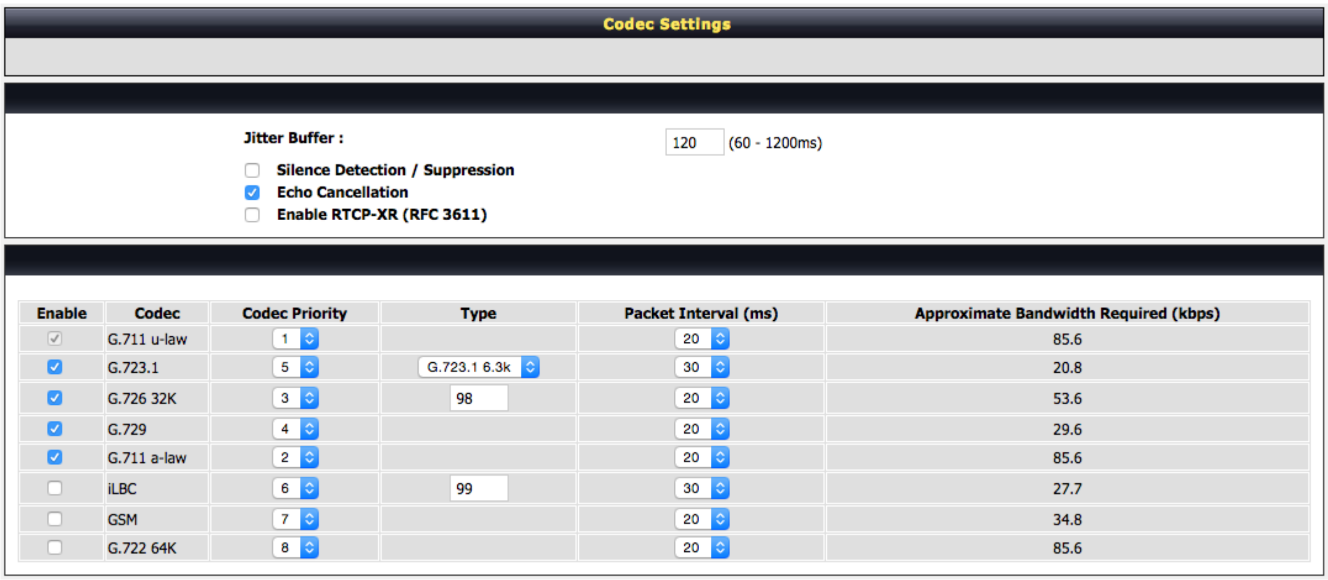
Port: Enter the listen port of the CDR call details record server.

RADIUS: Tick the checkbox to enable RADIUS as database and enter the necessary information of RADIUS including RADIUS Accounting Port, RADIUS Server Secret, RADIUS User ID and RADIUS Password.

2-3 Advanced Settings

2-3-1 Codec setting

Advanced Settings→ Codec settings



Codec Settings

Jitter Buffer : 120 (60 - 1200ms)

Silence Detection / Suppression

Echo Cancellation

Enable RTCP-XR (RFC 3611)

Enable	Codec	Codec Priority	Type	Packet Interval (ms)	Approximate Bandwidth Required (kbps)
✓	G.711 u-law	1		20	85.6
✓	G.723.1	5	G.723.1 6.3k	30	20.8
✓	G.726 32K	3	98	20	53.6
✓	G.729	4		20	29.6
✓	G.711 a-law	2		20	85.6
	iLBC	6	99	30	27.7
	GSM	7		20	34.8
	G.722 64K	8		20	85.6

Jitter Buffer: Enter the jitter of receiving packets.

Silence Detection / Suppression: Check the box to enable the silence packets and send less voice data (package) during the silent period while talking.

Echo Canceling: Check the box to remove echo and improve voice quality during conversation.

Enable RTCP-XR (RFC 3611): Enable RTCP-XR(RFC-3611) to report network quality.

Codec: Check the box to codec for the RoIP to support. All codecs are selected and supported by default. You can un-check the box that is not used.

Codec Priority: The priority of code for communication.

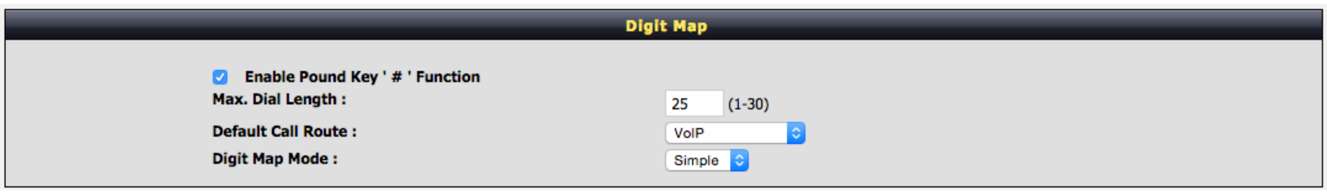
Packet Interval: Select the frame size of voice package from different codec. It defines the time interval for the RoIP to send a RTP packet or voice packet to the receiving side. The smaller the value, the greater the bandwidth takes, and larger values might cause voice delay.

Approximate Bandwidth Required: It shows the bandwidth required from different codec and packet interval.

2-3-2 Digit Map

Digit Map supports setting up order for different leading digits, in-/decrease digits to dial, or order for router dialing out based on different dialing rules of each ITSP. You can set up the rules based on applied situations.

Advanced Settings → Digit Map



Enable Pound Key ' # ' Function: Check to enable finishing dialing by pressing #.

