FXO-128o Analog VoIP Gateway

User Manual v1.0

Preface

Welcome

Thanks for choosing the **Gateway-128o Analog Gateway for VoIP**! We hope you will make full use of this rich-feature FXO VoIP Gateway.
Contact us if you need any technical support.

About This Manual

This manual provides information about the introduction of the analog VoIP gateway, and about how to install, configure or use it.
Please read this document carefully before install the gateway.

Intended Audience

This manual is aimed primarily at the following people:

- Users
- Engineers who install, configure and maintain the gateway.

Revision Record

Document Name	Document Version	Firmware Version
Gateway-128o FXO Analog VoIP Gateway User Manual V1.0	V1.0	

Contents

Product Introduction

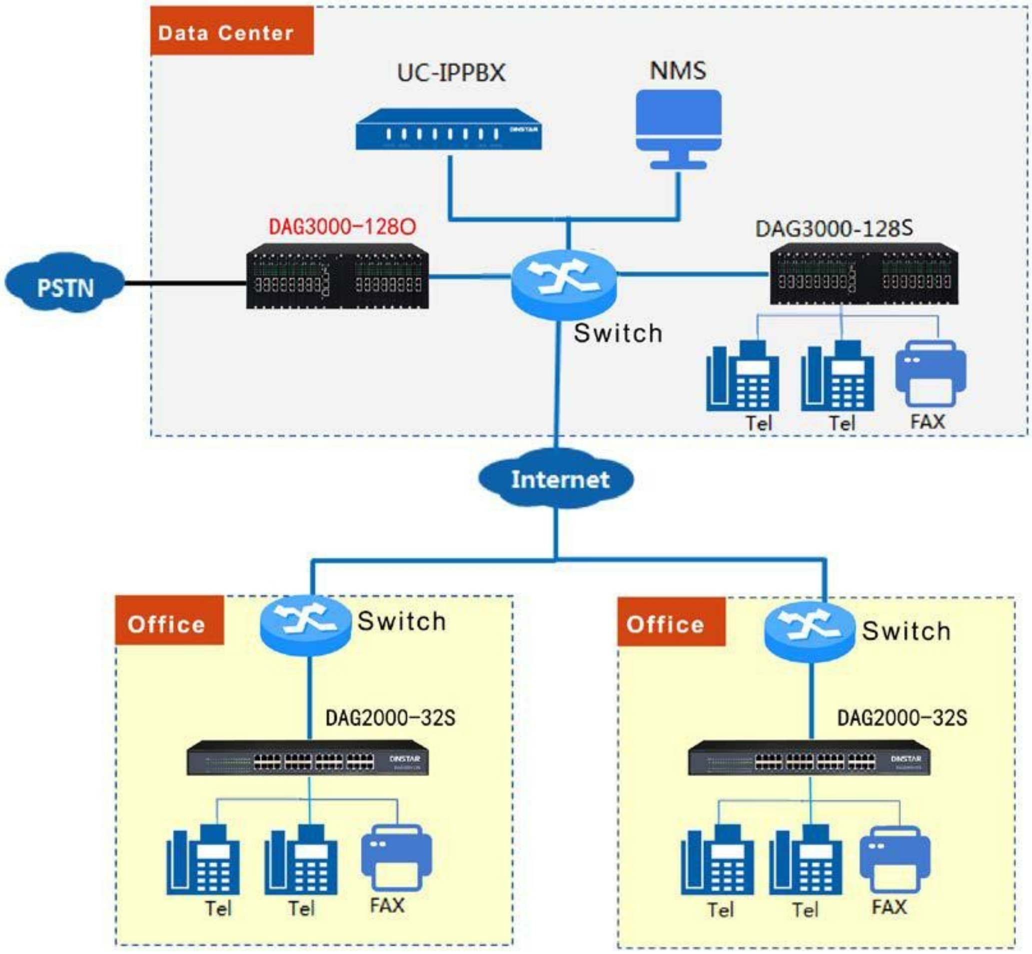
1.1 Overview

Gateway-128o is a multi-functional analog gateway offering seamless connectivity between IP-based telephony networks and legacy telephones (POTS), and PBX systems. It adopts modularized hardware design that allows to expand FXO ports by adding boards according to user's requirements. Each board has 8 FXO ports and the gateway supports 128 FXO ports at maximum.

Gateway-128o supports the standard SIP protocol and it's compatible with leading IMS/NGN platforms and SIP-based IP telephony systems. Which is ideally suited for small and medium businesses, call centers and multi-location environments that need VoIP services.

1.2 Application Scenario

The application scenario of Gateway-128o is shown as follows:



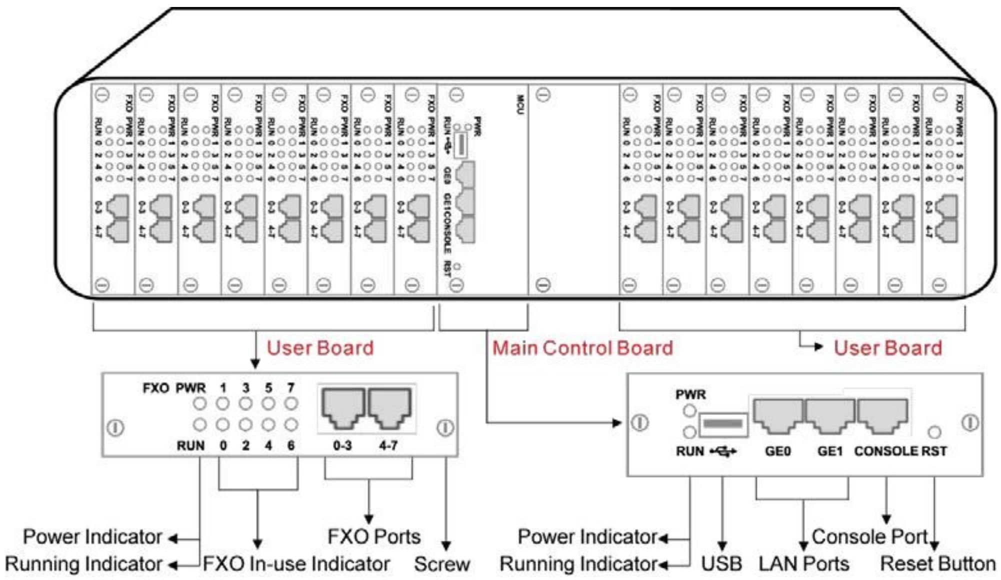
1.3 Product Appearance

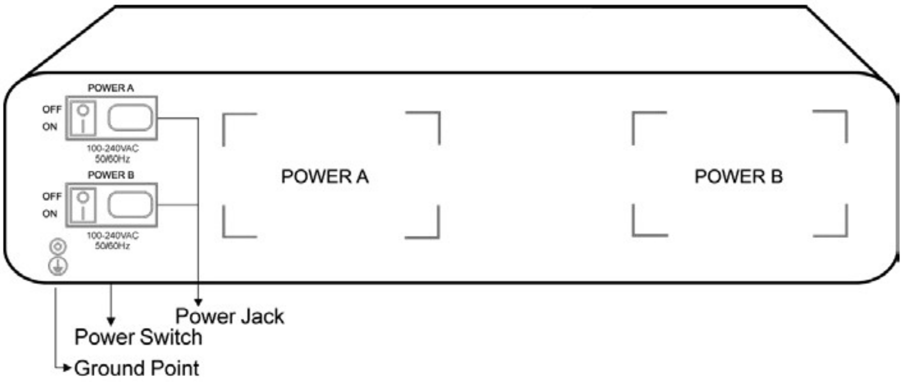
1.3.1 Appearance of Gateway-128o

Front View:

Back View:

1.3.2 Ports and Connector





The description of interfaces of Gateway-128o

Port Name	Connector	Description
Power Jack	Power Jack	To connect 100-240V AC 50-60HZ power supply
Power Switch	Power Switch	Turn on or turn off power supply
LAN Port (GE0/GE1)	RJ45	To connect to the IP network over a DSL modem or Router or a LAN switch
FXO Ports 0~7	RJ45	To connect to PSTN
Console Port	RJ48	Console port is used to carry out maintenance-related configurations
Reset Button	Reset Button	After the device is powered on, short press the RST button to restart the device; long press the RST button for about 6-10 seconds to restore the device to factory default settings.

The description of indicators of Gateway-128o

Indicator	Definition	Status	Description
PWR	Power Indicator	On	Power supply is normal .
		Off	There is no power supply or power supply is abnormal.
RUN	Running Indicator	Slow Flashing	The device is running properly

		Fast Flashing	SIP account is registered successfully.
		Off	The device is running improperly
FXO 0~7	Telephone In-use Indicator	On	FXO port is currently occupied by
		Off	FXO port is idle or faulty
LAN Port (GE0/GE1)	Link (Green)	Flashing	The gateway is properly connected to network
		Off	The gateway is not connected to network or network connection is improper

1.4 Features & Functions

Key Features

- Modularized Gateway with 128 FXO ports
- Fax over IP (T.38 and Pass-Through)
- Support IPv4 and IPv6
- TR069 and SNMP
- Multiple codecs: G.711A/U,G.723.1,G.729A/B etc.
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

Physical Interfaces

- Capacity

  Range from 8 to 128 FXO
- Capacity

  Range from 8 to 128 FXO

  Support 16 user board

  slots
- User Board

  2* RJ45 connectors with 8 FXO
- MCU Board 1*RS232,

  115200bps

  2*10/100/1000Mbps,

  RJ45 1* USB 2.0

Voice Capabilities & Fax

- G.711A/U law, G.723.1, G.729A/B, AMR, G.726, iLBC
- Silence Suppression
- Comfort Noise Generation(CNG)
- Voice Activity Detection(VAD)
- Echo Cancellation(G.168), with up to 128ms
- Adaptive (Dynamic) Jitter Buffer
- Programmable Gain Control
- T.38/Pass-through
- Modem/POS
- DTMF mode: Signal/RFC2833/INBAND
- VLAN 802.1P/802.1Q
- Layer3 QoS and DiffServ

FXO

- Connector: RJ45 with 4 FXO
- Dial Mode: DTMF/Pulse Dialing
- Caller ID: FSK, DTMF
- Polarity Reversal
- Answer Delay
- Busy Tone Detection
- No Current Detection

VoIP

- Protocol: SIP v2.0 (UDP/TCP),RFC3261,SDP,RTP(RFC2833), RFC3262,3263,3264,3265, 3515,2976,3311
- SIP TLS/SRTP
- RTP/RTCP, RFC2198, 1889
- RFC4028 Session Timer
- RFC3266 IPv6 in SDP
- RFC2806 TEL URI
- RFC3581 NAT,rport
- Outbound Proxy
- DNS SRV/ A Query/NATPR Query
- SIP Trunk
- Early Media/Early Answer

- NAT:STUN, Static/Dynamic NAT

Software Features

- Port Group
- Web ACL
- Telnet ACL
- Action URL
- Digitmap
- Routing Rules based Prefixes
- Caller/Called Number Manipulation

Network

- Static/Dynamic IP,
- PPPoE
- DHCP Client
- IPv4/IPv6
- TCP, UDP,TFTP, FTP, ARP,RARP, Ping, NTP, SNTP, HTTP/HTTPS, DNS
- Ping / Tracert
- DHCP Option 66,120,121

Environmental

- 1+1 Power Supply: 100-240VAC, 50-60 Hz
- Power Consumption:60W(Typical)

- Storage Temperature: -20 ℃ ~80 ℃
- Humidity:10%-90% Non-Condensing
- Dimensions(W/D/H): 440*300*135mm
- Unit Weight: 9.3kg

Maintenance

- SNMP v1/v2/v3
- TR069
- Auto Provisioning
- Web/Telnet
- Configuration Backup/Restore
- Firmware Upgrade via Web
- CDR

- Syslog
- Network Capture
- NTP/Daylight Saving Time
- IVR local Maintenance
- Cloud-based Management

2 Quick Installation

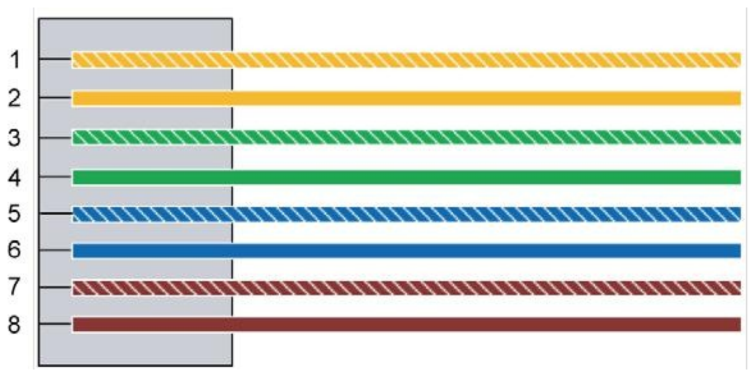
2.1 Installation Attentions

To avoid unexpected accident or device damage, please read the following instructions before installing the FXO GW device:

- Gateway-128o is equipped with RJ45 ports;
- Anti-jamming: to reduce the interference with telephone calls, it's highly recommended that telephone lines connected to the gateway should be placed away from power cables;
- Power supply: the gateway accepts AC input voltage of 100-240V. Please ensure safe and stable power supply;
- Network bandwidth: please ensure there is enough network bandwidth so as to guarantee stabilized running of the gateway;
- Ventilation: to avoid overheating, please do not pile up the gateway with other devices and make sure the gateway has good ventilation around.
- Temperature and humidity: to avoid any accident that might cause malfunction, it's advised to install the gateway in an equipment room where temperature and humidity are appropriate;
- Mechanical load: please make sure the gateway is placed steadily to avoid damage. It' s highly advised to horizontally place the gateway on a flat surface or a cabinet.

2.2 RJ45 Wire Sequence

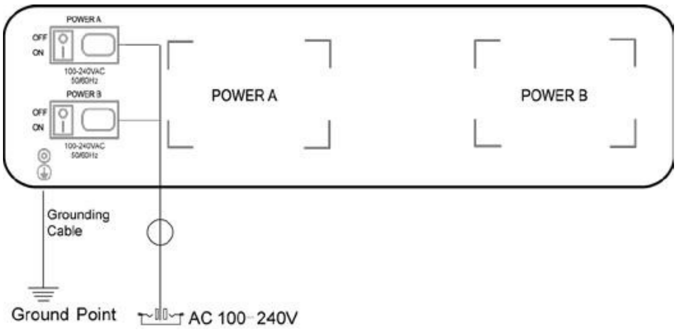
Gateway-128o is equipped with RJ45 interface as FXO port. The internal wire sequence of RJ45 cable is shown as follows:



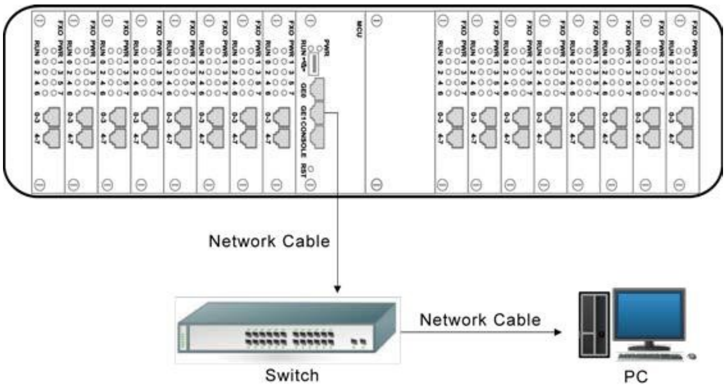
Pair 1	Orange & White
	Orange
Pair 2	Green & White
	Green
Pair 3	Blue& White
	Blue
Pair 4	Brown & White
	Brown

2.3 **Installation Steps**

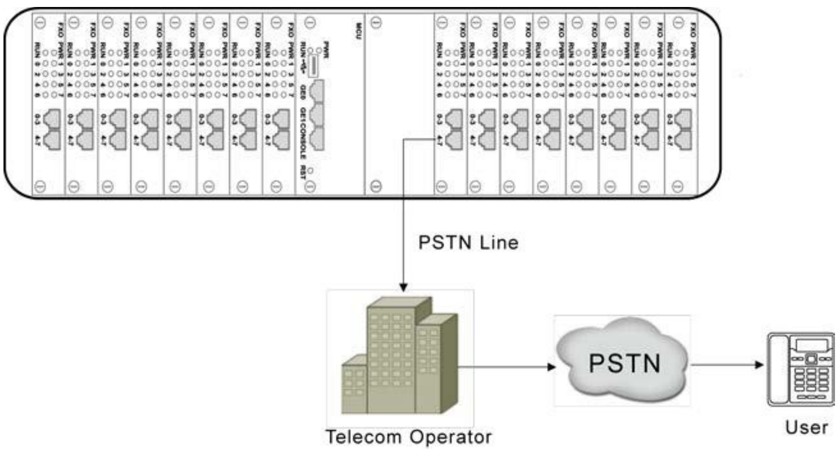
- Connect the power adapter to the power jack of the FXO GW device, connect one end of the ground cable to the grounding port on the back of FXO GW , and then connect the other end to the grounding bar of the shelf.



- Connect network cable to the GE0/GE1 port;



- Connect PSTN line to the FXO port(s).



3 Basic Operation

3.1 Call out and Call in via FXO Port

3.1.1 Call Out

- One-stage Dialing: After the gateway receives a call number sent from softswitch or IPPBX, if the number matches one of the dialing rules set on Advanced → Digit Map interface, the call will directly choose a FXO port to call out based on port selection rule.
- Two-stage Dialing: dial a FXO port's SIP account number from an extension of IPPBX, and then you will hear a dialing tone. After that, you will be able to dial any number of PSTN.

3.1.2 Call In

Dial the number of PSTN connected to a FXO port of the gateway, and then you will hear a dialing tone or a voice prompt of "please dial the extension number". Then dial the called number (extension number or telephone number), after the dialing is completed, the called number will be sent to IP server such softswitch or IPPBX.

Hotline auto-dialing: Dial the number of PSTN connected to a FXO port of the gateway, then the gateway will automatically route the call to designated extension number or telephone number according to preset hotline number.

3.2 Description of Feature Codes

Gateway-128o provides convenient telephone functions. Connect a telephone to the FXO port and dial a specific feature code, and you can query corresponding information.

***158#**	Dial *158# to query LAN IP
***114#**	Dial *114# to query the phone number of a FXO port
157	Dial *157*0 to set route mode; dial *157*1 to set bride mode
150	Dial *150*1 to set IP address as static IP address; dial *150*2 to set IP address as DHCP IP address
152	Dial *152* to set IPv4 address, for example: Dial *152*192*168*1*10# to set IPv4 address as 192.168.1.10

156	Dial *156* to set IPv4 gateway, for example: Dial *156*192*168*1*1# to set IPv4 gateway as 192.168.1.1
153	Dial *153* to set IPv4 netmask, for example: Dial *153*255*255*0*0*# to set IPv4 netmask as 255.255.0.0
160	Dial *160*1# to allow HTTP WAN access, Dial *160*0# to deny HTTP WAN access
***111#**	Dial *111# to restart the device
47	Dial *47* to allow call through IP address, for example: Dial *47*192*168*1*1# to allow to call through the IP address of 192.168.1.1
***51#**	Dial *51# to enable the call waiting service
***50#**	Dial *50# to disable the call waiting service
87	Dial *87* to trigger blind transfer, for example: Dial *87*8000#, and you can blind transfer to the extension number 8000
72	Enable unconditional call forwarding service. Example: Dial *72*8000, and calls will be unconditionally forwarded to extension number 8000
***73#**	Disable unconditional call forwarding service
90	Enable the ‘call forwarding on busy’ service. Example: Dial *90*8000, and calls will be forwarded to extension number 8000 when the called number is busy
***91#**	Disable the ‘call forwarding on busy’ service
92	Enable the ‘call forwarding on no reply’ service. Example: Dial *92*8000, and calls will be forwarded to extension number 8000 when there is no reply from the called number
***93#**	Disable the ‘call forwarding on no reply’ service
***78#**	Enable the ‘No Disturbing’ service
***79#**	Disable the ‘No Disturbing’ service
***200#**	Dial *200# to access voicemail
Flash/Hook	Switch between incoming calls, If not in session, flash/hook will switch a new channel for new call.