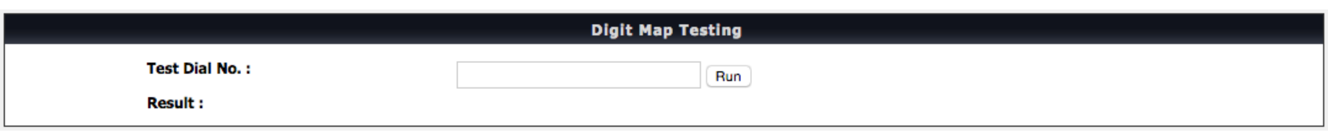
Default Call Route: Defines the default call route of the RoIP.

VoIP: The call route is VoIP only.

Deny: The call will be denied

Default VoIP Route Profile: Enter the Profile ID (ranging from 1-10) for the Default VoIP routing.

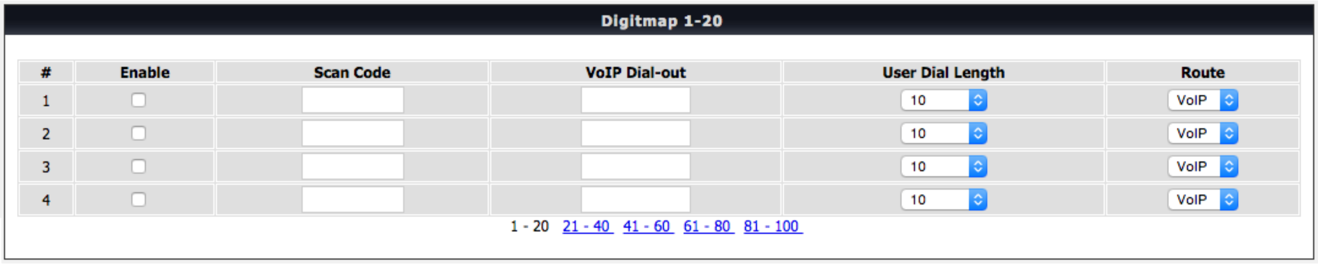
Advanced Settings → Digit Map



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

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

Advanced Settings → Digit Map



Enable: Check to enable this set of Digit Map

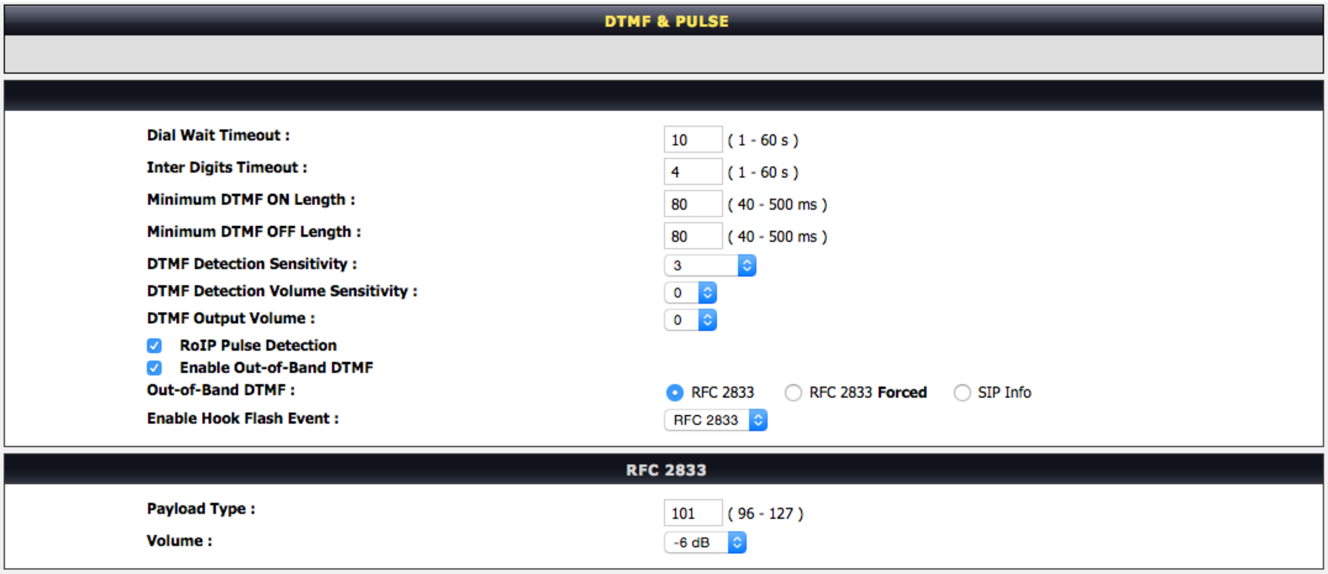
Scan Code: Enter the leading digits or short digits for dialing.

VoIP Dial-out: Enter the total digits of the dialed number.

User Dial Length: The maximum digits for dialed number are 25 digits. 0 indicates no maximum limitation. When the set-up dial length is dialed (pressing # is not required), the system will dial out through selected router. If the setting is 0, pressing 0 is required to speed up the system to dial out from selected router.

2-3-3 DTMF & PULSE

Advanced Settings → DTMF & PULSE



Dial Wait Timeout: Enter the timeout duration after the user picks up the phone set. If the first digit is not pressed after timeout, the system will display busy audio. Picking up or dialing from FXO, the 1^{st} digit must be pressed before timeout, or the system will display busy audio.

Inter Digits Timeout: Enter the timeout duration between the intervals of each key pressed. When exceeding the set timeout duration without entering further digits, the numbers entered will be dialed out automatically.

Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones): To adjust the speed of dialing other phone equipments (time of dial on and off.) If the dials fail too often, or having errors when dialing bank voice system, you can try to adjust the length to 100 or 120.

DTMF Detection Sensitivity: To adjust the sensitivity of receiving other telecommunication equipment dials. It might miss out the digits when the value is too low, and too many digits when the value is too high. Please adjust based on applied situation.