Note: Some of the above feature codes will be realized on the available platform.

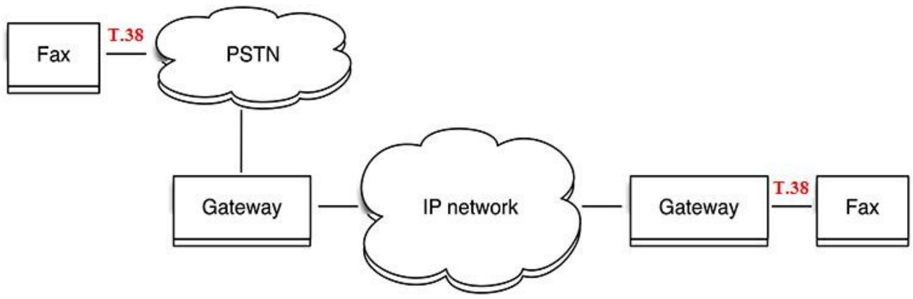
3.3 Send or Receive Fax

3.3.1 Fax Mode Supported

- T.38（IP-based）
- Pass-Through（VBD)
- Adaptive Fax Mode (automatically match with the peer fax mode)

3.3.2 Explanation of T.38 and Pass-through

T.38 is an ITU recommendation for allowing transmission of fax over IP networks in real time. Under the T.38 mode, analog fax signal is converted into digital signal and fax signal tone is restored according to the signal of peer device. Under the T.38 mode, fax traffic is carried in T.38 packages.



Pass-through: Under the pass-through mode, fax signal is not converted and fax traffic is carried in RTP packets. It uses the G.711 A or G711U codec in order to reduce the damage to fax signal.

Adaptive Fax Mode: automatically match with the fax mode of the peer device.

3.4 Local IVR Operation

3.4.1 Inquiring the IP address

Connect a PSTN line to one of the FXO ports of the gateway, and then use a mobile phone or a fixed telephone to dial the number of the PSTN line. After you hear a dialing tone or a voice prompt, dial *158# to inquire the IP address of the gateway.

3.4.2 Factory Reset

Connect a PSTN line to one of the FXO ports of the gateway, and then use a mobile phone or a fixed telephone to dial the number of the PSTN line. After hearing a dialing tone or a voice prompt, dial *166*000000#, and you will hear “successful setting”, then hang up the phone and the gateway is reset to factory defaults.

3.4.3 IP Address configuration

Before configuration, please ensure:

- The gateway is power on;
- Device has been connected to network;
- The PSTN line has been connected to the FXO port of the gateway.

Configure dynamic IP address by DHCP:

1) Pick up the phone
2) dial *150*2# and then hang up the phone
3) If the voice prompt indicates ‘setting successfully’, please restart the gateway after 10 seconds.

Configure Static IP address:

Take the configuration of IP address ‘172.16.0.100’ as example.

1) Pick up the phone,
2) dial *150*1# and then hang up the phone

Then configure IP address and subnet mask as follow:

- Configure IP address

1) Pick up the phone
2) dial *152*172*16*0*100# and then hang up the phone.

- Configure subnet mask

1) Pick up the phone
2) dial *153*255*255*0*0# and then hang up the phone.

- Configure gateway IP address

1) Pick up the phone
2) dial *156*172*16*0*1# and then hang up the phone.

- Query the IP address of the gateway:

1) Pick up the phone,
2) dial *158#.

If the gateway uses PPPoE method to get IP address，the IP address needs to be configures through web browser.

Note: The telephone will play voice prompt “setting successfully” if the step is correct.

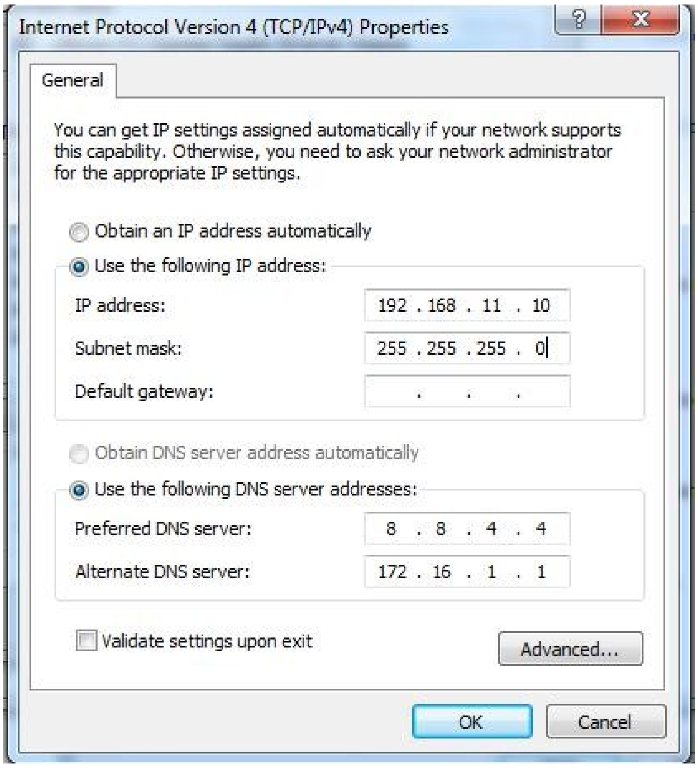
4 Configurations on Web Interface

4.1 Preparations for Login

4.1.1 Network Connection

Firstly, connect the device to the network and connect a PSTN line to the FXO port(See section 2.3 for details). Then dial *158# to query the IP address of the gateway.

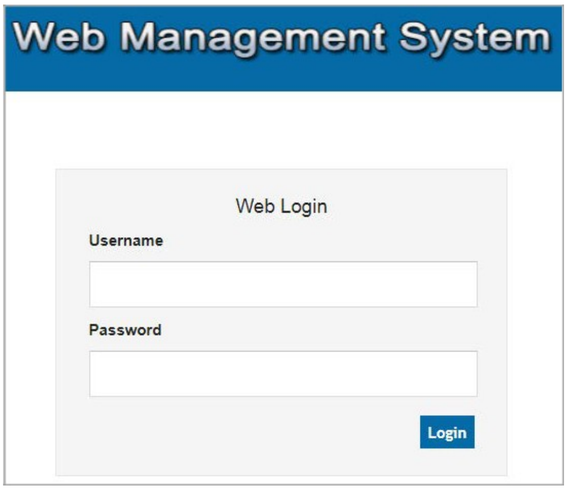
The default IP address of the device is 192.168.11.1. It is recommended to modify the IP address of the local computer to ensure that it is in the same network segment with the device. Take windows 7 as an example, change the IP address of the local computer to 192.168.11.10:



Secondly, check the connectivity between the PC and the device. Click **Start** → **Run** of PC and enter **cmd** to execute 'ping 192.168.11.1' to check whether the IP address of the gateway runs normally.

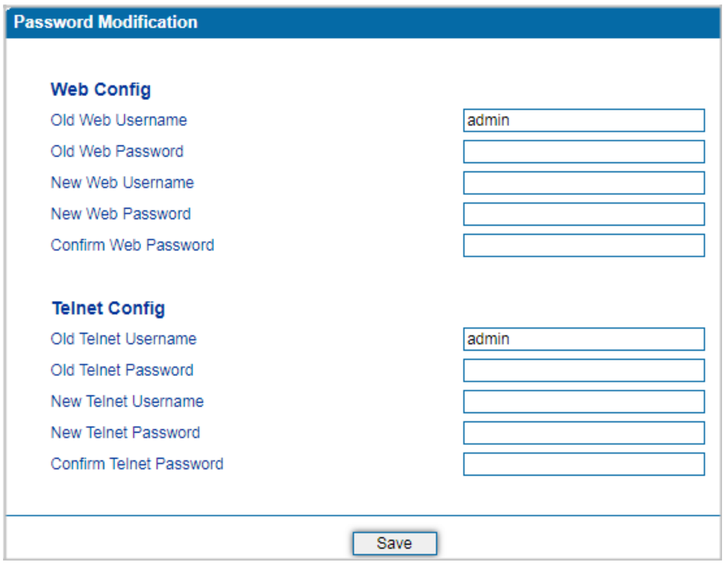
4.1.2 Log In Web Interface

Open a web browser and enter the IP address of the gateway (the default IP is 192.168.11.1). Then the login GUI will be displayed.



Enter default username and password: **admin/admin**, then click **Log in** to enter into the Web interface. And then you can see the following Web interface.

It is suggested that you should modify the username and password for security consideration on the **Security -> Password Modification** interface.

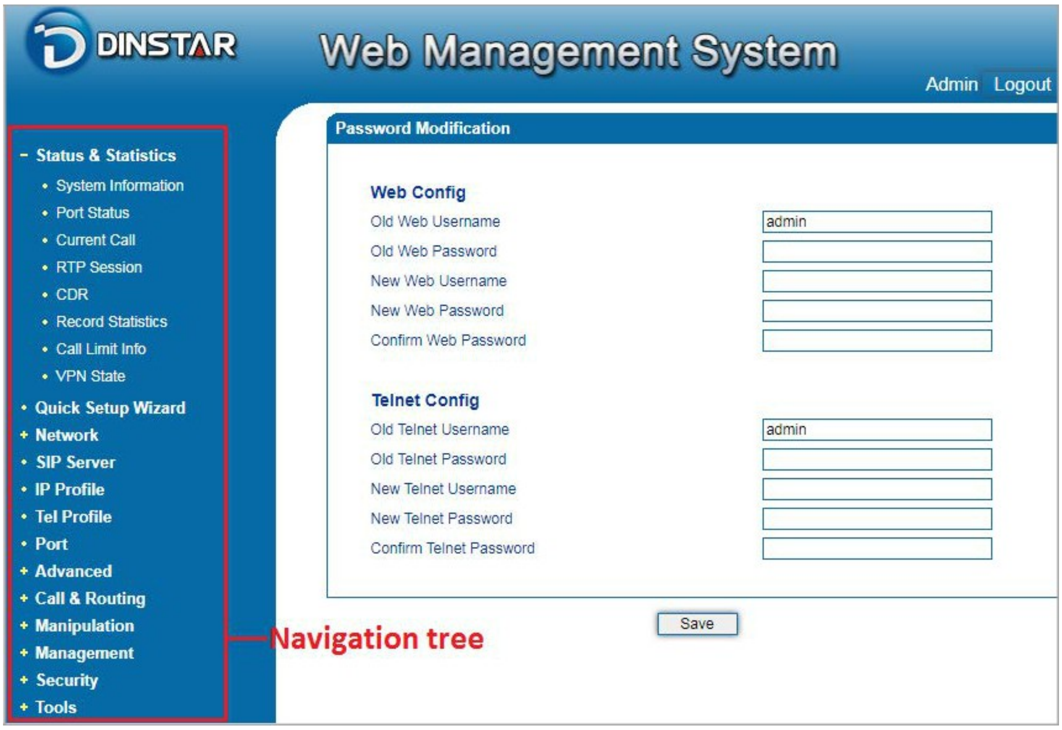


4.2 Navigation Tree

The web management system of the Gateway-128o VoIP device consists of the navigation tree and detailed configuration interfaces.

Choose a node of the navigation tree to enter into a detailed configuration interface.

Figure 4-1 Navigation Tree of Web Interface



4.3 Status & Statistics

The 'Status & Statistics' menu mainly displays all kinds of information. It includes the following sub-menus: System Information, Port Status, Current Call, RTP Session, CDR, Record Statistics and VPN State .

4.3.1 System Information

Log in the Web interface, and then click **Status & Statistics** → **System Information**, and the following page will be displayed. On the page, you can view the information of device ID, MAC address, network mode, IP addresses, version information, server register status and so on.

Figure 4-3-1 System Information

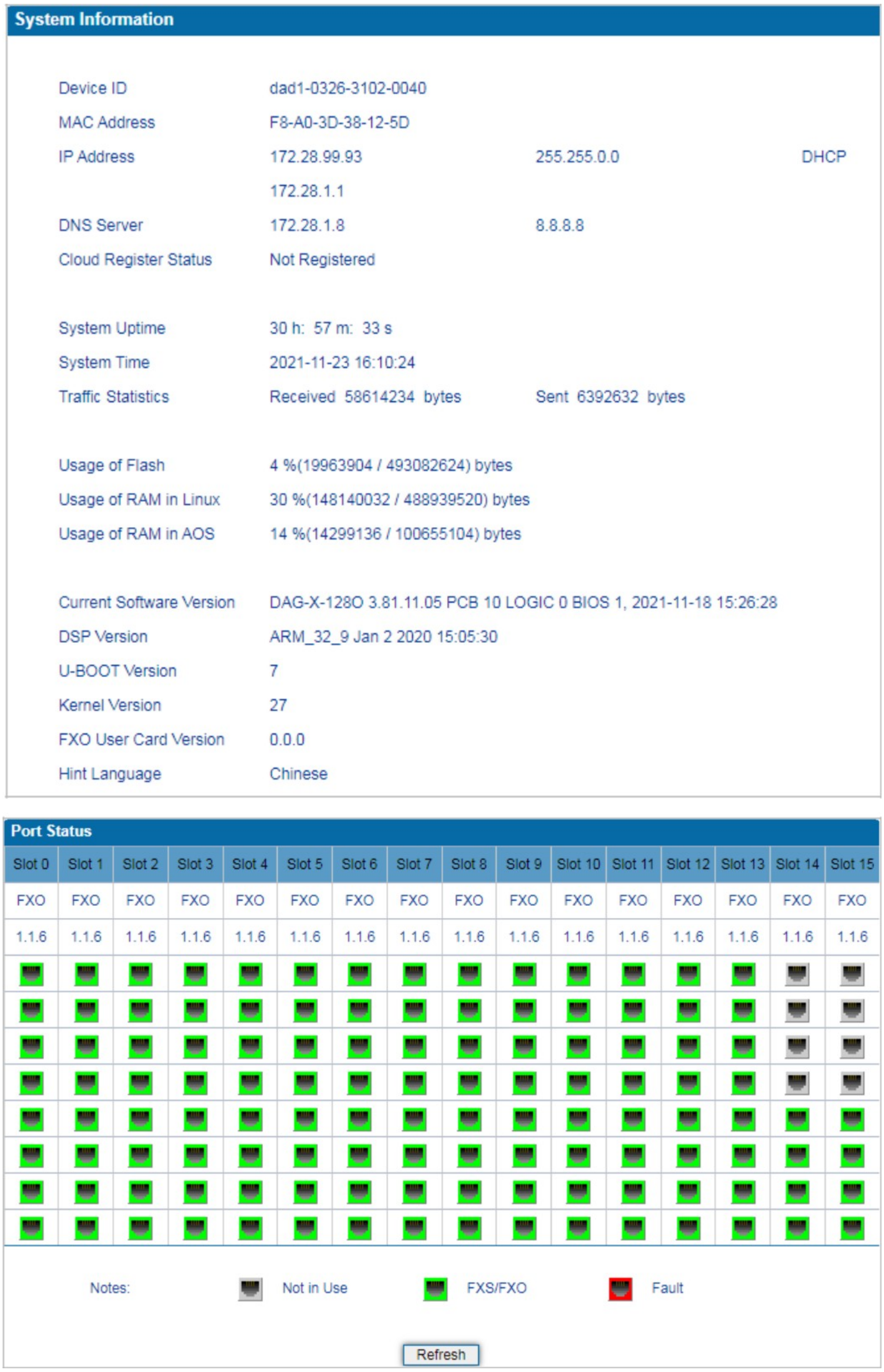


Table 4.3.1 Explanation of Items on System Information Interface:

Parameter	Explanation
Device ID	A unique ID of each device. This ID is used for warranty and cloud server authentication.
MAC address	Hardware address of the gateway
IP Address	There are three kinds of IP address: **DHCP: Obtain IP address automatically**. FXO GW is regarded as a DHCP client, which sends a broadcast request and looks for a DHCP server from the LAN to answer. Then the first discovered DHCP server automatically assigns an IP address to the FXO GW from a defined range of numbers. **Static IP Address:** Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a **dynamic IP address,** which is assigned *ad hoc* at the start of each session, normally changing from one session to the next. If you choose static IP address, you need to fill in the following information: ● IP Address: the IP address of the FXO GW; ● Subnet Mask: the netmask of the router connected the FXO GW; ● Default Gateway: the IP address of the router connected the FXO GW; **PPPoE:** PPPoE is an acronym for point-to-point protocol over Ethernet, which relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting the users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device or cable modem. PPPOE IP address refers to IP address assigned through the PPPoE mode. If you choose PPPoE, you need to fill in to fill in the following information: ● Username: the account name of PPPoE ● Password: the password of PPPoE ● Server Name: the name of the server where PPPoE is placed
DNS Server	IP addresses of primary DNS server and standby DNS server are displayed.
Cloud Register Status	Whether the FXO GW device is registered to cloud or not.
System Uptime	The running time of the FXO GW device since it is powered on.
NTP Status	Succeed: the FXO GW device is sync to NTP server successfully; Failed: the FXO GW device fails to be sync to NTP server. Then you

	should check network connection and the NTP server.
Traffic Statistics	Total bytes of message received and sent by the device.
Usage of Flash	Detailed usage of Flash memory
Usage of RAM in Linux	detailed RAM usage of Linux core
Usage of RAM in AOS	Detailed RAM usage of AOS
Current Software Version	The software version that runs on the FXO GW device. Model name, version number and the software development date are displayed.
Backup Software Version	Backup software is for the purpose of backup. When the current software fails, the backup software version will work.
U-boot Version	U-boot version
Kennel version	Linux Kennel version
FS Version	File system version
Hint Language	The current language of the FXO GW device

4.3.2 Port Status

The following figure shows the registration information of ports and port groups. Users can view the registration status of each port and port group of the gateway device through this page.

The description of ports and port groups is as follows:

Port : An FXO port

Port group: It is composed of several ports. In some cases, multiple ports can be registered with the same account and can be used for incoming and outgoing calls using the same phone number.

Port

Port No.	Type	SIP User ID	User Status	Port Status	Call Status
0	FXO	---	---	OnHook	Idle
1	FXO	---	---	OnHook	Idle
2	FXO	---	---	OnHook	Idle
3	FXO	---	---	OnHook	Idle
4	FXO	---	---	OnHook	Idle
5	FXO	---	---	OnHook	Idle
6	FXO	---	---	OnHook	Idle
7	FXO	---	---	OnHook	Idle

Select Slot: Slot 0

Port Group

Group	Port	SIP User ID	User Status
---	---	---	---

Refresh

4.3.3 Current Call

Call statistics for each port of the device, including: port number, port type, source, destination, connection time, and duration.

Current Call

Port	Type	Source	Destination	Connected Time	Duration(s)
---	---	---	---	---	---

Select Slot: Select Al

Refresh

4.3.4 RTP Session

On the **Status & Statistics** → **RTP Session** page, you can view the real-time RTP session information, including: port, payload type, packet period, local port, peer IP, peer port, sent packets, received packets, lost packets, jitter and duration.