DTMF Detection Volume Sensitivity: Adjust the sensitivity of displaying numbers.

DTMF Output Volume: Adjust the Tx volume.

RoIP Pulse Detection: To enable detection of PULSE dial sends from a phone set.

Enable Out-of-Band DTMF: To make sure detected correctness of pressing buttons during the calls. It can be applied to computer audio service calls (ticket purchasing, bank service through phone calls, and etc.) Please consult with ITSP when using RFC 2833 or SIP Info. You can choose payload type and volume adjustment when RFC2833 is applied.

Enable Hook Flash Event: Send Flash to remote devices by using RFC2833 or SIP Info

Deactivate: Don't send out Hook Flash

Automatically: The sent Hook Flash will follow the format of DTMF used by Out-of-Band

RFC2833

Payload Type: payload type of RFC2833, default: 101

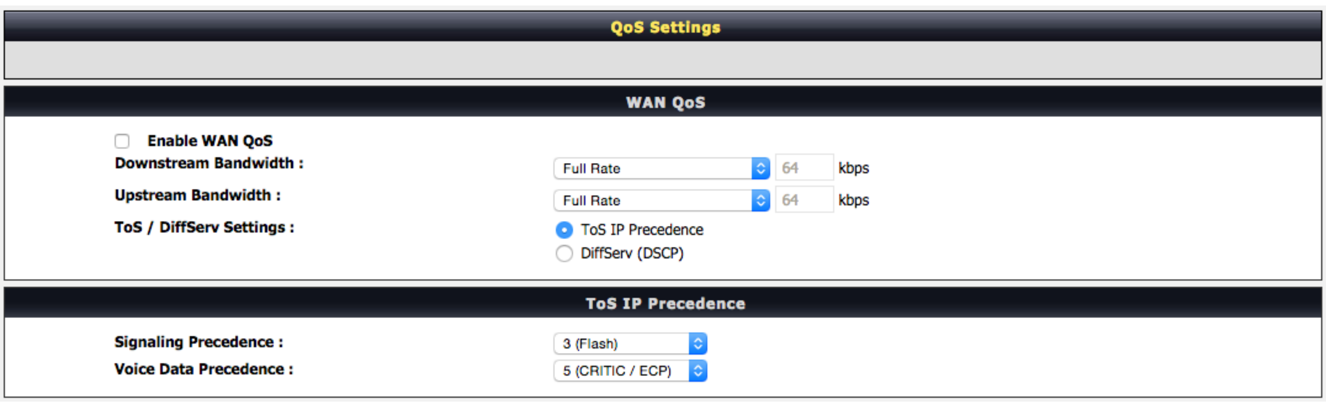
Volume: Select the volume of RFC 2833 from the drop-down menu.

2-3-4 QoS Settings

QoS cannot guarantee 100% audio quality. Theoretically when requested bandwidth is more than actual bandwidth, the audio quality will be compromised. QoS only have guaranteed-priority to send out the voice packet, but cannot guarantee the remote device can receive and play the audio successfully. In addition, Qos cannot choose and request which downloaded packet to receive (in priority). For priority options, ToS/DSCP must be activated.

Note: Not all routers are equipped with ToS/DSCP, if not; the remote VoIP devices cannot receive the audio packets with priority.

Advanced Settings → QoS



Enable WAN QoS: Check the box to guaranty the voice quality. Voice packets have the highest priority in IP networks, and the data transmission is distributed to less bandwidth.

ToS IP Precedence / DiffServ (DSCP): After activated, the transmitting device will cooperate outer Switch/Router to send the audio packet with priority; the system will set up Precedence parameter at TOS or DSCP parameter at DS. The priority is more supreme if the parameter is higher. The parameter of Signaling (SIP Signal) and Voice (RTP) can be set up individually.

ToS IP Precedence: Select the precedence for signaling (data) and voice (voice data).

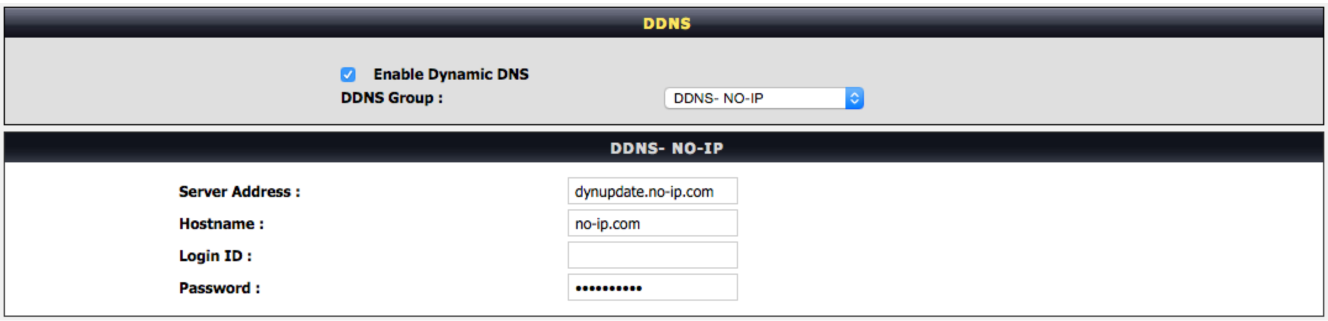
DiffServ (DSCP): Select the number of signaling (data) and voice (voice data) values.

Note: For the VoIP, ToS IP Precedence and DiffServ are the same function. You only select one for priority marking.

2-3-5 DDNS

DDNS (Dynamic Domain Name Service) Settings aim to solve the problem that Radio Gateways cannot be set under NAT using dynamic IP. Only activate this setting when the Radio Gateway is set under NAT using dynamic IP and without supporting DDNS.

Advanced Settings → DDNS



Choose DDNS Group: Available for DynDNS, TZO, PeanutHull, NO-IP or DDNS Server from other agencies. If it's the latter one, please apply the account in advance.

Server address: IP or URL of registered DDNS Server

Hostname: Enter the URL of the system (or NAT) – applied from domain name registration providers (e.g. VoIPGateway01.dyndns.org).

Login ID/Password: Enter the Login ID and password used to log-in to the DDNS server.

Automatic IP Detection: Only activate it when it's set up under NAT

Note: If the Radio Gateway is set up under NAT, then enter the hostname in the NAT IP/Domain that is the same as the Hostname of the DDNS.

2-3-6 NAT Traversal

If your ITSP is equipped with Firewall function, then this setting is not necessary. Please set up NAT Traversal if your ITSP is built under NAT Frame (under other broadband router), or using P2P (Peer-to-Peer) without VSP function. Now the IP used by Radio Gateway should be virtual IP.

Advanced Settings → NAT Traversal

Enable (NAT Public IP): Check the box to enable NAT.

NAT IP/Domain:

If your broadband router use fixed public IP, then enter the fixed public IP in NAT IP/Domain. If your broadband router doesn't use a fixed IP, then enter the URL with DDNS.

> **Note:** If you are setting a public IP in this field, it has to be a static public IP, otherwise VoIP communication may not be established properly. Please contact your ISP to check if your Internet connection has static public IP addresses.
>
> **Gateway Port (UDP):** 5060
>
> **RTP Port (UDP): Start Port is 9000 (2 Port for each line)**
>
> **Public Phone Book Admin Port: 1690**
>
> Http Port (TCP): 80 (You can skip this if you're not setting this remotely.)

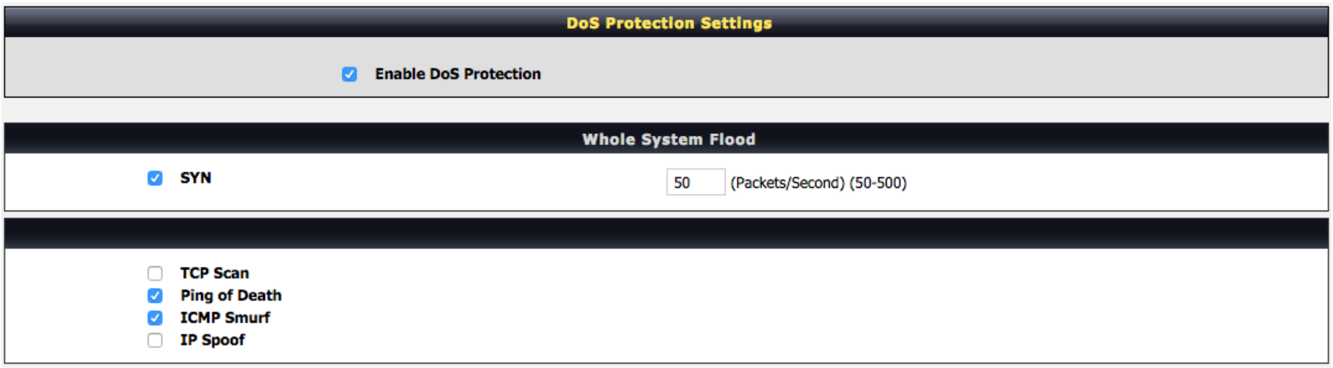
Enable STUN (Simple Traversal of UDP over NAT) Server: Check the box to use the STUN protocol prevents problems from setting the IP sharing function. (Some NATs do not support this protocol.)

STUN Server IP/Domain and Port: Enter the IP address and listen port of the STUN server.

2-3-7 DoS Protection Settings

UDP (User Datagram Protocol) is applied in VoP (Voice-over-Packet), please don't check "UDP" if your calls are aborted.

Advanced Settings → DoS Protection Settings

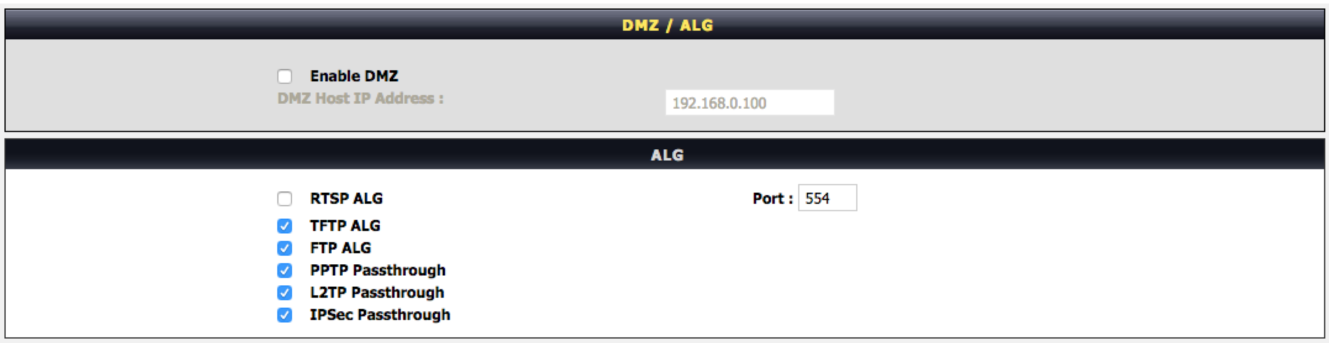


Enable DoS Prevention: Check the box to prevent DoS attacks from WAN. There are various types of DoS attacking. Leave settings in this field to the default if you are not familiar with it.

2-3-8 DMZ / ALG

DMZ (Demilitarized Zone) allows the server on the LAN site to be directly exposed to the Internet for accessing data and to forward all incoming ports to DMZ Host. Adding a client to the DMZ may expose that computer to a variety of security risks; so only use this option as a last resort.