Figure 4-3-4 Real-time RTP Session Information

RTP Session

Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Sent Packets	Recv Packets	Lost Packets	Jitter	Duration(s)
2	T.38	20	8008	172.16.95.50	8000	487	273	0	0	27

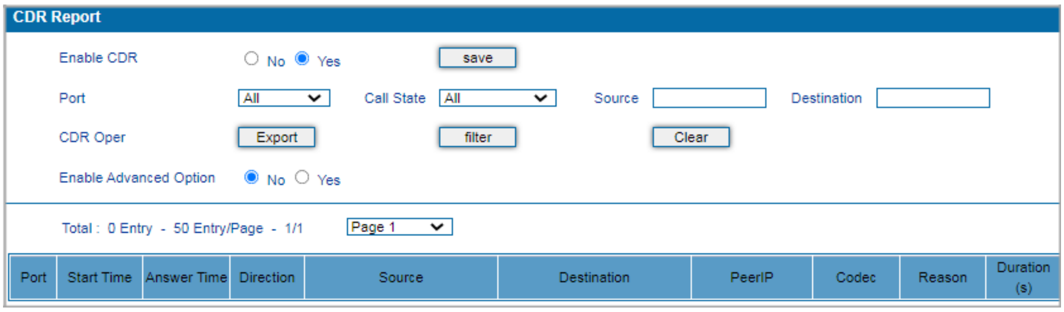
Refresh

4.3.5 CDR

CDR (Call Detail Record): is a data record produced by a telephone exchange or a telecommunication device, which contains the details of a telephone call that passes through the device.

On the **Status & Statistic → CDR** page, you can enable the CDR function and view the details of all calls through the FXO ports of the FXO GW device. You can also export, filter or clear the CDRs. 5000 pieces of CDRs can be saved at most.

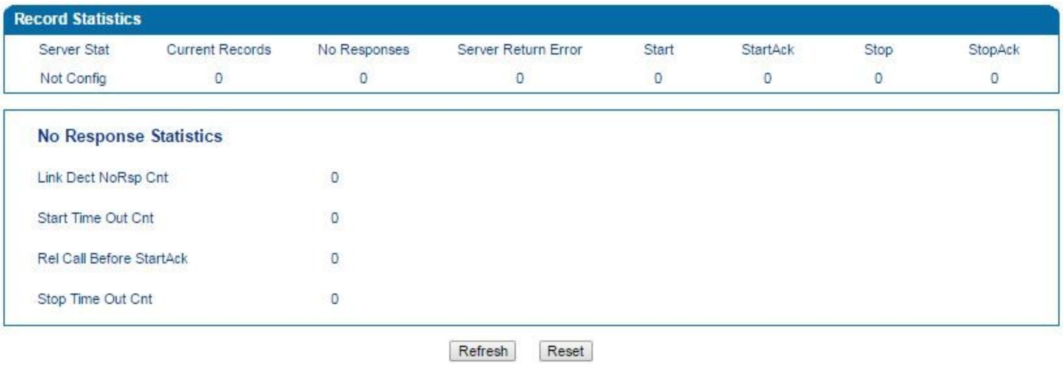
Figure 4-3-5 CDRs of FXO Ports



4.3.6 Record Statistics

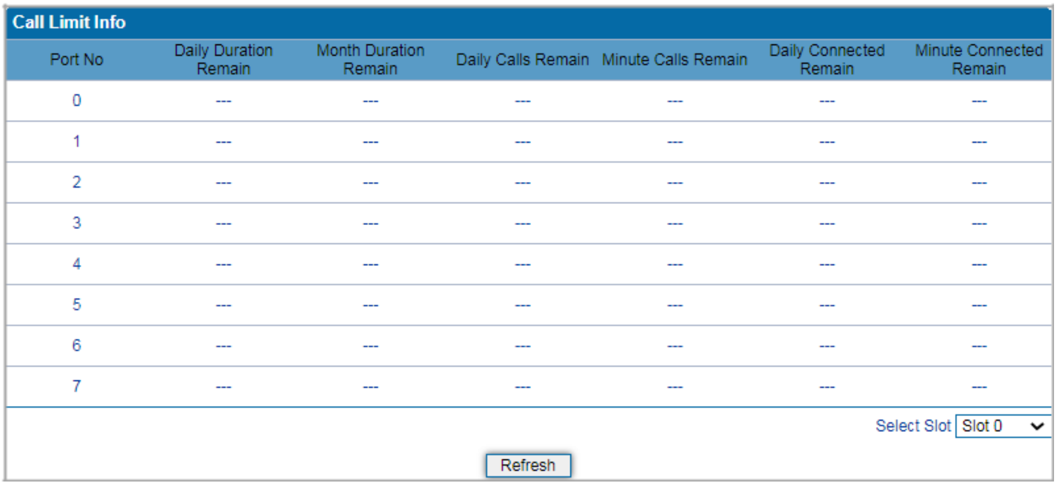
On the **Status & Statistic → Record Statistics** page, record statistics including server status, count of current records, count of no response, count of server return errors, count of record starts, count of record startAck, count of record stops and count of stopAck are displayed.

Figure 4-3-6 Record Statistics



4.3.7 Call Limit Info

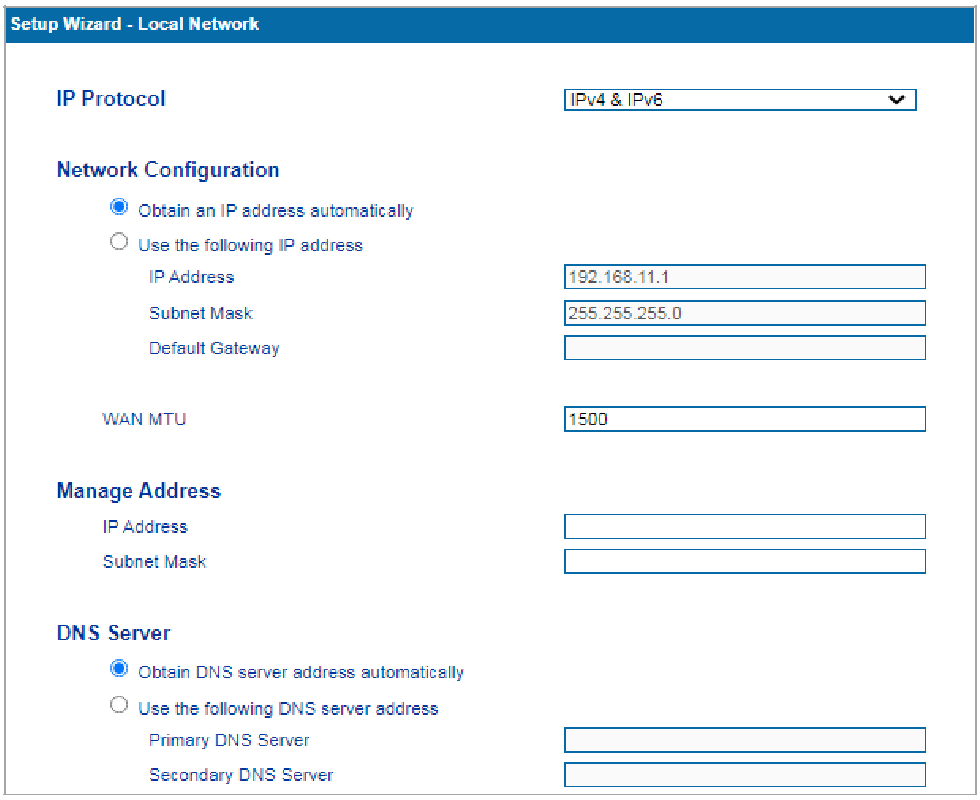
If you configure call limit on the "Call & Routing -> Call Limit" for the port, you can check the remaining call duration and number of calls of the configured port.



4.4 Quick Setup Wizard

Quick setup wizard guides user to configure the device step by step. User only needs to configure network, SIP server and SIP port in the Quick Setup Wizard interface. Basically, after these three steps, user is able to make voice call via the FXO GW device.

For the configurations of network, SIP server and SIP port, please refer to 4.5 , 4.5 and 4.9 .



4.5 Network

4.5.1 Local Network

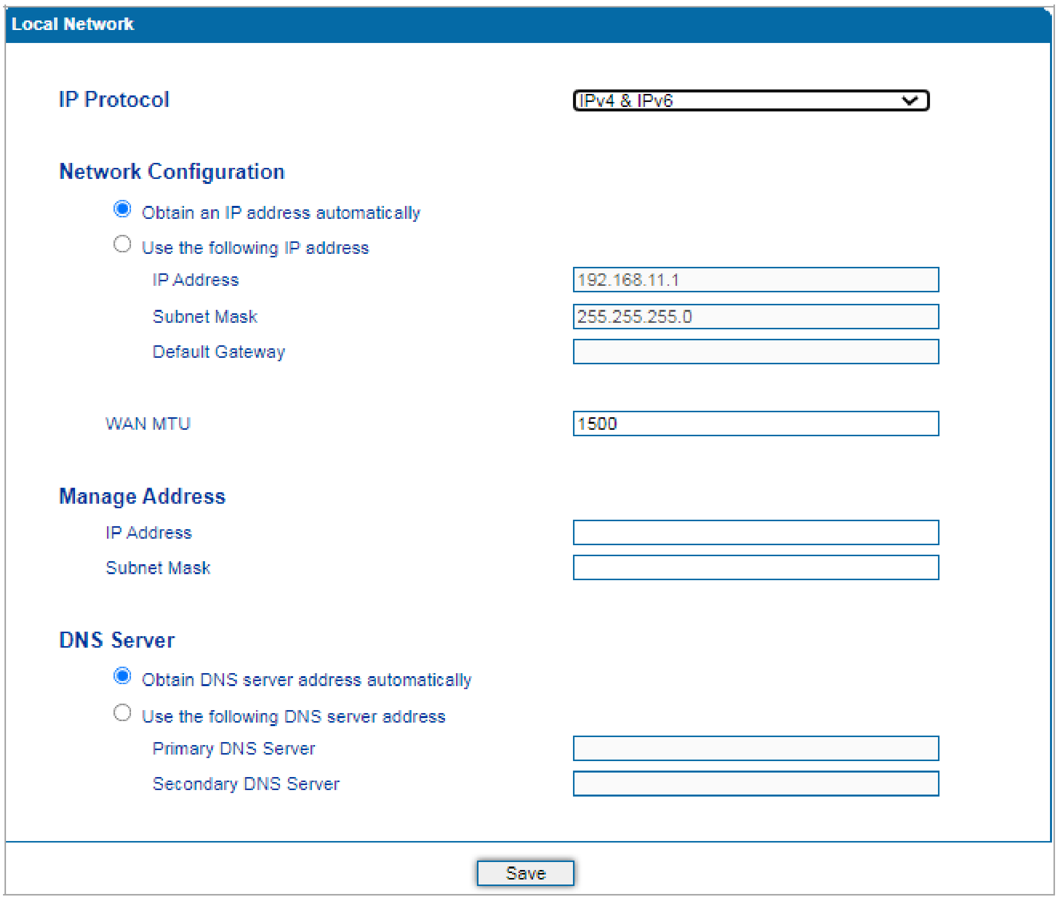
The user can configure the IP protocol, IP address obtain method, management address and DNS server of the FXO GW device on the "Network Local Network" page.

FXO GW analog gateway supports IPv4 and IPv4&IPv6 two IP protocols and supports three IP address obtain methods (that is, automatically obtain through a DHCP server, setting a static IP address, and through PPPoE).

Note:

a. When it is configured to "obtain an IP address automatically", it is necessary to ensure that there is a DHCP Server in the network and it is working normally.
b. The management IP address and the network IP address cannot be in the same network segment.
c. After the configuration is complete, you need to restart the device to make the configuration take effect.

Figure 4-5-1 Network Setting



4.5.2 VLAN（Virtual Local Area Network）

In order to control the impacts brought by broadcast storms, you can divide the local-area network into three VLAN groups, including data VLAN, voice VLAN and management VLAN on the **Network** → **VLAN** page.

Management VLAN transmits management-related packets, such as packets of SNMP, TR069, Web and Telnet, while voice VLAN transmits the VoIP signals and voices produced by the device itself. Data VLAN transmits data packets.

Figure 4-5-2 Configure VLAN

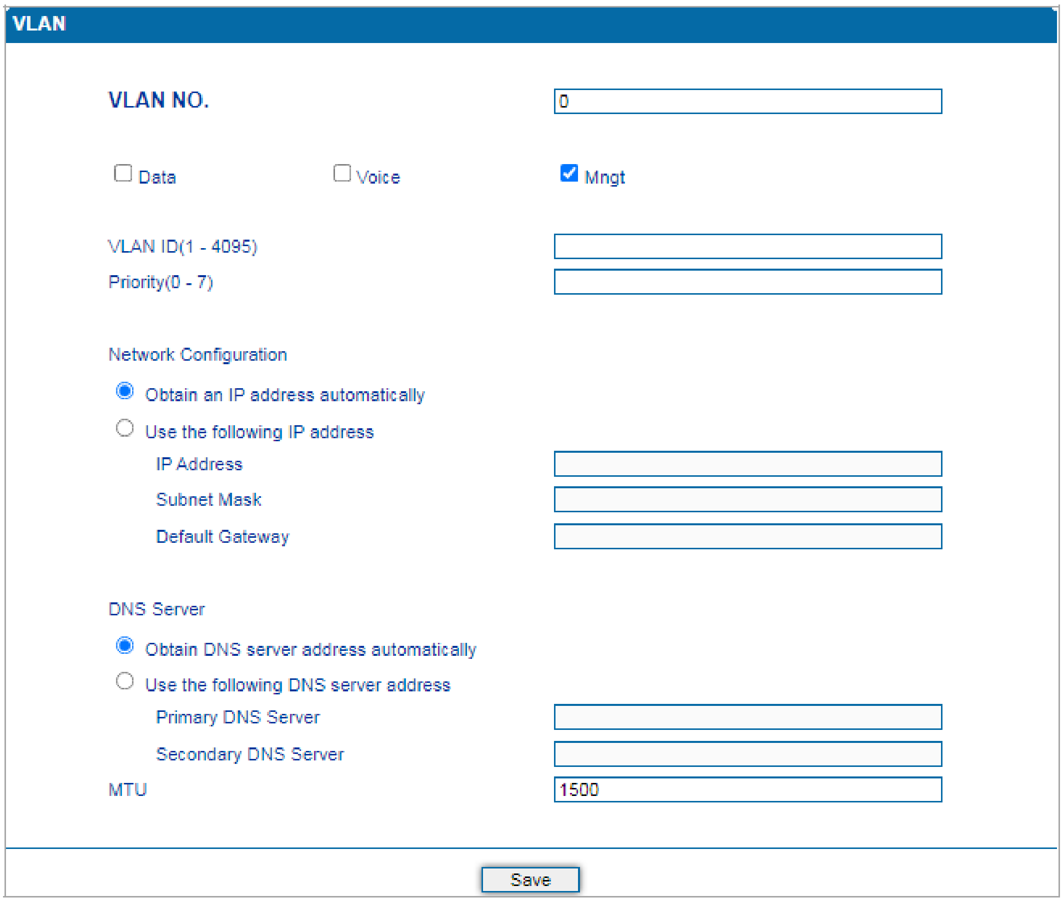


Table 4.5.2 Explanation of VLAN Parameters

Parameter	Explanation
Data/Voice/Management	Select what kind of messages are allowed to go through this VLAN. For example, if the checkbox on the left of data is selected, it means data messages are subject to the following network setting of this VLAN.
VLAN ID(0-4095)	Set an ID to identify a VLAN based on 802.1Q protocol. Range is from 0 to 4095.

Priority（0-7）	Set the priority of a VLAN based on 802.1P protocol. 0 is the highest priority.
Network Setting	Set a DHCP IP address or static IP address for a VLAN, and set the IP address of the DNS server used by the VLAN.

Note: After the configurations are finished, you need to restart the device for the configurations to take effect.

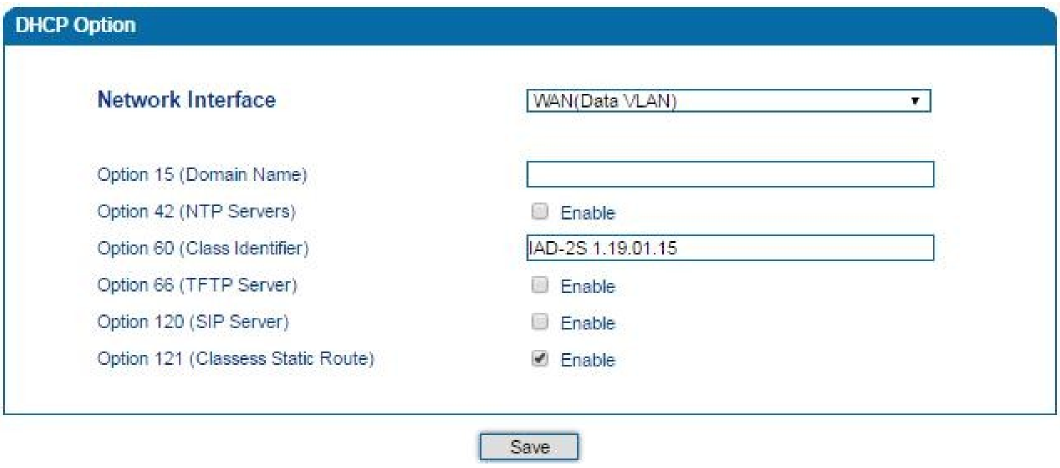
4.5.3 DHCP Option

When the FXO GW device works as a DHCP client and applies for an IP address, DHCP server will return packets which include an IP address as well as configuration information of enabled option fields.

The following is the meaning of the option fields involved in FXO GW (that means the following option fields are enabled, DHCP server will return information of corresponding option fields:

- Option 15: to set a DNS suffix;
- Option 42: to specify NTP server;
- Option 60: to define VCI (vendor class identifier) of FXO GW on the DHCP server;
- Option 66: to specify TFTP server which will assign software version to FXO GW;
- Option 120: to fetch SIP server address;
- Option 121: to obtain classless static route. FXO GW will add these static routes to the static route table after it fetches them from DHCP server.

Figure 4-5-3 Configure DHCP Option

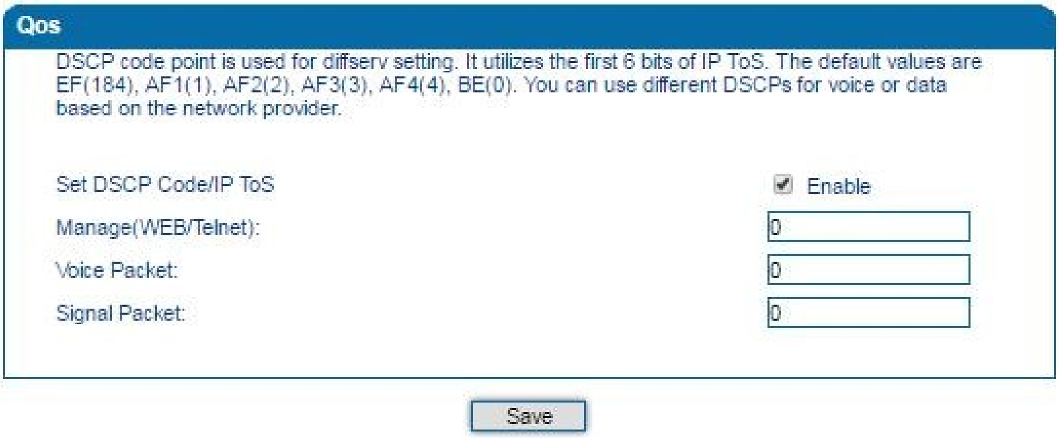


Network Interface: choose which VLAN to send request to DHCP server (or to receive information from DHCP server).

4.5.4 QoS

The FXO GW device can label QoS priority on the IP messages it sends out, so as to resolve network delay or network congestion. Meanwhile, the device can give different QoS tags for management-related packets of Web/Telnet, voice packets and signal packets.

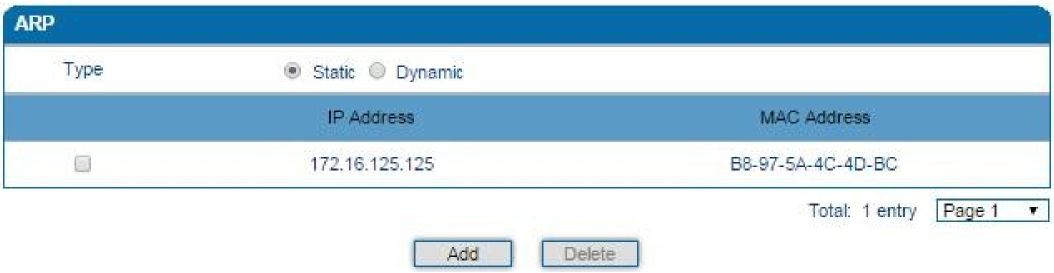
Figure 4-5-4 Qos



4.5.5 ARP

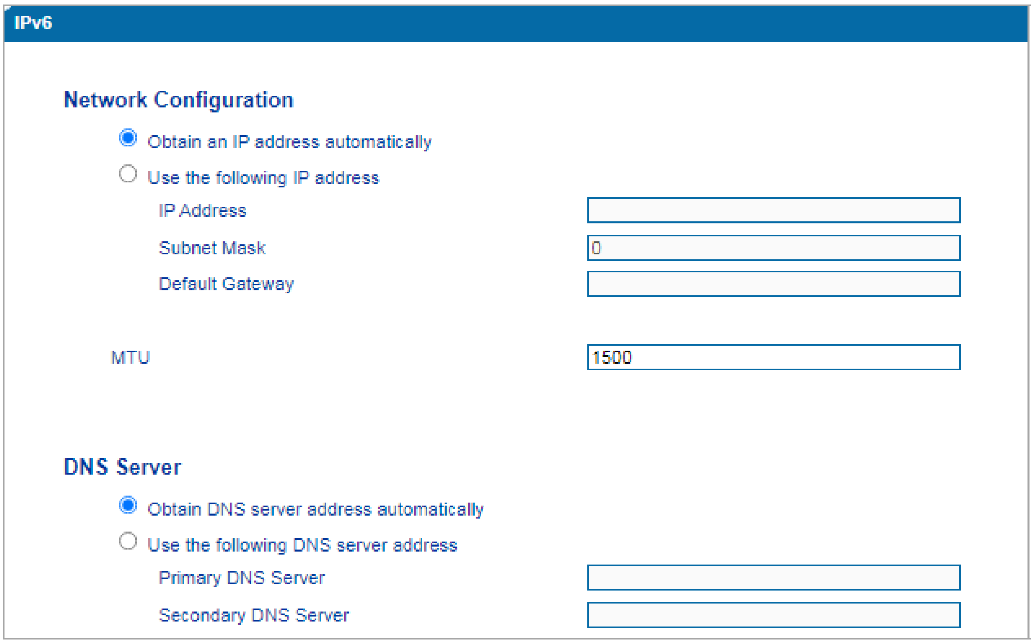
ARP is address resolution protocol, which helps to get the MAC address of a device through its IP address. Under TCP/IP network environment, each host is assigned with a 32-bit IP address, but MAC address needs to be known for message transmission in the physical network. In the above case, ARP can help convert IP address into MAC address.

Table 4-5-5 Explanation of Parameters for ARP



4.5.6 IPv6 Network

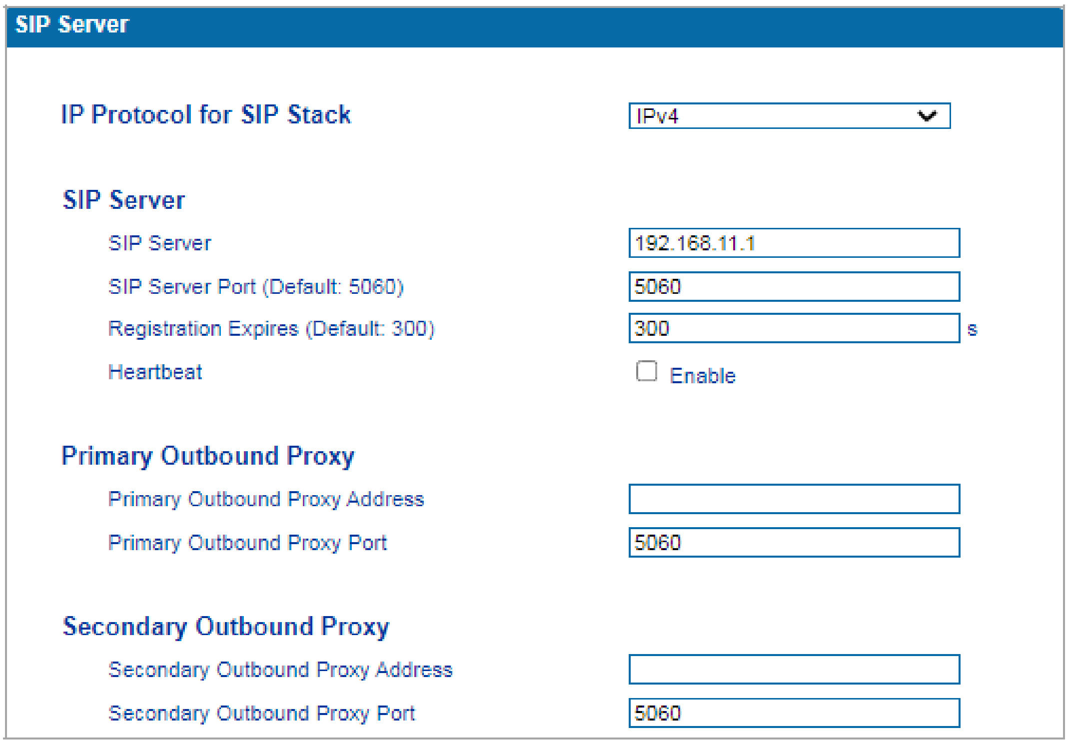
When IPv4&IPv6 is selected as the IP protocol in the local network, information such as IPv6 address, subnet mask, default gateway, and primary/secondary DNS servers can be configured. The IPv6 network configuration interface is shown in Figure 4-5-6:



4.6 SIP Server

SIP server is the main component of VoIP network and is responsible for establishing all SIP calls. SIP server is also called SIP proxy server or register server. Both IPPBX and softswitch can act as the role of SIP server.

Figure 4-6 Configure SIP Server Information



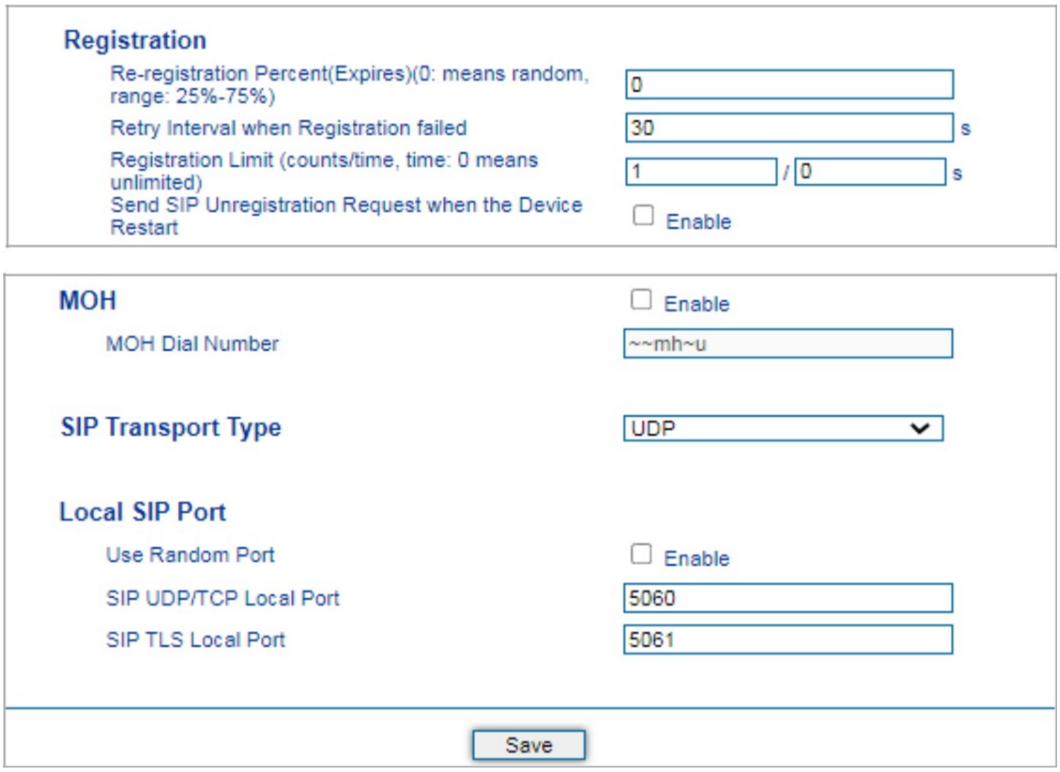


Table 4.6 Explanation of Parameters for SIP Server

Parameter	Explanation
IP Protocol for SIP Stack	Choose SIP protocol stack, support IPv4 and IPv6, please choose the corresponding protocol according to the actual connected server
SIP Server Address	The IP address or domain name of the SIP server. It is provided by VoIP service provider.
SIP Server port	The service port of the SIP server. It is 5060 by default.
Registration Expires	It is used to avoid excessively frequent registrations. When the time that is set expires, the FXO GW device will send register request to the SIP server. The time is 300s by default.
Heartbeat	Heartbeat is used to check the connection between the FXO GW device and SIP server.
Outbound Proxy Address	The IP address or domain name of outbound proxy server, which is provided by VoIP service provider.
Outbound Proxy Port	Service port of outbound proxy server. It is 5060 by default.
Retry Interval when Registration failed	The retry interval after a registration fails. Default: 30s
Registration times per	The maximum number of registrations in a second. 0 means no

second	limitation for registrations.
SIP Transport Type	The way of SIP-based transmission. It can be UDP, TCP, TLS or Automatic. Default: UDP.
Use Random Port	If this parameter is selected, the local port of the FXO GW device for using SIP services is chosen by random.
SIP UDP/TCP Local Port	The UDP/TCP port of FXO GW device for using SIP services. Default SIP UDP/TCP local is 5060.
SIP TLS Local Port	The TLS port of FXO GW device for using SIP services. Default SIP TLS local port is 5061.

Usually, SIP server does not participate in media processing. Under SIP network, media always use end-to-end negotiating. Simple SIP server is only responsible for the establishment, maintenance and cleaning of sessions, while relatively-complex SIP server (SIP PBX) not only provides basic calling and conversational support, but also offers rich services such as Presence, Find-me and Music On Hold.

SIP server based on Linux platform, such as: OpenSER、sipXecx，VoS，Mera etc.

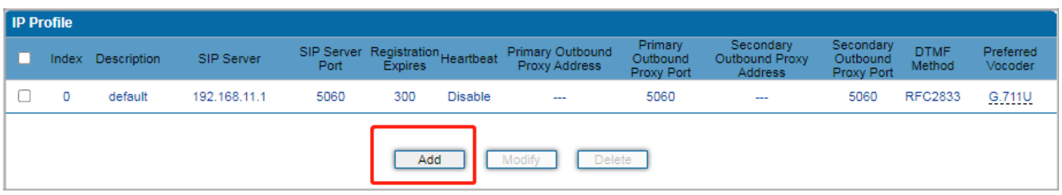
SIP server based on windows platform, such as :mini SipServer、Brekeke，VoIPswitch etc.

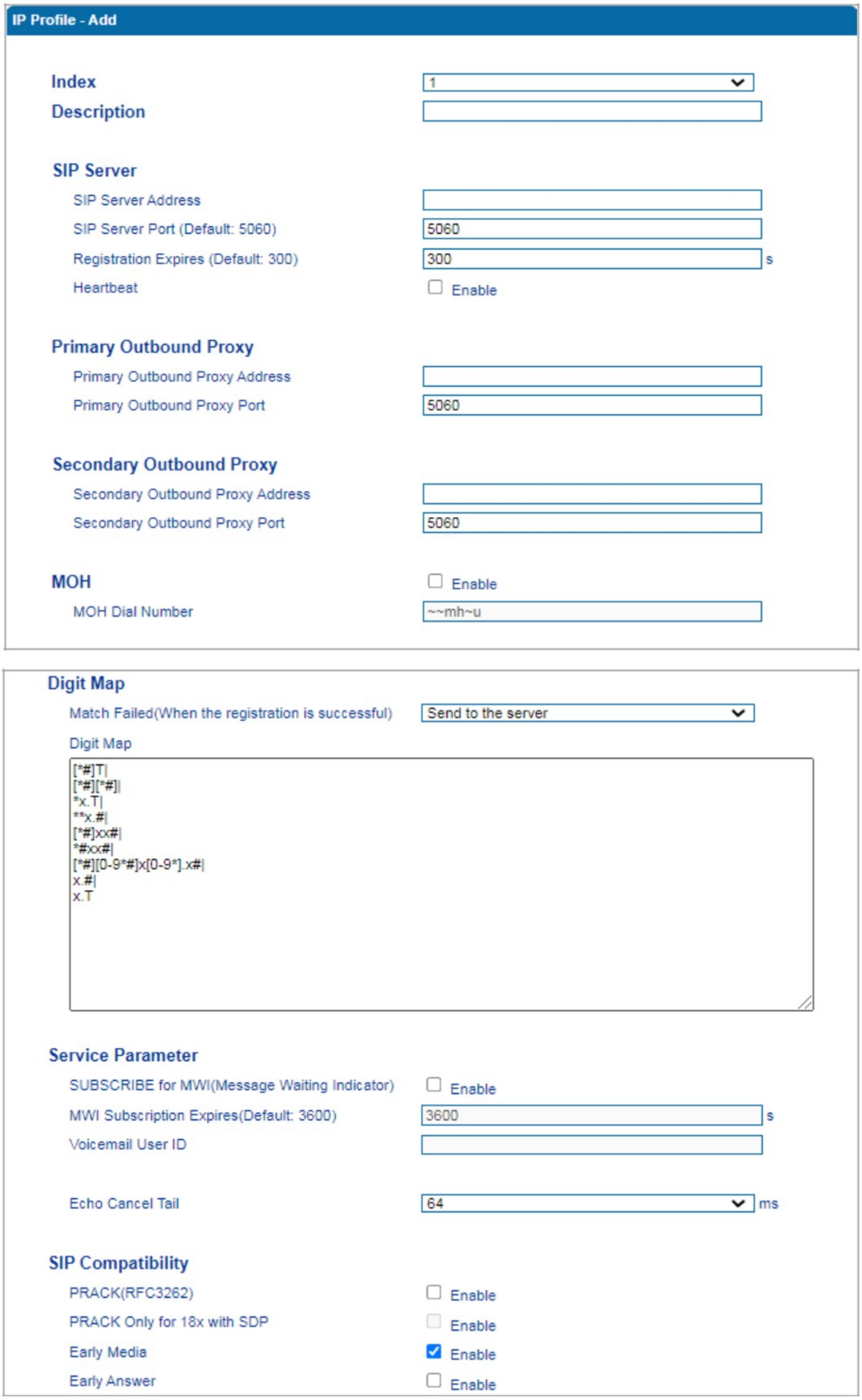
Carrier-grade soft switch platform, such as Cisco, Huawei, ZTE etc.

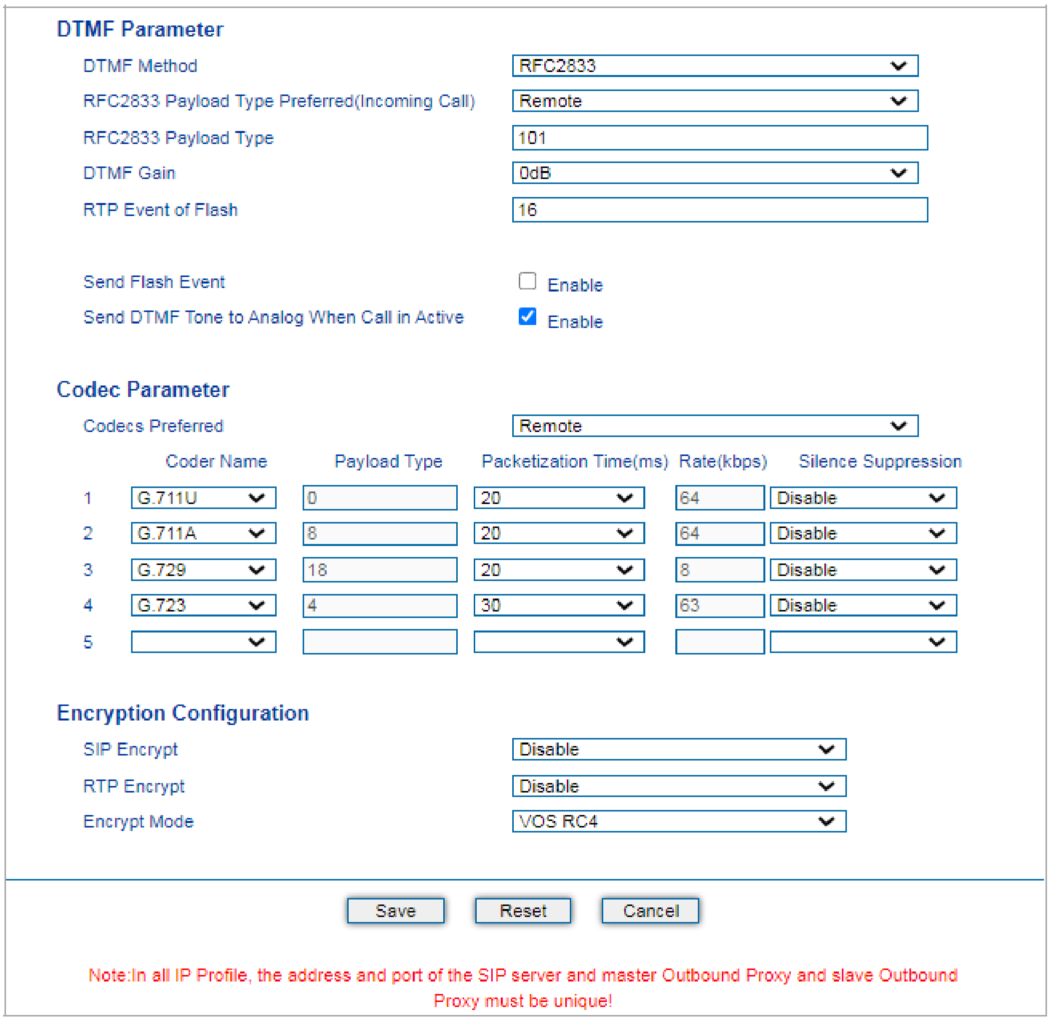
4.7 IP Profile

The device supports simultaneous registration to multiple SIP servers and making calls. Different ports can be configured with different SIP server addresses and use different voice codecs as needed. IP profiles are used to create SIP server addresses, proxy servers, dialing rules, service parameters, dialing parameters, voice codec and other parameter configuration for ports. When configuring the port, you can save the IP profile index and use it. For index configuration, refer to the "Port " page.