Advanced Settings → DMZ /ALG



Enable DMZ: Check the box to enable DMZ feature.

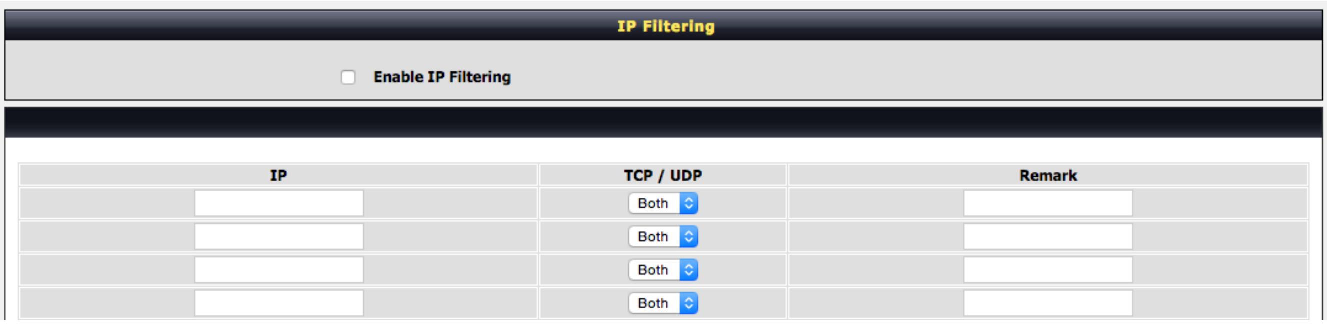
DMZ Host IP Address: Enter the IP address of that computer as a DMZ Host with unrestricted Internet access.

RTSP ALG: Enable ALG for RTSP multimedia stream.

2-3-9 IP Filtering

IP Filtering is to restrict internal users to access the Internet

Advanced Setting → IP Filtering



Enable IP Filtering: Check the box to enable IP Filtering

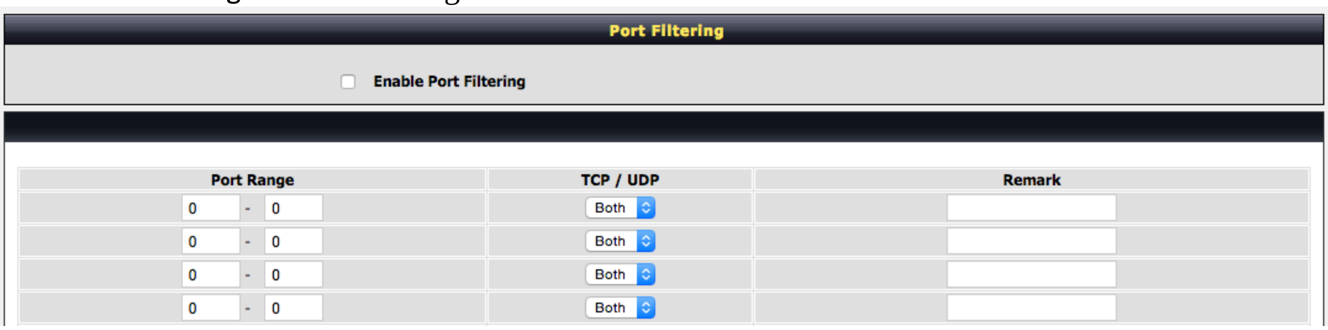
IP: Enter the IP address you'd like to filter out. The restricted IP cannot send data to the Internet.

TCP/UDP: The Communications Protocol, you can select TCP, UDP or Both.

2-3-10 Port Filtering

Port filtering allows you to control all data transmitting over routers. When the ports used by the source are within the restricted scope, they will be filtered without transmission.

Advanced Setting → Port Filtering



Enable Port Filtering: To enable port filtering

Port Range: Enter the port range that will be denied access to the Internet. If it's 80, and the protocol is set up with Both or TCP, all computers cannot use http (port 80) service – surfing on the Internet is restricted.

TCP/UDP: The Communications Protocol, you can select TCP, UDP or Both.

2-3-11 MAC Filtering

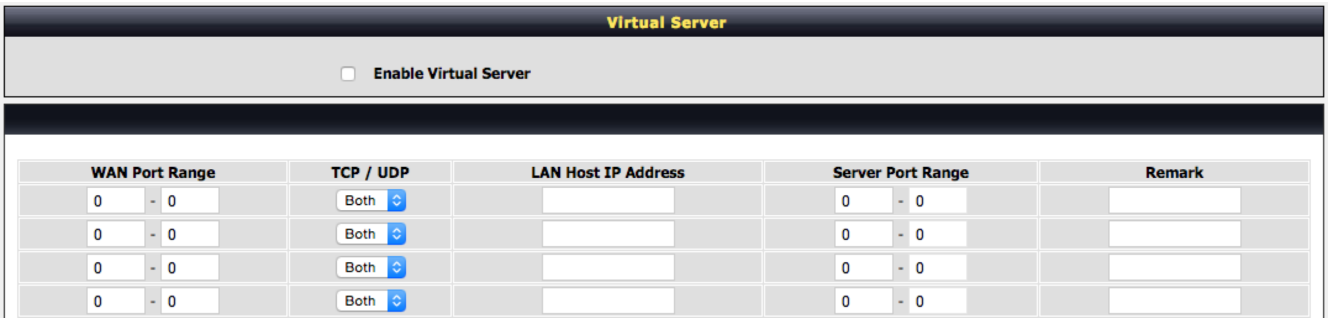
Advanced Settings → MAC Filtering

Enable MAC Filtering: Enter a MAC address to prevent particular devices from accessing and transmitting data to Internet.

2-3-12 Virtual Server

Enable users on Internet to access the WWW, FTP and other services. When remote users access Web or FTP servers through WAN IP address, they will be routed to internal LAN Server, and send the packets properly to internal LAN Server based on external requirement.

Advanced Settings → Virtual Server



WAN Port Range: Enter the port range for the WAN side.

TCP/UDP: Select the communication protocols used by the server, **TCP**, **UDP** or **Both**.

LAN Host IP Address: Enter the IP address of the device that provides various services.

Server Port Range: Enter the port range used by the LAN host.

Remark: Enter comments

2-3-13 UPnP

Advanced Settings → UPnP

Enable UPnP Server: To support plug-and-use

2-3-14 IGMP

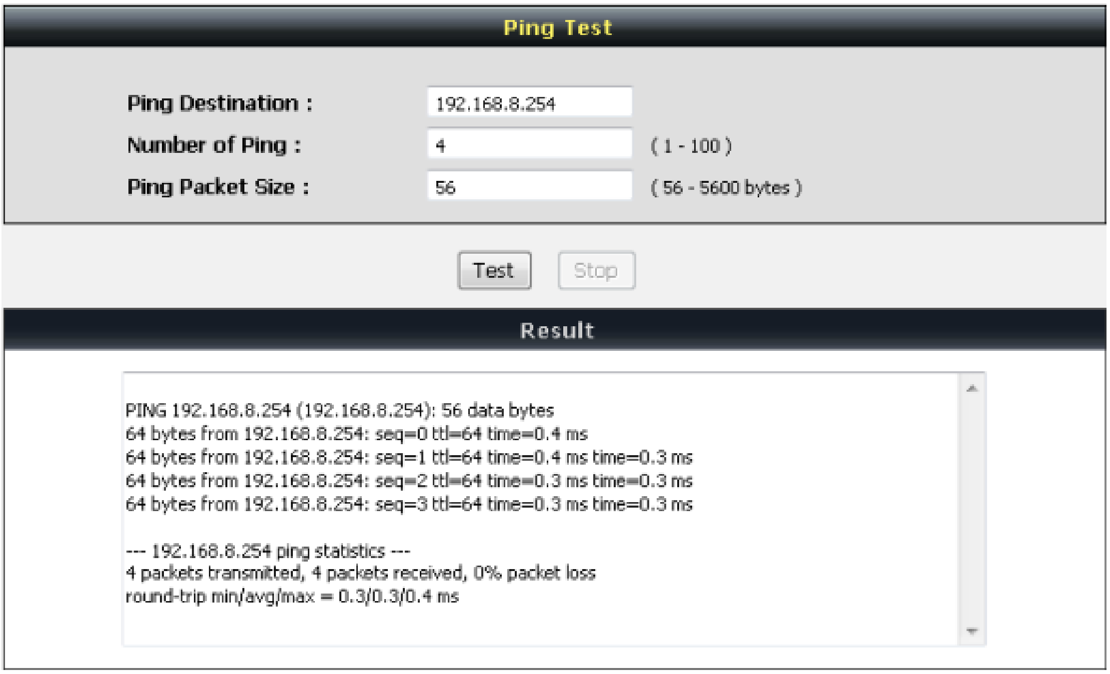
Enable IGMP Proxy：Check to enable Internet Group Management Protocol

2-4 Tools

2-4-1 Ping Test

Use “Ping” to verify if a remote peer is reachable. Enter a remote IP address and click “Test” to ping the remote host. The result would be shown on **Result** Table.

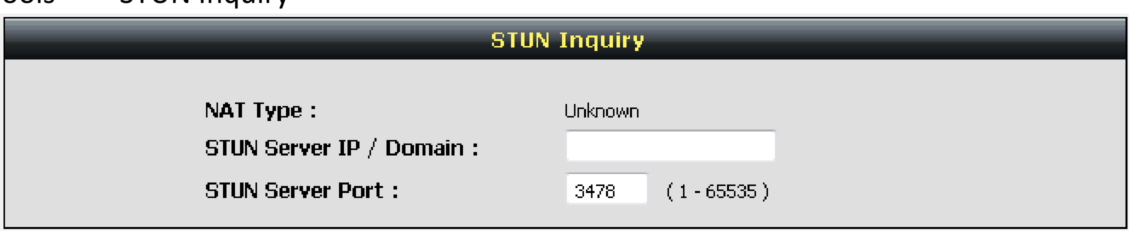
Tools → Ping Test



2-4-2 STUN Inquiry

Use “STUN Inquiry” to detect your IP sharing device’s NAT type and communication between a STUN server and client. STUN is applicable only when NAT Type is shown “Full cone” or “Open.”

Tools → STUN Inquiry



NAT Type: It shows the NAT type of your router.