When the device is only registered to one SIP server, the IP profile does not need to be configured, and the default IP profile can be used. When the device needs to register to multiple SIP servers, click the "Add" button to create a new IP profile, as shown in the figure below:




IP Profile - Add
Index
1
Description
SIP Server
SIP Server Address
SIP Server Port (Default: 5060)
5060
Registration Expires (Default: 300)
300
s
Heartbeat
Enable
Primary Outbound Proxy
Primary Outbound Proxy Address
Primary Outbound Proxy Port
5060
Secondary Outbound Proxy
Secondary Outbound Proxy Address
Secondary Outbound Proxy Port
5060
MOH
Enable
MOH Dial Number
~~mh~u
Digit Map
Match Failed(When the registration is successful)
Send to the server
Digit Map
[*#]T|
[*#][*#]|
*x.T|
**x.#|
[*#]xx#|
*#xx#|
[*#][0-9*#]x[0-9*].x#|
x.#|
x.T
Service Parameter
SUBSCRIBE for MWI(Message Waiting Indicator)
Enable
MWI Subscription Expires(Default: 3600)
3600
s
Voicemail User ID
Echo Cancel Tail
64
ms
SIP Compatibility
PRACK(RFC3262)
Enable
PRACK Only for 18x with SDP
Enable
Early Media
Enable
Early Answer
Enable

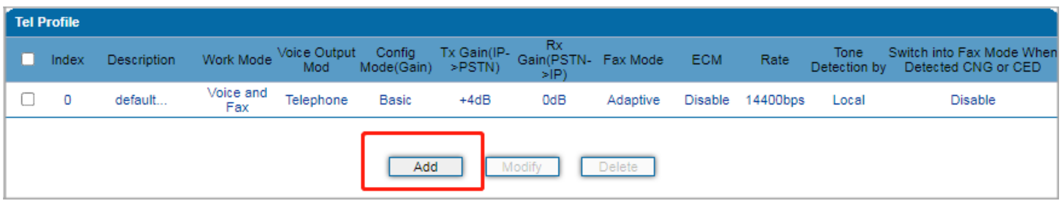


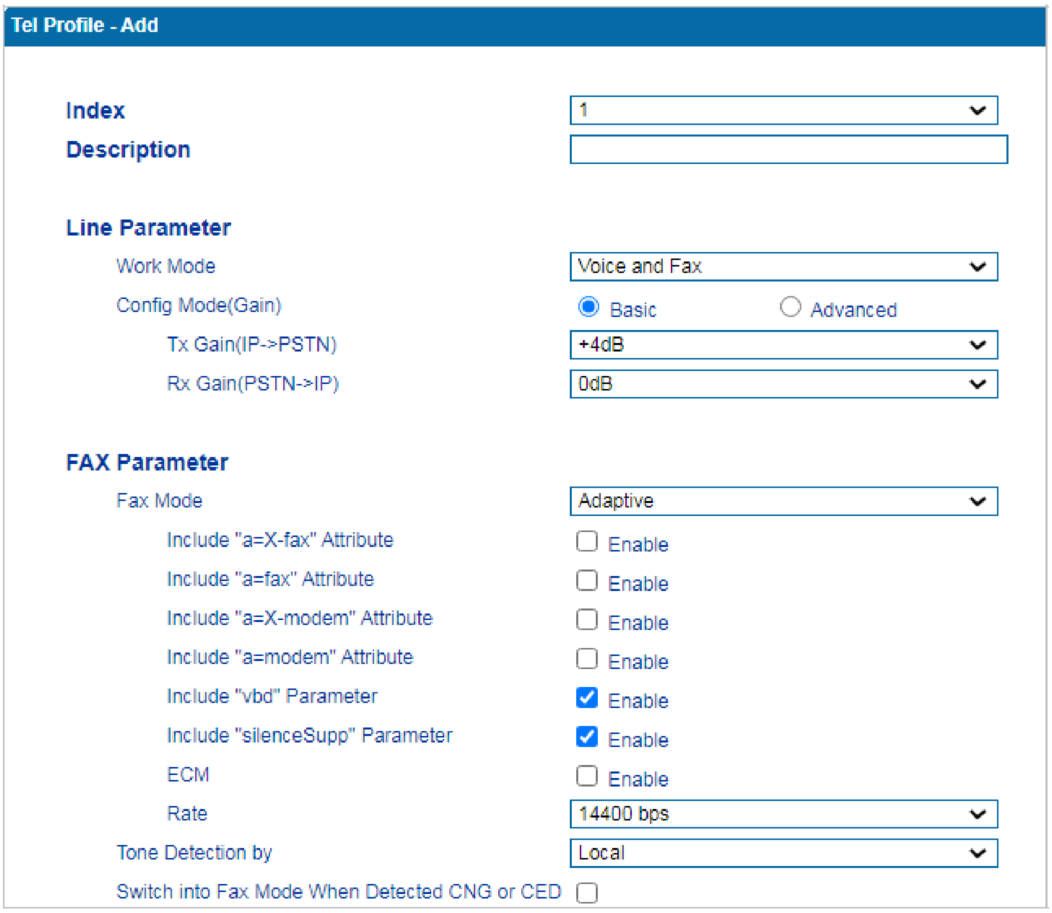
4.8 Tel Profile

The device supports setting different values for the line parameters to each port. Different ports can be configured with different gains and fax parameters as needed. Tel profile is used to create a configuration group of line parameters, service parameters, and fax parameters for the port. During port configuration, you can save the Tel profile index and use it. For index configuration, refer to the "Port " page.

Under normal circumstances, the Tel profile does not need to be configured, and the default Tel strategy can be used.

When you need to set different line parameters, business modes, or fax modes for different ports, you can add Tel profile through the "Add" button. As shown below:

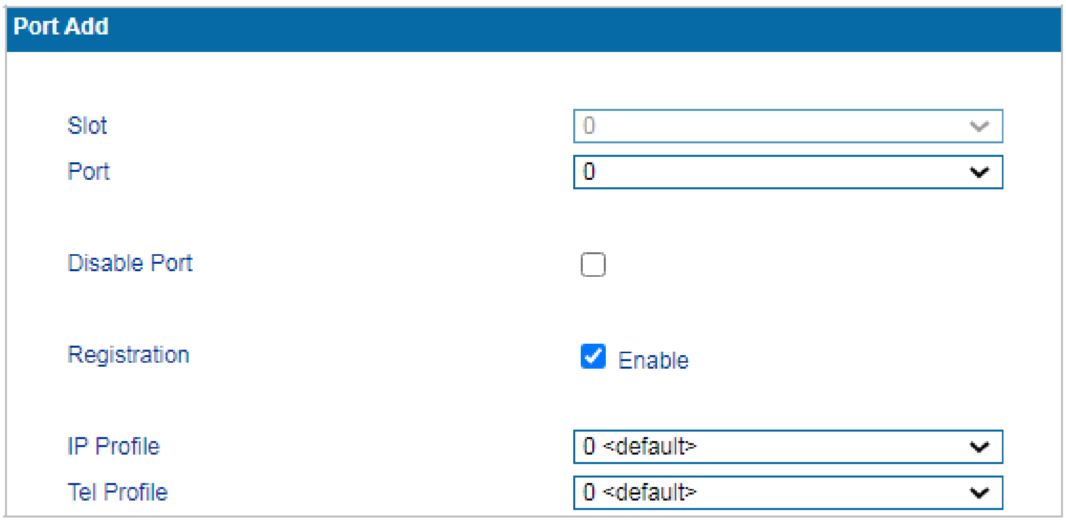




4.9 Port

A unique SIP account used for registration can be configured for each FXO port of FXO GW device. Parameters of the SIP account include port number, whether to register, primary display name, primary SIP user ID, primary Authenticate ID, primary Authenticate password, off-hook auto-dial number, caller ID and so on.

Figure 4-9 Configure SIP Account for Port Registration



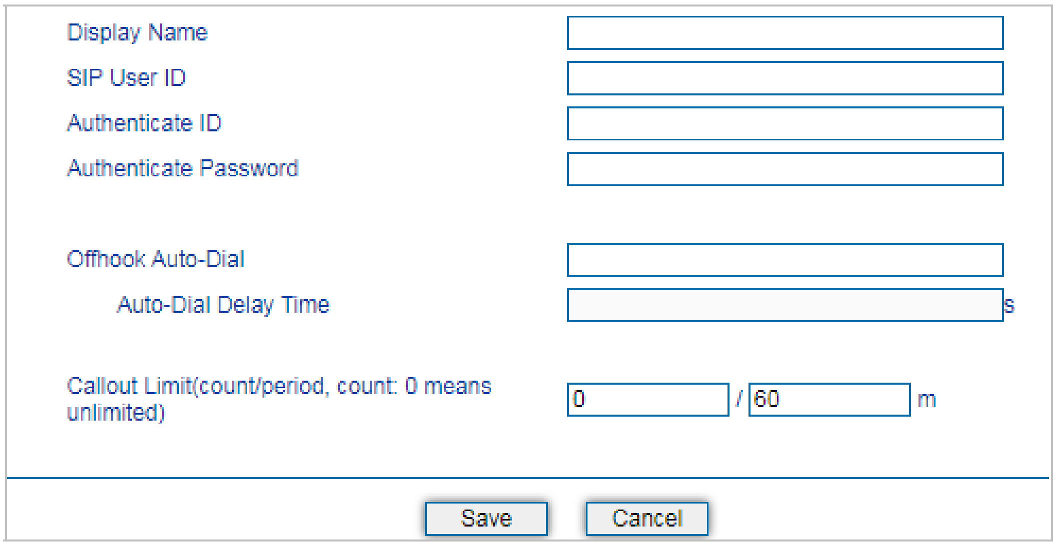


Table 4-9 Explanation of Parameters Related to SIP Registration

Parameter	Explanation
Port	The FXO port corresponding to this account
Disable port	Whether to disable port temporally
Registration	Whether to enable registration for the port
IP Profile	IP profile (need to be created in advance, refer to 4.6 IP profile configuration)
Tel Profile	IP policy (need to be created in advance, refer to 4.6 IP policy configuration) Tel profile(need to be created in advance, refer to 4.7 Tel profile Configuration)
Display Name	Description of SIP account. It is used to identify the SIP account.
SIP User ID	User ID of the SIP account, which is provided by VoIP service provider (ITSP) for registration. Usually it is in the form of digits similar to phone number or an actual phone number.
Authenticate ID	SIP service subscriber's authenticate ID used for authentication of registration. It can be identical to or different from SIP User ID.
Authenticate password	SIP service subscriber's authenticate ID used for authentication of registration
Off-hook Auto-dial	An extension or phone number is pre-assigned here so that the number is automatically dialed as soon as user picks up the phone
Auto-dial Delay Time	How long the auto-dial number is prolonged. If it is set as 3s, the auto-dial number is dialed after 3 seconds pass.

Callout Limit	Maximum number of call out within the configured time

4.10 Advanced

4.10.1 Line Parameter

On the **Advanced→ line** page, you can configure FXO parameters which include for call progress tone, auto gain control, fax parameters and so on.

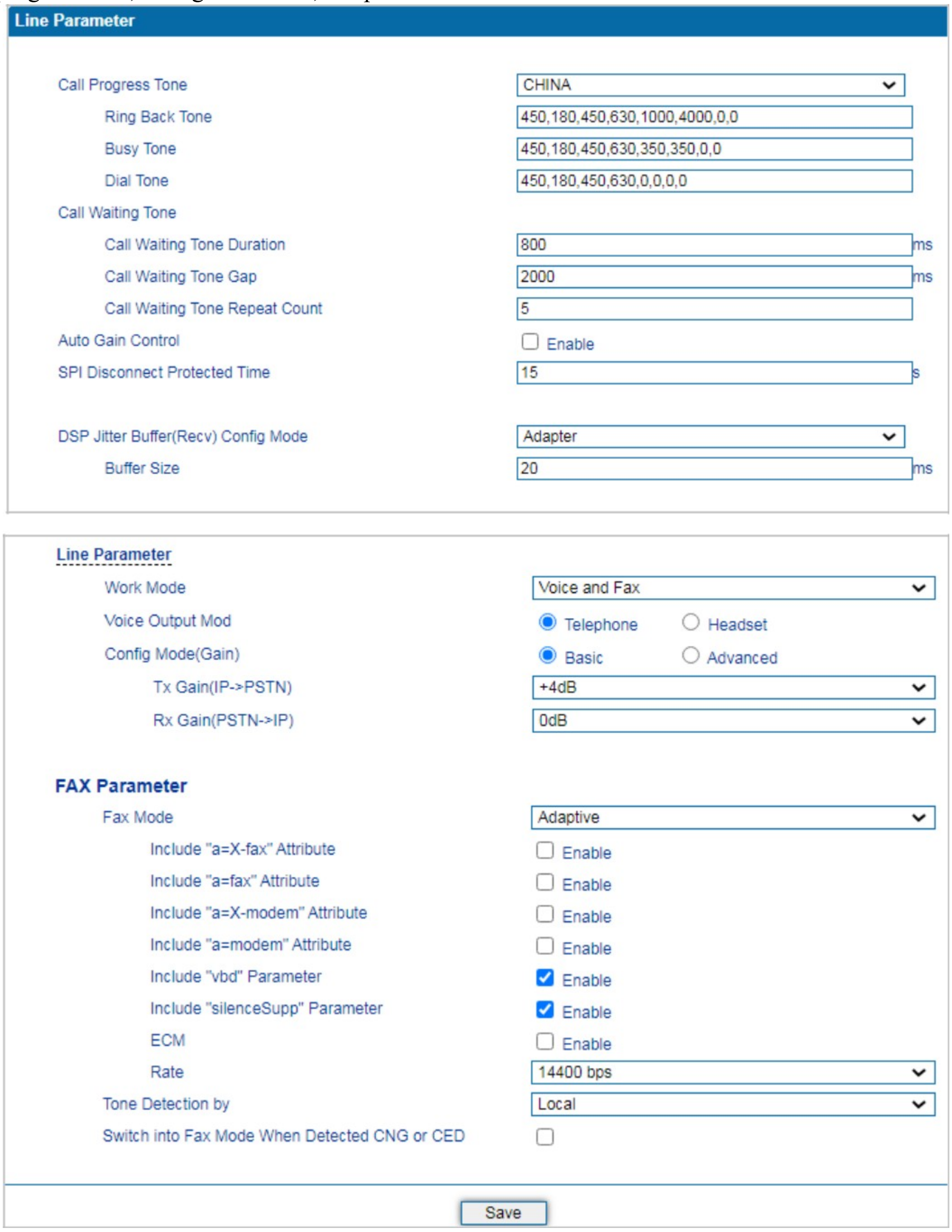


Table 4.10.1 Explanation of Line Parameters

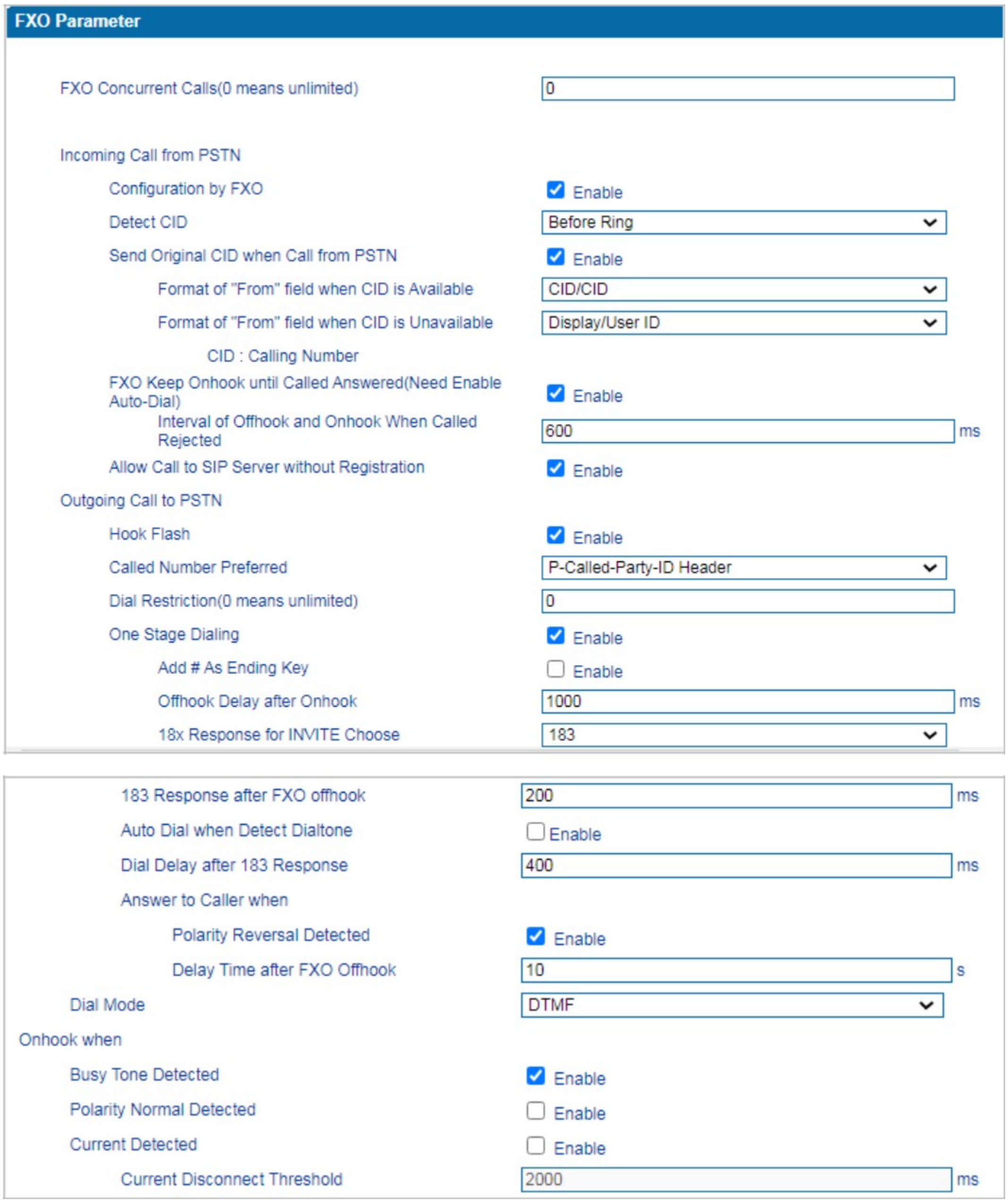
Parameter	Explanation
Call Process Tone	The signal tone standard after a phone is picked up. Choose national standards from the drop-down box. Default value is USA.
Call Waiting Tone	Set the duration, interval and number of repetitions for the call waiting tone
Auto Gain Control	Whether to enable automatic gain control
Work Mode	To set the FXO ports work in both Voice and Fax mode. There are several configure options: • Voice and FAX: to be able to make call and use FAX service • Voice Only: allows to make call only, Fax doesn't work if you connect a fax machine • Fax Only: allows to make Fax call only. • POS only: allows to connect POS terminal only
Gain mode	IP to PSTN(RX): adjust gain value to analog phone PSTN to IP(TX): adjust gain value from analog phone
FAX Parameter	The Gateway-128odevice supports the three fax modes: T.38 (IP-based), T.30 (Pass-Through) and Adaptive Fax Mode (automatically match with the peer fax mode).
Fax Mode	There are three fax modes: T.38, T.30(Pass-through), and Adaptive.
Include "a=X-fax" Attribute	If this parameter is enabled, "a=X-fax" attribute will be carried in SDP
Include "a=fax" Attribute	If this parameter is enabled, "a=fax" attribute will be carried in SDP
Include "a=X-modem" Attribute	If this parameter is enabled, "a=X-modem" attribute will be carried in SDP
Include "a=modem" Attribute	If this parameter is enabled, "a=modem" attribute will be carried in SDP
ECM	Whether to enable 'Error Correction Mode'（ECM）.
Rate	The rate of sending or receiving fax, default value is 14400bps.
Tone Detection by	Fax sound is detected by caller, callee or automatically.
Switch into Fax Mode	If this parameter is enabled, the system will switch into fax

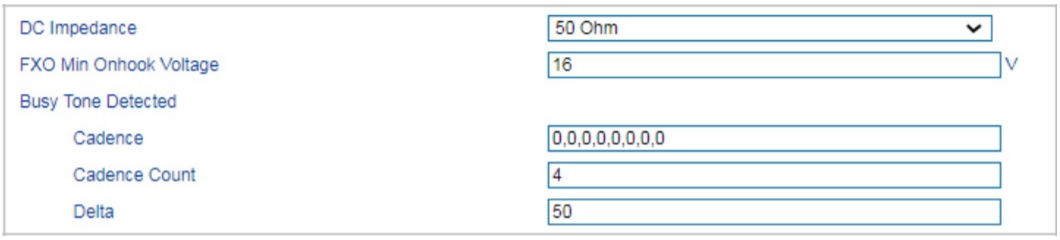
When Detect CNG or CED	mode when CNG or CED is detected.