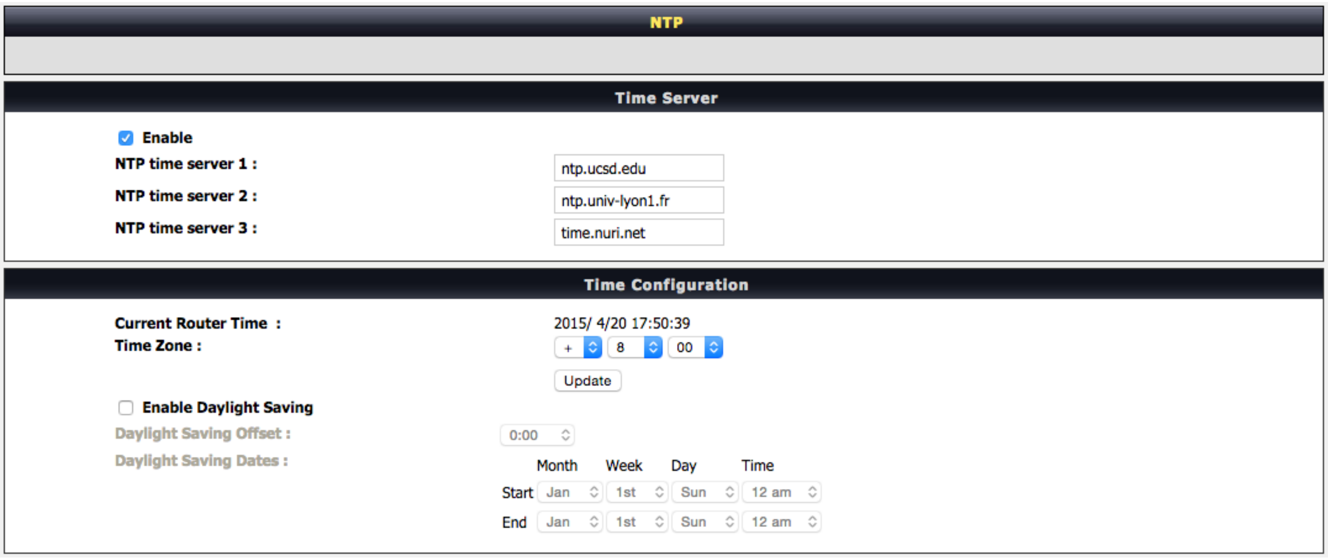
STUN Server IP/Domain: Enter the IP address or URL of the STUN server for query.

STUN Server Port: Enter the listening port of STUN Server.

2-5 System Settings

2-5-1 NTP

Enable to sync up with Internet time servers
System settings → NTP



Automatically synchronize with Internet time servers: The RoIP should automatically sync up with time servers.

First NTP time server: Select the desired domain name of a NTP server as first priority.

Second NTP time server: Select the domain name of a NTP server as second priority.

Current Router Time: It shows the current time of the RoIP.

Time Zone: Select your time zone from the drop-down menu.

Enable Daylight Saving: To enable/disable daylight saving time.

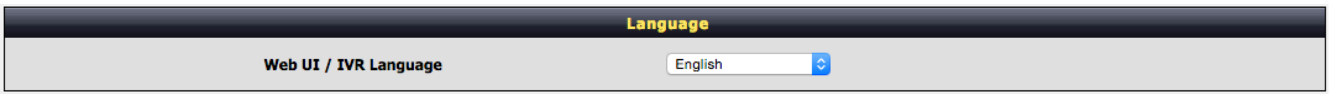
Daylight Saving Offset: Set the current time zone offset for your location.

Daylight Saving Dates: Set the start and end dates for daylight saving time.

2-5-2 Language

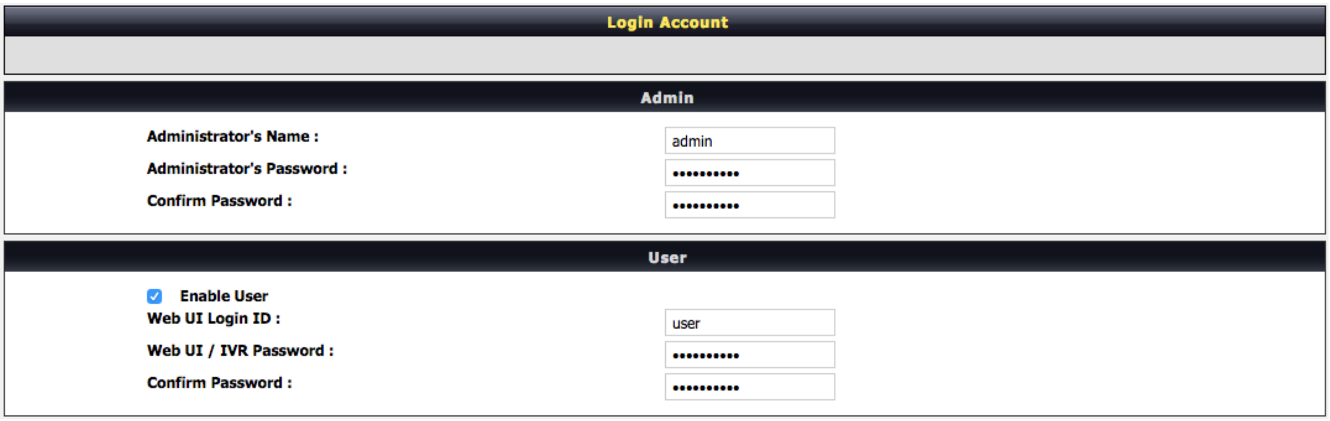
The system provides English, Traditional Chinese, and Simplified Chinese for displaying text on web pages. Changing the language setting also changes the language for IVR (Interactive Voice Response).

System settings → Language



2-5-3 Login Account

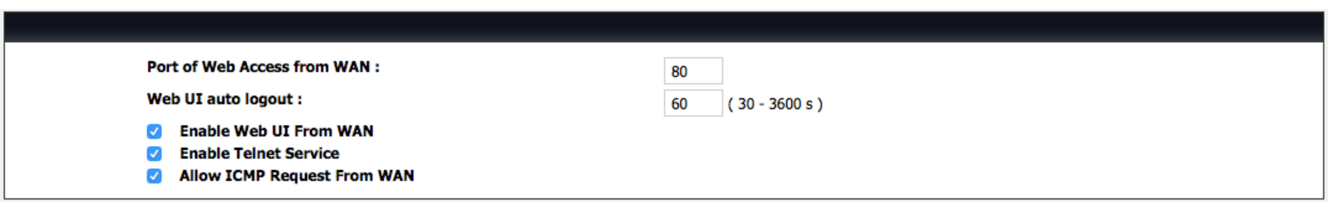
System settings → Login Account



Note: There are two operating levels when entering the Web UI. Logging-in as the ADMIN allows you to change all settings. A Web UI USER has access only to certain settings.