4.10.2 FXO Parameter

On the **Advanced→ FXO Parameter** page, you can configure FXO parameters which include incoming/outgoing call from PSTN, DC impedance, busy tone detected and so on.

Figure 4-10-2 Configure FXO Parameters





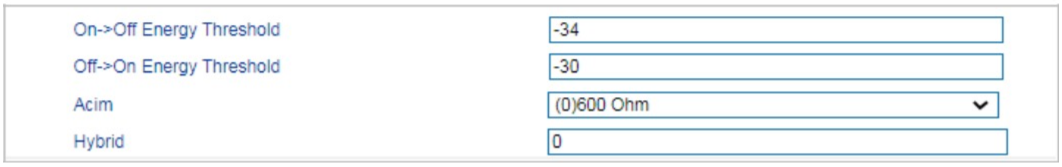


Table 4.10.2 Explanation of FXO Parameters

Parameter	Explanation
FXO Concurrent Calls	Limit the number of concurrent FXO calls (0 means no limit, and the maximum number is the total number of FXO ports) which means the number of call requests received by the gateway per second. to prevent the call server from initiating a large number of calls instantly, causing traffic shocks. It is designed to prevent the server from initiating a large number of calls at the same time and causing traffic shocks.
Configuration by FXO	When the incoming call from PSTN, you can enable or disable the FXO configuration. The FXO configuration function includes Detect CID, Send Original CID and so on.
Detect CID	When a call comes to the FXO port, FXO detects the calling number and the order of ringing. The system has two modes: first ringing and then detecting CID, first detecting CID and then ringing. The PSTN line sending CID methods usually include: sending CID before ringing, and sending CID after ringing. Therefore, when FXO detects CID, it needs to be set according to the way of PSTN line sending CID.
Send Original CID when Call from PSTN	When enabled, the caller ID of the extension will display on the PSTN side when dialing the extension.When it is not enabled, the caller ID of the extension will be display the number of the FXO port.
FXO Keep On-hook until Called Answered	After enabled, when the PSTN calls into the FXO gateway, the FXO device will go off-hook after the extension number dialed is connected. If this function is disabled, when the user dials in to the FXO port, the FXO first off-hook , and then initiates a call request to the IP.
Allow Call to SIP Server without Registration	Allow the port to initiate a call request without registering to the SIP Server. At this time, the device works in point-to-point mode.

Called Number Preferred	When making an outgoing call, the device obtains the called number from the SIP message of the remote end. According to the content of the SIP request, the called number may be obtained from the following three fields: ● P-Called-Party-ID Header ● Request-line) ● To header
Dial Restriction	When FXO gateway calls the PSTN, set a simultaneous dialing limit (0 means no restriction).
One Stage Dialing	Enabled by default, the call mode of FXO gateway means that when the FXO device makes an outgoing call, the called number obtained from the SIP message is sent to the analog end digit by digit at a time.
Add # As Ending Key	When FXO gateway makes an outgoing call, it will automatically add # after the original number as the end key to dial out together.
Off-hook Delay after On-hook	When FXO gateway calls the PSTN, the delay time for the FXO device to go off-hook after on-hook (default 1000ms).
18x Response for INVITE Choose	When the device receives the INVITE request from the remote end, it sends 180 or 183 as a temporary response code to the IP side.
183 Response after FXO Off-hook	When FXO gateway calls the PSTN, after the FXO goes off-hook, it will respond to the 183 to the IP side after a specified time (200ms by default).
Auto Dial when Detect Dial tone	When FXO gateway calls the PSTN, the FXO gateway will automatically dial out when it detects the dial tone of the PSTN.
Dial Delay after 183 Response	When FXO gateway calls the PSTN, it sends 183 messages after a specified interval to dial the PSTN.
Polarity Reversal Detected	When FXO gateway calls the PSTN, the way that FXO answers the caller is to detect the polarity reversal. After enabled, if a polarity reversal is detected, it will be reported to the caller for response. If the PSTN side cannot provide the polarity reversal detected, this function is invalid.
Delay Time after FXO Off-hook	The time for the FXO device to detect the polarity reversal and answer the caller should be less than this value. The system defaults to 10s. If the time is exceeded, the called is considered to have answered. This parameter is mostly used when there is no reverse polarity on the PSTN.
Dial Mode	FXO gateway calls the PSTN and supports 3 dialing methods:

	DTMF, Pulse, Pulse before DTMF
On-hook when Busy Tone Detected Polarity Normal Detected Current Detected	After enabling this function, the FXO gateway calls the PSTN, the FXO device hang up when: busy tone detected ,polarity normal detected and current detect .
DC Impedance	The impedance parameters when FXO gateway is connected to PBX or PSTN.
FXO Min On-hook Voltage	Minimum on-hook voltage of FXO gateway
Busy Tone Detected Cadence	The busy tone detection cadence needs to be set according to the busy tone system of the PSTN. If you do not know the busy tone standard, you can use the busy tone detection function to detect the busy tone cadence.
Cadence Count	The cadence count is used to detect the validity of the busy tone. When multiple busy tone beats are continuously detected, it is as a valid busy tone.
Delta	The error value of busy tone detection cadence
On→Off Energy Threshold	The energy threshold of busy tone from On to Off .
Off→On Energy Threshold	The energy threshold of busy tone from Off to On.
Acim	The value of AC impedance
Hybrid	The value of hybrid balance parameters

4.10.3 Media Parameter

Media parameters mainly include RTP start port, DTMF parameter, preferred Vocoder, etc.

Figure 4-10-3 Configure Media Parameters

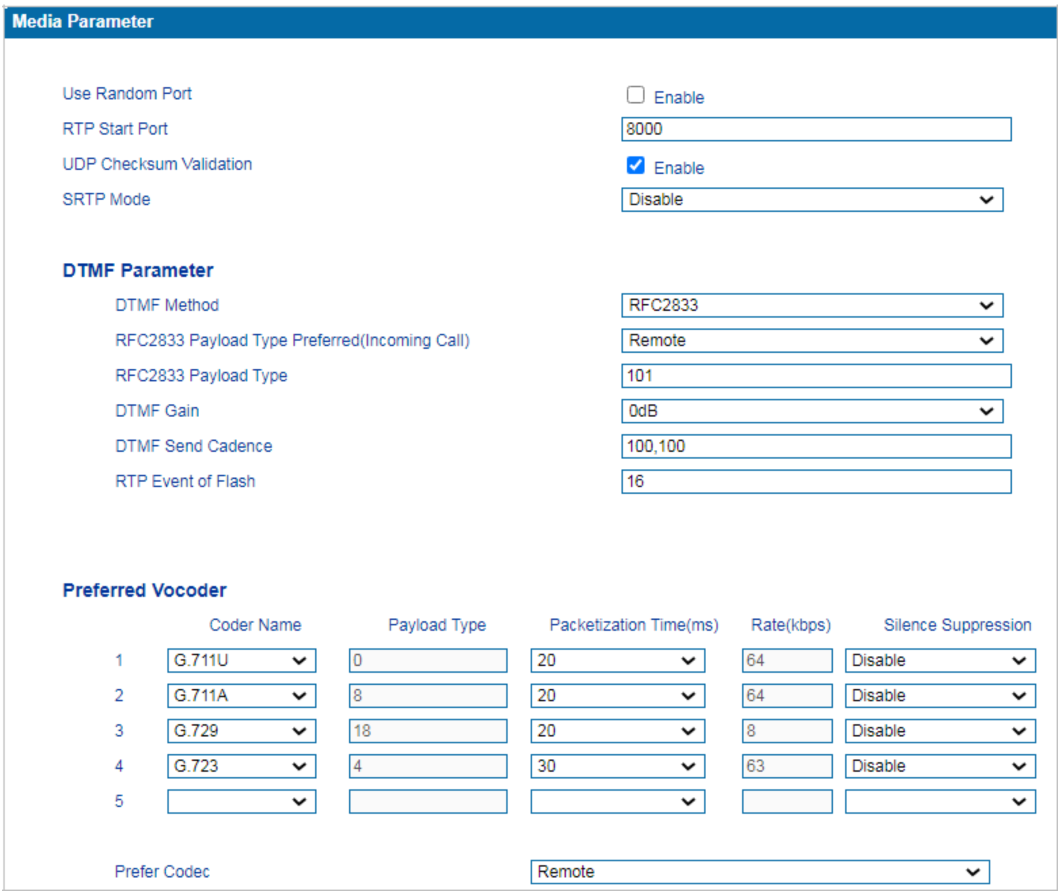


Table 4.10.3 Explanation of Media Parameters

Parameter	Explanation
Use Random Port	If this parameter is enabled, the FXO GW device will choose a port by random as the start port for RTP.
RTP Start Port	When 'Use Random Port' is not selected, you need to configure a start port for RTP. Default RTP start port is 8000
UDP Checksum Validation	Choose whether to enable header checksum of UDP
SRTP Method	Include Disable, AUTO and Force
RFC2833 Payload Type Preferred (Incoming Call)	For an incoming call, choose local or remote RFC2833 payload type as the preferred payload type
RFC2833 Payload Type	Local payload value, default value is 101
DTMF Gain	Default value is 0 DB
DTMF Send Cadence	The interval for sending DTMF signal. The default value is 200ms.
Event of Flash	If this parameter is enabled, the FXO GW device will send

	flash-hook event to remote terminal, and thus user does not need to handle it locally
Coder Name	The device supports G.729, G.711U, G.711A, G.723, G.726-16/24/32/40. When outgoing calls are made, G.729 will be used.
Payload Type	Each kind of coding has a unique load value, refer to RFC3551.
Packetization Time	The time for voice packaging
Rate	Voice data flow rate; It is defaulted by system.
Silence Suppression	Default value is 'disabled'. If this parameter is enabled, VoIP transmission bandwidth can be saved, and meanwhile network congestion can be avoided.
Prefer Codec	Choose local or remote codec as the preferred codec

4.10.4 Service Parameter

Service parameters include timeout for dialing, digit map,MWI message and so on.

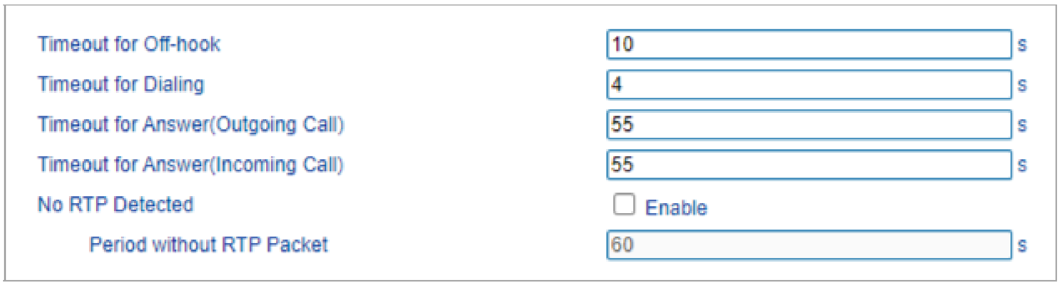


Table 4.10.4-1 Explanation of Service Parameters

Parameter	Explanation
Timeout for off-hook	Mainly used to define a timer that when the user is off hook an analog phone without dial any digits
Timeout for dialing	With the help of dialing timeout, you can limit the time between two digits while users are typing the digits of a number through an extension. If the timeout expires, the gateway will consider the dialing has finished and will try to send message to SIP server. Default value is 4 seconds.
Timeout for answer(Outgoing call)	This parameter determines how long the caller party will wait for answer when making outgoing calls through a phone.
Timeout for answer(Incoming call)	This parameter determines how long the phone rings when there are incoming calls

No RTP Detected	If this parameter is enabled, the situation will be detected when there is no RTP packets received during the set time period.
Period without RTP Packet	The time period when there is no RTP packets received.

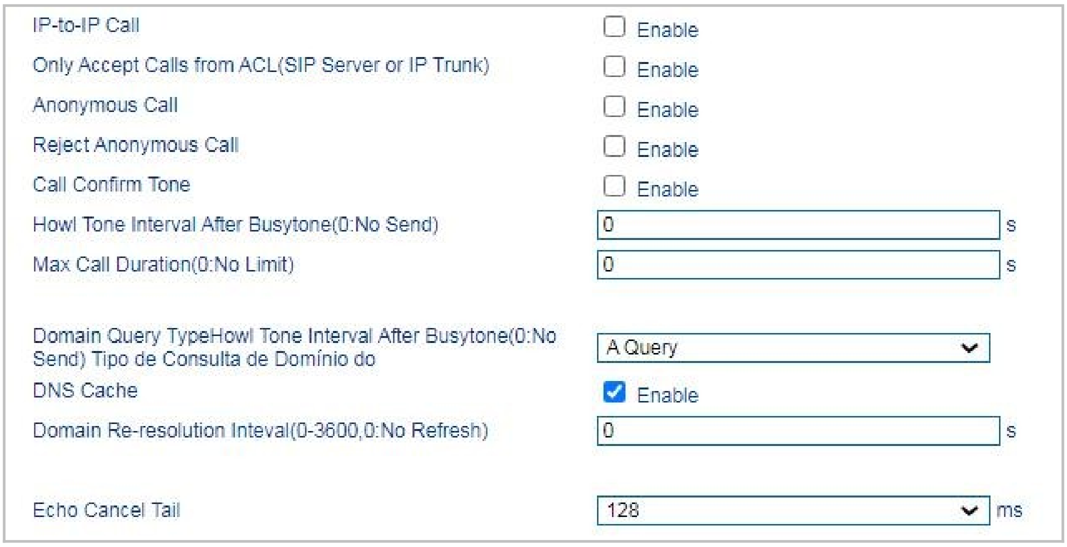


Table 4.10.4-2 Explanation of Service Parameters

Parameter	**Explanation**
IP-to-IP Call	If this parameter is enabled, user can dial IP address through a phone to call destination gateway.
Only Accept Call from ACL (SIP server or IP Trunk)	If this parameter is enabled, the device only accepts incoming call from SIP server only. Default value is 'not enable'.
Anonymous Call	If this parameter is enabled, 'anonymous' will be included in SIP message.
Reject Anonymous Call	If this parameter is enabled, all anonymous calls will be rejected. (default is 'not disable').
Call Confirm Tone	After enabling, the device will play a ringback tone even if it does not receive a 180 response.
Howl Tone Interval After Busytone (0: No Send)	How long does the howl tone play after the busy tone
Max Call Duration(0: No Limit)	If it exceeds the set time, the call will hang up directly (default 0 means no limit)
Domain Query Type	The way to query the domain supported: A query, SRV query, and NAPTR query

DNS Cache	Whether to enable DNS cache
Domain Re-resolution Interval	The time of domain resolution interval
Echo Cancel Tail	When there is an echo during a call, the dsp can eliminate the echo. When the echo cancellation time is longer than the echo delay, the echo can be eliminated. The default is 128 milliseconds.

Digit map is used for number dialing of calls through FXO ports of the FXO GW device.

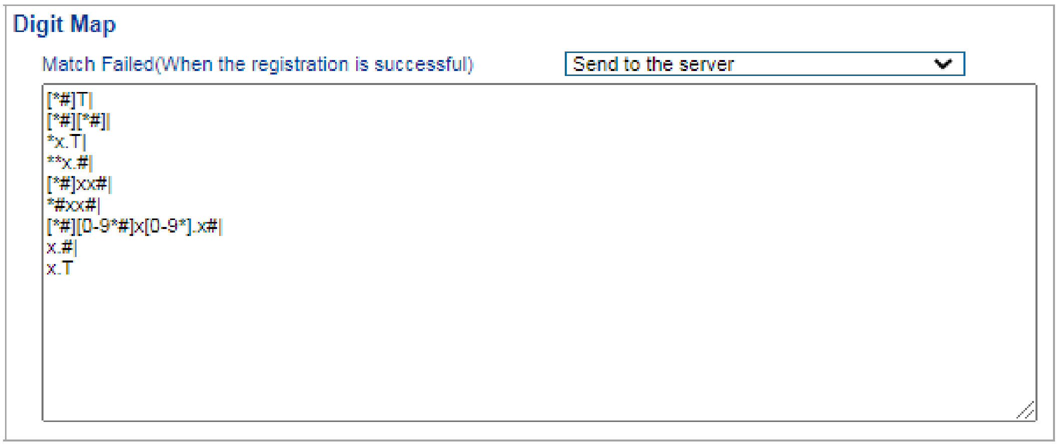


Table 4.10.4-3 Explanation of Parameters for Digit Map

Supported Objects	Digit	0-9
	T	Timer
	DTMF	A digit, a timer, or one of the symbols of A, B, C, D, #, or *
Range	[ ]	One or more DTMF symbols enclosed in the [], but only one DTMF symbol can be selected
Range	()	One or more expressions enclosed the (), but only one can be selected
Separator	|	Separate expressions or DTMF symbols.
Subrange	-	Two digits separated by hyphen (-) which matches any digit between and including the two digits.
Wildcard	x	Matches any digit of 0 to 9
Modifiers	.	Matches 0 or more times of the preceding element
Modifiers	?	Matches 0 or 1 times of the preceding element

4.10.5 SIP Compatibility

SIP parameters include attended transfer trigger, early media, session timer, heartbeat interval and so on.

Figure 4-10-5.1 Configure SIP Parameters

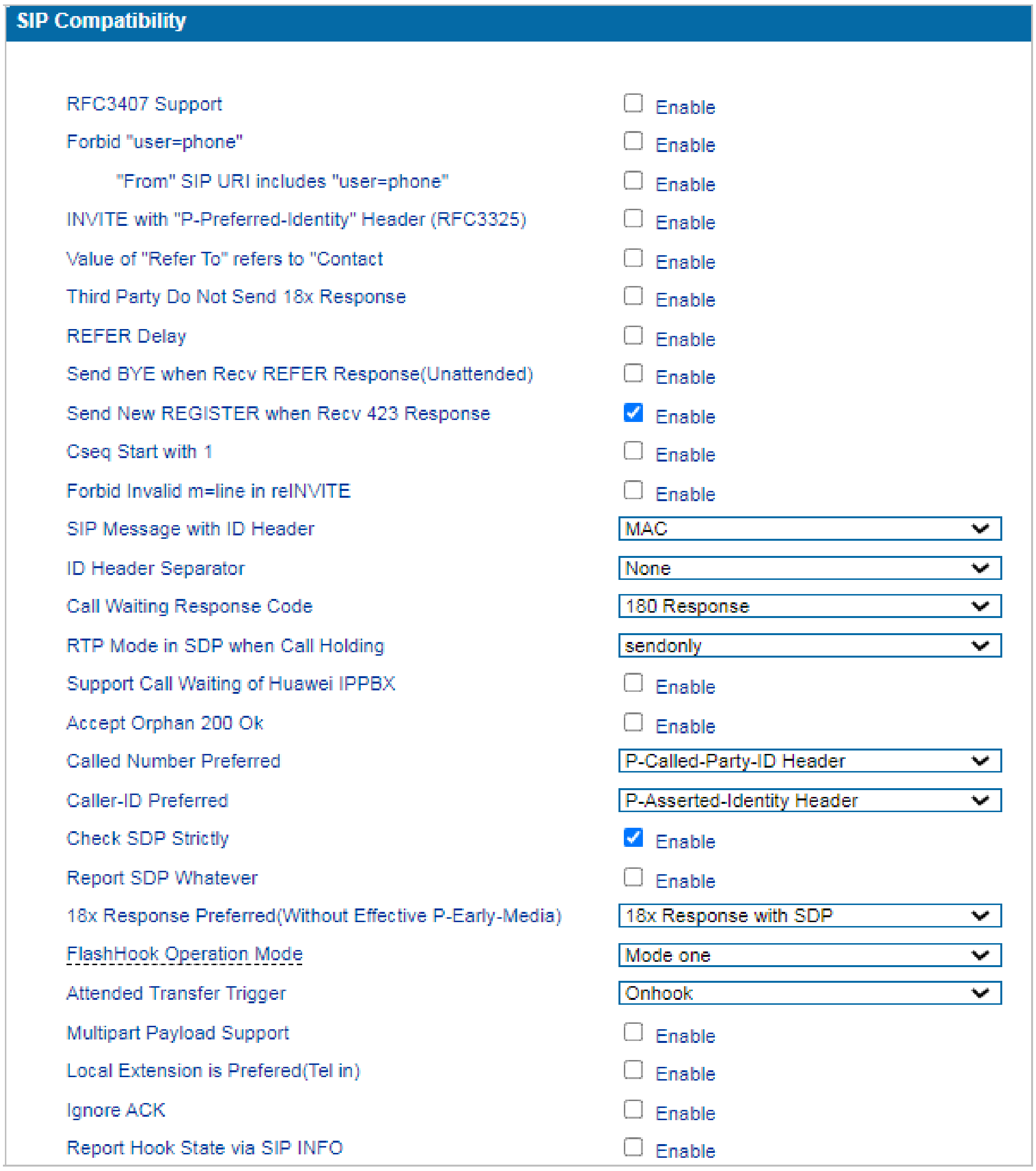


Table 4.10.5-1 Explanation of SIP Parameters

Parameter	Explanation
RFC3407 Support	Whether to enable RFC3407 support. If this parameter is enabled, the device will support RFC3407 which defines the SDP capability of backward compatibility.

"From" SIP URI Includes "user=phone"	If this parameter is enabled, 'user=phone' will be contained in URI. When calls are routed to PSTN network, the called number will be got from user name. Default value is 'not enable'.
INVITE with "P-Preferred-Identity" Header (RFC3325)	If this parameter is enabled, "P-Preferred-Identity" header will be added in INVITE message for anonymous call (Support RFC3325).
Value of "Refer To" refers to "Contact"	If this parameter is enabled, 'contract header' needs to be filled in in the 'refer to' field of a SIP message.
Third Party Do Not Send 18x Response	If this parameter is enabled, the third party will not send 18x response during an attended transfer.
Send BYE when Recv REFER Response (Unattended)	If this parameter is enabled, the third party will send BYE to release session after receiving REFER during a blind transfer.
Send New REGISTER when Recv 423 Response	If this parameter is enabled, the value of 'expires' header will be automatically updated and REGISTER will be re-sent after receiving of 423 response.
CSeq Start with 1	If this parameter is enabled, the value of CSeq starts with '1'.
Forbid Invalid m=line in reINVITE	If this parameter is enabled, the device will prevent 'invalid m=line' from being carried in the SDP of re-INVITE.
SIP Message with ID Header	SIP message with ID field, including MAC or SN optional
ID Header Separator	Select a separator for the MAC or SN of the ID field
Call Waiting Response Code	User can choose 180 or 182 as call waiting response code
RTP Mode in SDP when Call Holding	Use 'send only' or 'inactive' as RTP mode during call holding.
Support Call Waiting of Huawei IPPBX	If this parameter is enabled, the device will support call waiting of Huawei IPPBX.
Accept Orphan 200 OK	If this parameter is enabled, the FXO GW device will support different 'to-tag 200 OK' in an INVITE session.
Called Number Preferred	Choose P-Called-Party-ID header or Request-Line
Caller-ID Preferred	Choose P-Asserted-Identity header or From Header
Report SDP Whatever	If this parameter is enabled, SDP will be reported anytime
18x Response Preferred	Choose '18x Response with SDP', 'Last 18x Response' or 'Local Ring Tone Only'

Flash-hook Operation Mode	Choose Mode one, Mode two or Mode three
Attended Transfer Trigger	Choose ‘On-hook’ or ‘Flash-hook +4’
Ignore ACK	When enabled, the device will not resend the 200 OK response message even if it does not receive an ACK message after going off-hook

Figure 4-10-5.2 Configure Default SIP Parameters & Early Media

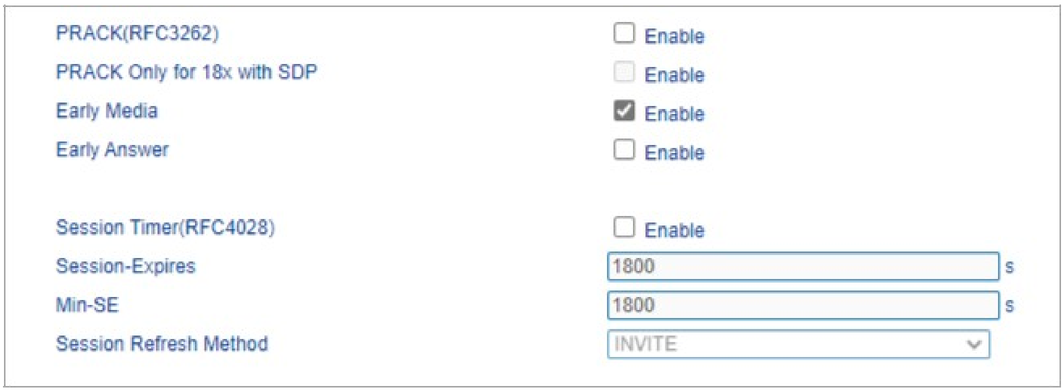


Table 4.10.5-2 Explanation of Default SIP Parameters & Early Media Parameters

Parameter	**Explanation**
PRACK(RFC3262)	If this parameter is enabled, the FXO GW device supports reliable transmission of provisional response
PRACK Only for 18x with SDP	If this parameter is enabled, only PRACK will be sent when there’s SDP in 18x response
Early Media	If this parameter is enabled, the FXO GW device supports the receiving of Early Media.
Early Answer	If this parameter is enabled, the FXO GW device supports early answer
Session Timer (RFC4028)	Whether to enable ‘session timer’, default value is ‘not enable’.
Session-Expires	The interval for refreshing session; default value is 1800s. The Session-Expires header field conveys the session interval for a SIP session.
Min-SE	The minimum interval for refreshing session; default value is 1800s. The Min-SE header field indicates the minimum value for the session interval.
Session Refresh Method	The method to refresh session; default value is INVITE.

Figure 4-10-5.3 Configure Timer in SIP Protocol

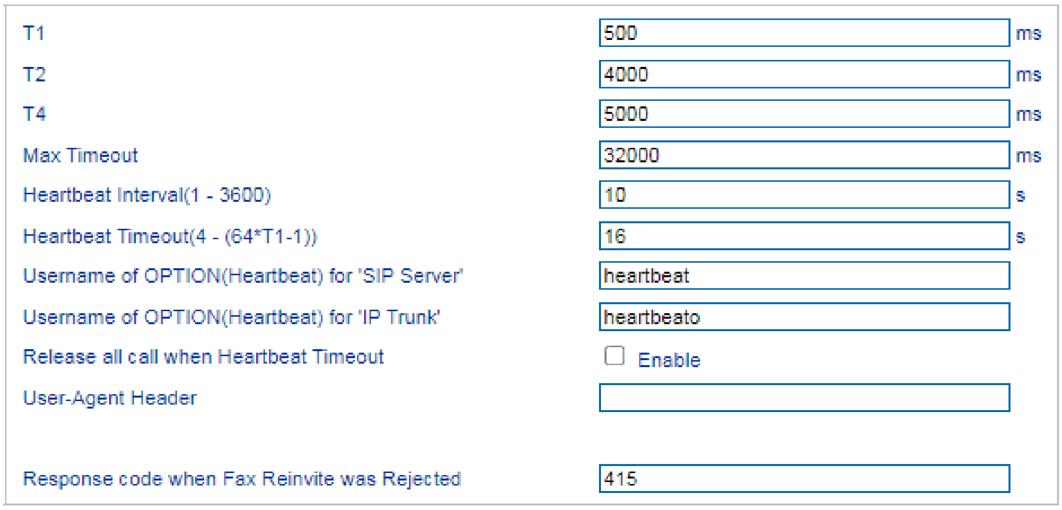
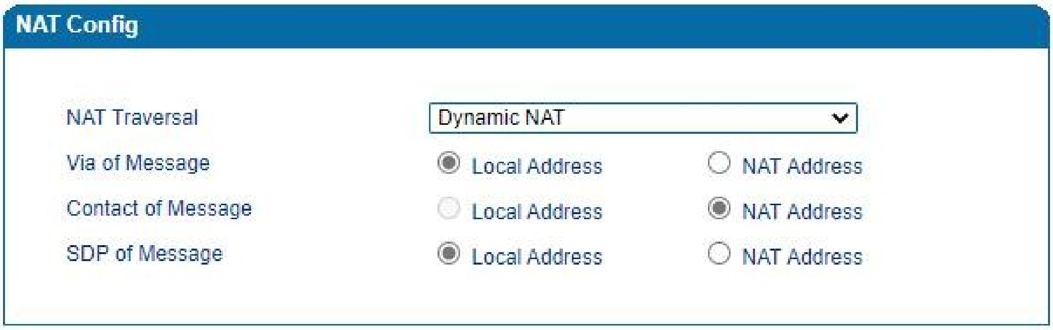


Table 4.10.5-3 Explanation of Timer Parameters in SIP Protocol

Parameter	Explanation
T1	Value of T1 timer in SIP protocol, default is 500ms
T2	Value of T2 timer in SIP protocol, default is 4000ms
T4	Value of T4 timer in SIP protocol, default is 5000ms
Max Timeout	The max timeout of sending or receiving SIP messages, default is 32000ms
Heartbeat Interval	The interval for sending heartbeat message, Default is 10s.
Heartbeat Timeout	The timeout for heartbeat message to be sent, default to 16s
Username of OPTION(Heartbeat) for "SIP Server"	The user ID part of OPTION SIP message in the heartbeat request for SIP server
Username of OPTION(Heartbeat) for "IP TRUNK"	The user ID part of OPTION SIP message in the heartbeat request for IP trunk
Release all call when Heatbeat Timeout	If enabled, when the heartbeat times out, all calls will be disconnected
User-Agent Header	Customize User-Agent header field value
Response code when Fax Reinvite was Rejected	Customize the SIP response code when fax re-invite was rejected

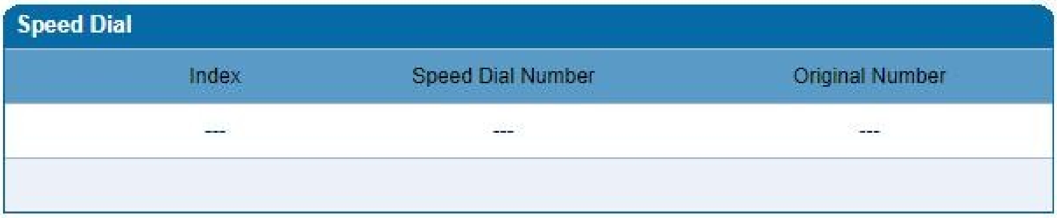
4.10.6 NAT Parameter



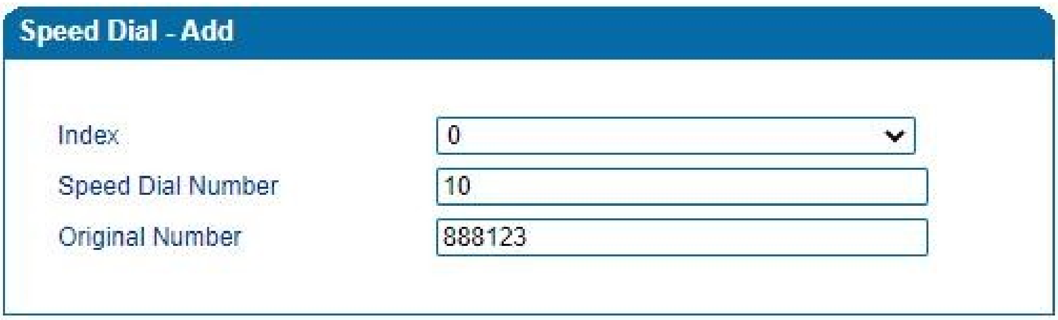
NAT Traversal (Network Address Translator Traversal) is a computer networking technique of establishing and maintaining Internet protocol connections across gateways that implement network address translation (NAT). NAT breaks the principle of end-to-end connectivity originally envisioned in the design of the Internet.

STUN（Simple Traversal of UDP over NATs） is a lightweight protocol that allows applications to discover the presence and types of NATs and firewalls between them and the public Internet. It also provides the ability for applications to determine the IP addresses allocated to them by the NAT. STUN works with many existing NATs, and does not require any special behavior from them. STUN doesn't support TCP connection and H.323.

4.10.7 Speed dial



Speed dial is a function that is available on telephones which provides an easy method of calling a telephone number by pressing fewer digits on the keypad. The tool enables one to save, organize, and have easy and quick access to regularly dialed numbers.



Speed Dial

	Index	Speed Dial Number	Original Number
☐	0	10	888123

Total: 1 Entry

4.10.8 Feature Code

Feature Code

Feature	Codes	Use Default	Status
Device Function			
Inquiry LAN IP	*158#	☑	Enable
Inquiry Phone Number	*114#	☑	Enable
Inquiry PortGroup Number	*115#	☑	Enable
Inquiry Registration Status	*168#	☑	Enable
Remove Login Limit	*154#	☑	Enable
Setting IP Mode	*150*	☑	Enable
Configure IP Address	*152*	☑	Enable
Network Subnet Mask Configure	*153*	☑	Enable
Network Gateway Configure	*156*	☑	Enable
Port Voice Up	*170#	☑	Enable
Port Voice Down	*171#	☑	Enable
Allow Configuration by FXO	*149*	☑	Enable
Reset Basic Configuration	*165*	☑	Enable
Reset Factory Configuration	*166*	☑	Enable
Restart Device	*111#	☑	Enable

Call Function	Codes	Use Default	Status
Call by IP	*47*	☑	Enable
Call Waiting Activate	*51#	☑	Enable
Call Waiting Deactivate	*50#	☑	Enable
Blind Transfer	*87*	☑	Enable
Call Forward Unconditional Activate	*72*	☑	Enable
Call Forward Unconditional Deactivate	*73#	☑	Enable
Call Forward Busy Activate	*90*	☑	Enable
Call Forward Busy Deactivate	*91#	☑	Enable
Call Forward No Reply Activate	*92*	☑	Enable
Call Forward No Reply Deactivate	*93#	☑	Enable

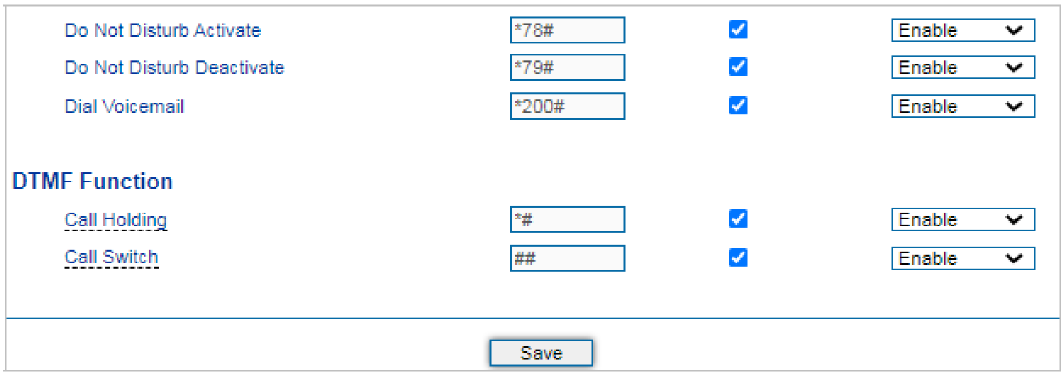


Table 4.10.8 Explanation of Feature Code

Parameter	**Explanation**
Inquiry LAN port IP address	Dial*158# to obtain device's LAN port IP address
Inquiry Phone Number	Dial*114# to obtain port account
Inquiry PortGroup Number	Dial *115# to obtain port group number
Setting IP Mode	*150*0#, means pppmodem, *150*1#, means static IP, *150*2#, means obtain IP address by DHCP, *150*3#, means pppoe.
Configure IP Address	*152*+IP, set gateway IP address
Network subnet mask configure	*153*+subnet mask, set gateway subnet mask
Network Gateway Configure	*156*+gateway IP, set gateway
Reset Basic Configuration	Dial *165*000000# to restore default username/password and network configuration
Reset Factory Configuration	*166*000000#, reset factory
Restart Device	*111#, restart device
Call holding	During a call, dial*# into call hold.（Recovery the call through hook flash or *#）
Call by IP	Directly dial the end user IP to call
Call Waiting Activate	*51#, enable call waiting function
Call Waiting Deactivate	*50#, forbid call waiting function
Blind Transfer	If the call transfer to 801, first hook flash and then dial the * 87 * 801#
Call Forward Unconditional	*72*+ phone number#, transfer the call from the phone

Activate	number
Call Forward Unconditional Deactivate	*73#, forbid call forward unconditional
Call Forward Busy Activate	*90*+ forward busy number#
Call Forward Busy Deactivate	*91#, forbid call forward busy
Call Forward No Reply Activate	*92*+ forward no reply number#
Call Forward No Reply Deactivate	*93#, close this function
Do Not Disturb Activate	*78#, enable DND function
Do Not Disturb Deactivate	*79#, close DND function
Dial Voicemail	*200#, visit voice mail box

4.10.9 System Parameter

System parameters include NTP, daylight saving time, daily reboot time, web parameter, telnet parameter and remote management.

NTP (Network Time Protocol） is a computer time synchronization protocol.