Password: Default password is all blank, please re-name Admin, and Web UI Login ID, also re-set the passwords.

System settings → Login Account



HTTP Server Port: Enter the port number for accessing web-based interface from the WAN port. This setting can only be changed through LAN Port access, not through WAN Port. The Port number of LAN Port access is 80. If you connect by bridging, and HTTP Port number is changed to 8080, even using LAN IP, you still have to use 8080 Port to access. "0" indicates you cannot access web page through WAN, but you can still access through LAN.

Web Idle Time Out: Enter the range of effective time when log-in the web interface. The user will be disconnected from the web page to allow others to log-in.

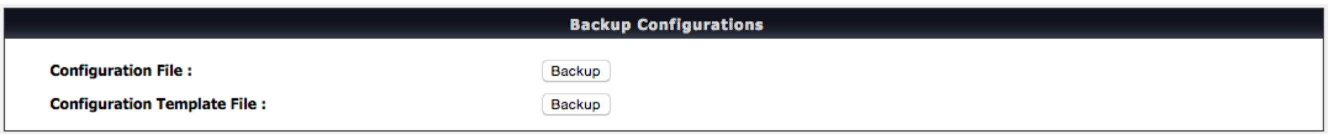
Enable Web UI: If disable this function, you cannot access web pages through WAN or LAN, please be careful for this setting.

Enable Telnet Service: Check the box to enable Telnet

2-5-4 Backup / Restore

Backup Configurations File

System settings → Backup / Restore



Configuration File: Click to back up all the configurations

Configuration Template File: Click to save your current settings to a template file for editing.

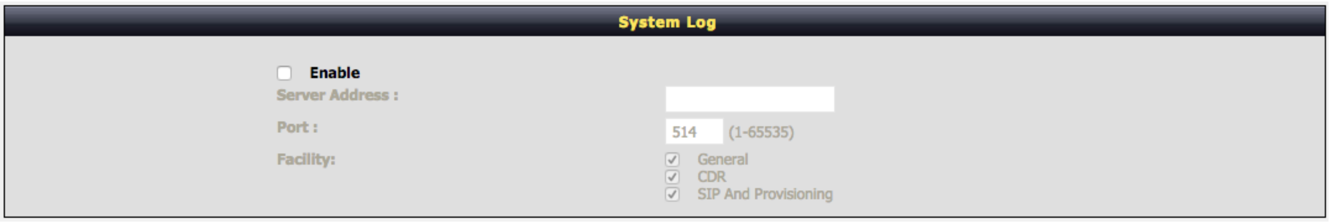
Restore Default Settings

System settings → Backup / Restore

After uploading Configuration Files or Restoring Default Configurations, they can be applied after clicking System settings → Save / Reboot →Reboot.

2-5-5 System Log

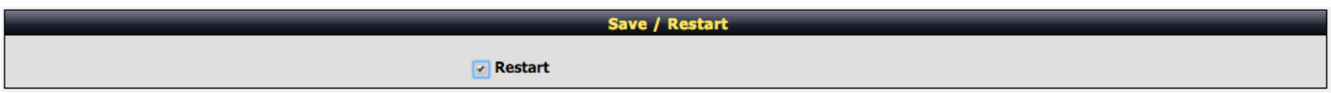
System settings → System log



Enable and fill in the Server Address and Port Number, the system will send the record to specified server.

2-5-6 Save/Reboot

System settings → Save/Reboot

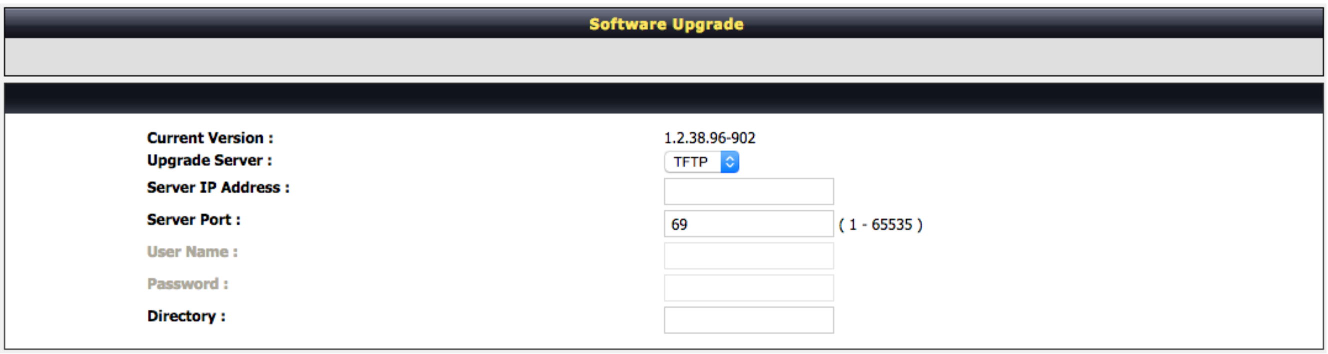


Reboot: Check "Reboot" and click "Yes" to reboot.

2-5-7 Software Upgrade

The Radio Gateway supports remote software upgrade. Please consult your VoIP Service Provider for information about the following details.

System settings → Software upgrade



Upgrade Server: Select the upgrade method: **TFTP**, **FTP**, or **HTTP**.

Server IP Address: Enter the server's IP address.

Server Port: Enter the server's port.

User Name/ Password: Enter the account information for accessing the server (if needed)

Directory: Enter the location of the firmware file.

2-5-8 Logout

Radio Gateway allows single log-in only. After saving the changed settings, please log out or reboot so the next user can log in and change the settings.

System settings → Logout