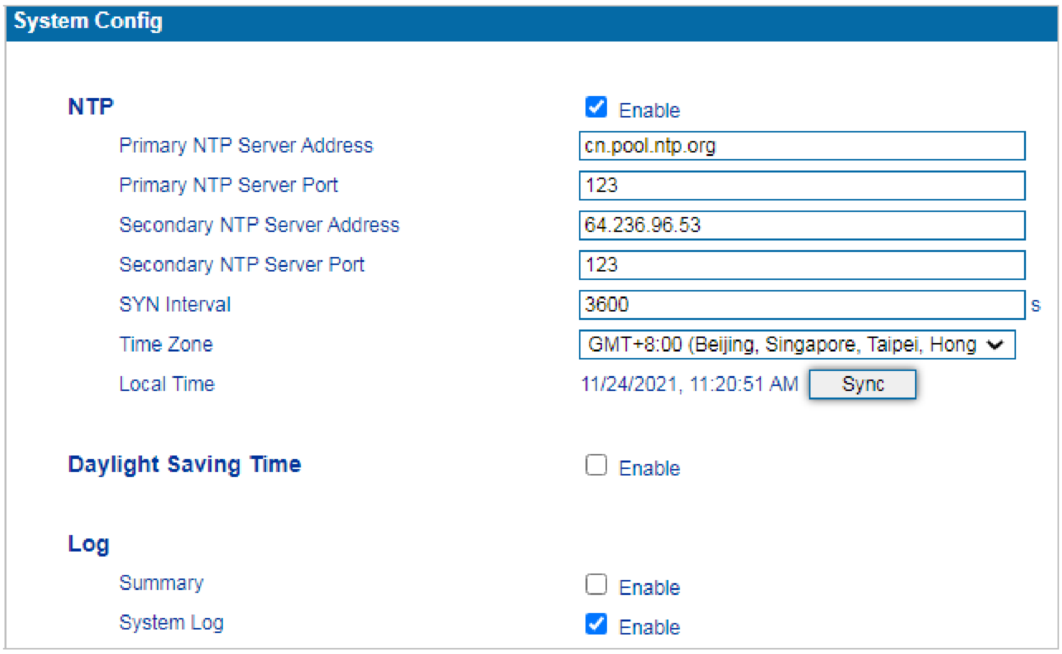


Figure 4-10-9 Configure System Parameters

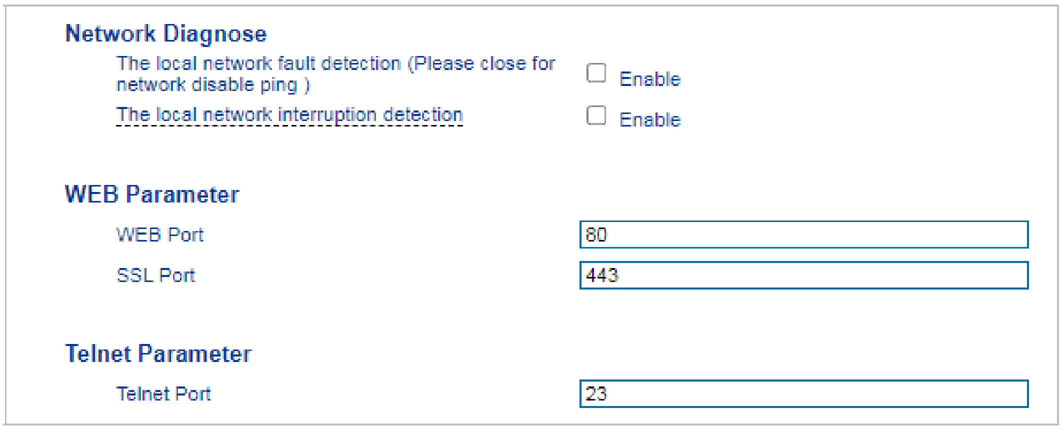


Table 4.10.9 Explanation of System Parameters

Parameter	Explanation
NTP	To enable or disable NTP
Primary NTP server address	The IP address of primary NTP server; default IP address is us.pool.ntp.org.
Primary NTP server port	The service port of primary NTP server; default port is 123.
Secondary NTP server address	The IP address of secondary NTP server ; Default IP address is 64.236.96.53
Secondary NTP server port	The service port of secondary NTP server; Default port is 123
SYN Interval	The interval to synchronize the time of the Gateway-128o. Default value is 3600s.
Time Zone	The time zone of the device; Default configuration is United States central time, Chicago.
Daylight Saving Time	Enable or disable daylight saving time
Summary	Record detailed call logs
System Log	Save key logs when the system is running
The Local Network Fault Detection	When the network is unavailable, report the local network failure
The Local Network Interruption Detection	When the network is disconnected, report the local network interruption
WEB Port	The web port of the device; Default port is 80
SSL Port	The SSL port; Default is 443

Telnet port	Listening port of telnet service; Default port is 23

Note: After Web port and Telnet port are configured, please restart the device for the configurations to take effect.

4.11 Call & Routing

4.11.1 Port Group

When two or more FXO ports need to register with a same SIP account, you can group the ports together and then set an account for the group on the **Call & Routing** → **Port Group** page.

Parameters of port group include registration, primary display name, primary SIP user id, primary authentication ID and password, secondary display name, secondary SIP user id, secondary authentication ID and password, off-hook auto dial, auto dial delay time, port select, etc.

Figure 4-211-1 Add Port Group

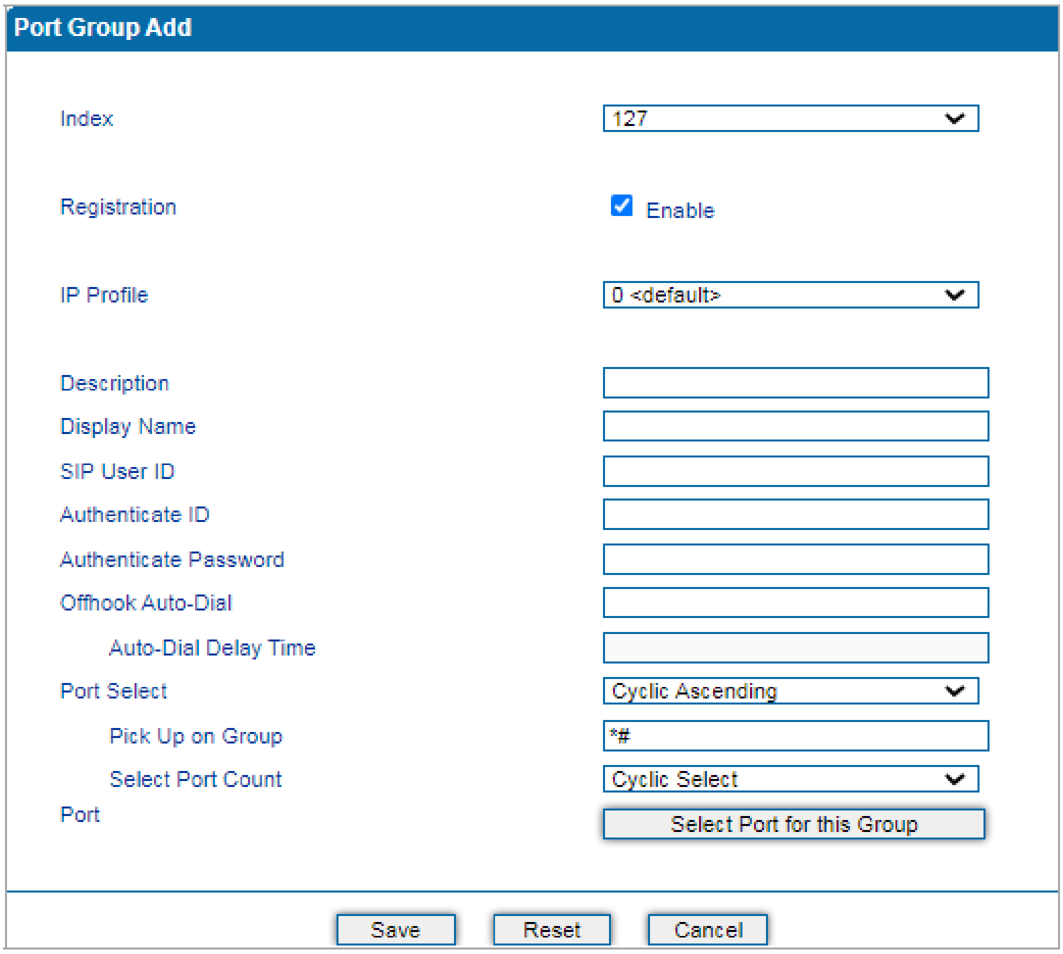


Table 4.11.1 Explanation of Parameter for Port Group

Parameter	Explanation
Index	The NO. of the port group; It uniquely identifies a route.
Description	The description of the port group; it is used to identify the port group.
Display Name	Display name of the port group, which will be used in SIP message, for example: INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDPpc33.atlanta.com;branch=z9hG4bK776asdhds Max-Forwards: 70 To: Bob <sip:bob@biloxi.com> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Here Bob and Alice is the display name
SIP User ID	User ID of this SIP account, which is provided by VoIP service provider (ITSP). It is usually in the form of digit similar to phone number or an actual phone number.
Authenticate ID	SIP service subscriber's ID for authentication; it can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's password for authentication
Off-hook Auto-Dial	An extension or phone number is pre-assigned here so that the number is automatically dialed as soon as user picks up the phone
Auto-dial Delay time	How long auto-dialing will be delayed
Port Select	It specifies the policy for selecting a port for ringing in the port group • Ascending: the device always selects a port from the minimum number. • Cyclic ascending: the device always selects a port from a number next to the number selected last time. If the maximum number was selected last time, the next selected number is the minimum number. The sequence moves in cycles like this. • Descending: the device always selects a port from the maximum number. • Cyclic descending: the device always selects a port from a number next to the number selected last time. If the minimum number was selected last time, the next selected number is the maximum number. The sequence moves in cycles like this. • Group ring: all ports ring at the same time

Port	Select ports for this port group

4.11.2 IP Trunk

A peer-to-peer VoIP call occurs when two VoIP phones communicate directly over IP network without IP PBX between them. IP trunk helps establish peer-to-peer call between gateway and VoIP phones. IP trunk will be used in routing configuration.

Figure 4-11-2 Configure IP Trunk

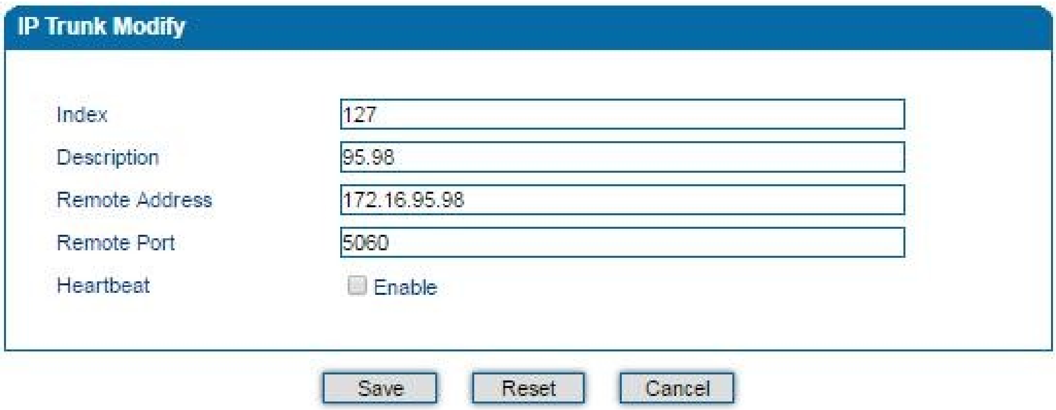


Table 4.11.2 Explanation of IP Trunk Parameters

Parameter	Explanation
Index	The No. of the IP trunk; range is from 0 to 127.
Description	The description of the IP trunk; it is used to n identify the IP trunk.
Remote Address	IP address or domain name of the peer device
Remote Port	SIP port of the peer device
Heartbeat	Whether to enable the 'Heartbeat' function for the IP trunk. Default value is 'not enable'. If heartbeat is enabled, the device will send "OPTION" to the peer device.

4.11.3 Routing Parameter

Routing parameter determines a call is routed before or after manipulation.

Figure 4-311-3 Configure Routing Parameter

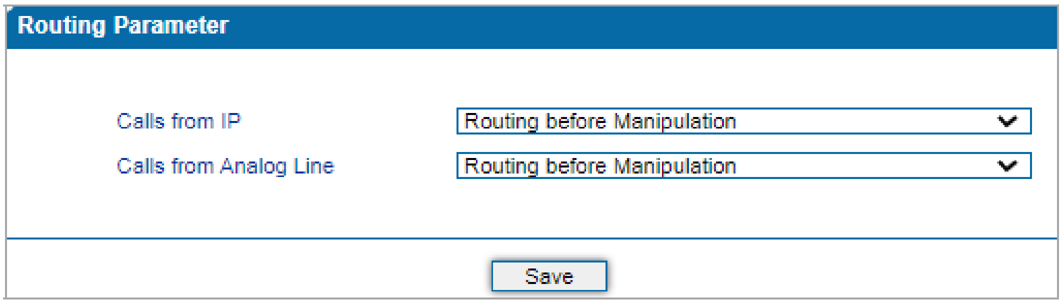


Table 4.11.3 Explanation of Routing Parameters

Parameter	Explanation
Calls from IP	Choose calls from IP network are routed before manipulation or after manipulation.
Calls from Analog Line	Choose calls from analog lines are routed before manipulation or after manipulation.

4.11.4 IP → Tel Routing

Calls from IP network can be routed to FXO port or port group of the FXO GW device through IP →Tel routing.

Figure 4-11-4 Add IP →Tel Route

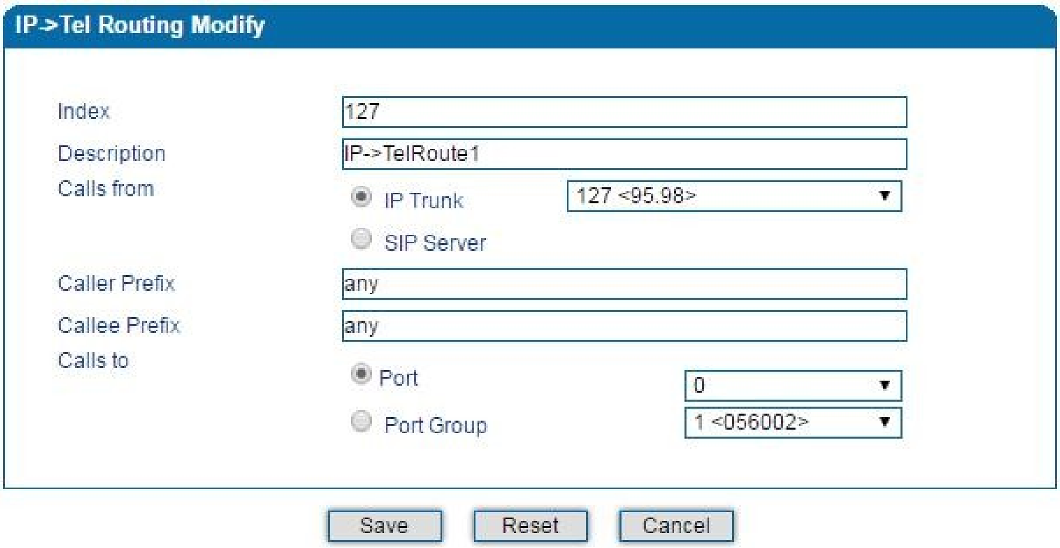


Table 4.11.4 Parameter Explanation of IP →Tel Routes

Parameter	Explanation
Index	Index of the IP →Tel routing; range is from 0 to127; 0 is the highest priority.

Description	Description of the IP →Tel routing; it is used to identify the IP → Tel routing.
Calls from	Choose calls from IP trunk or SIP server; 'any' means any IP addresses.
Caller Prefix	The prefix of the caller number, which helps match routing exactly. Its length is less than or equal to the caller number. For example, if caller number is 2001, the caller prefix can be 200 or 2. 'Any' means the prefix matches any caller number.
Callee Prefix	The prefix of the called number, which helps match routing exactly. Its length is less than or equal to the called number. If the called number is 008675526456659, the called prefix can be 0086755 or 00.,"any" means the prefix matches any called number
Calls to	Which port or port group to which calls are routed.

4.11.5 Tel → IP/Tel Routing

Calls from the FXO port or port group can be routed to IP trunk or ports of SIP server/other device through Tel →IP/Tel routing.

Figure 4-11-5 Add Tel →IP/Tel Route

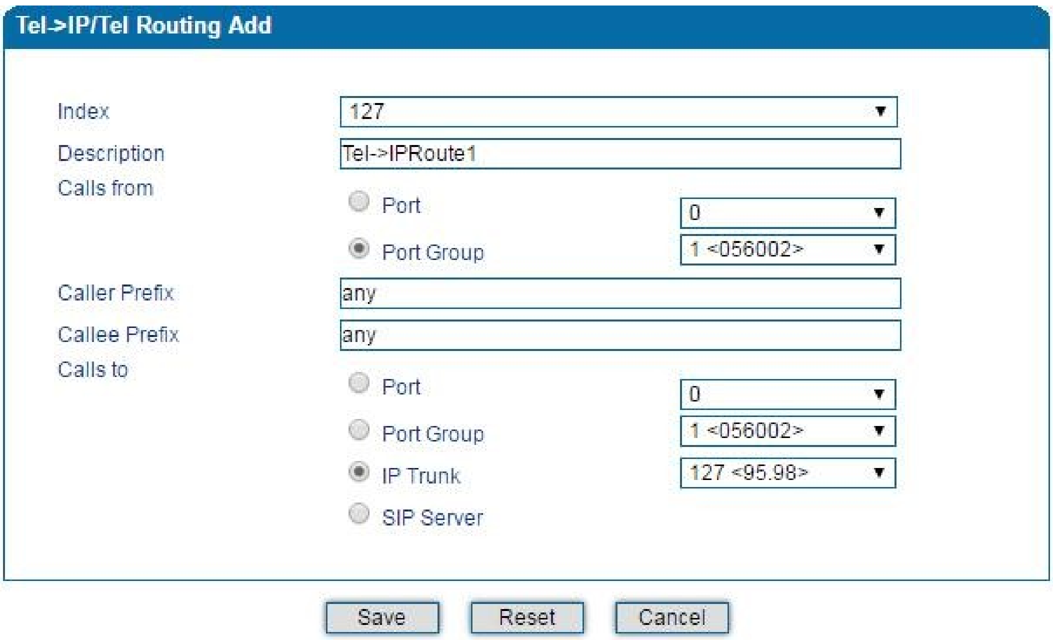


Table 4.11.5 Explanation of Tel →IP/Tel Route

Parameter	Explanation
Index	The index of this Tel →IP/Tel routing; range is from 0 to 127. Each index cannot be used repeatedly. Routing priority: 0 is the highest priority.

Description	The description of this Tel →IP/Tel routing; it is used to identify the routing.
Calls From	Choose calls are from a port or a port group
Caller Prefix	The prefix of the caller number, which helps match routing exactly. Its length is less than or equal to the caller number. For example, if caller number is 2001, the caller prefix can be 200 or 2. 'any' means the prefix matches any caller number.
Callee Prefix	The prefix of the called number, which helps match routing exactly. Its length is less than or equal to the called number. If the called number is 008675526456659, the called prefix can be 0086755 or 00. 'any' means the prefix matches any called number.
Calls to	Choose calls are routed to a port, port group, IP trunk or SIP server

4.11.6 Call Limit

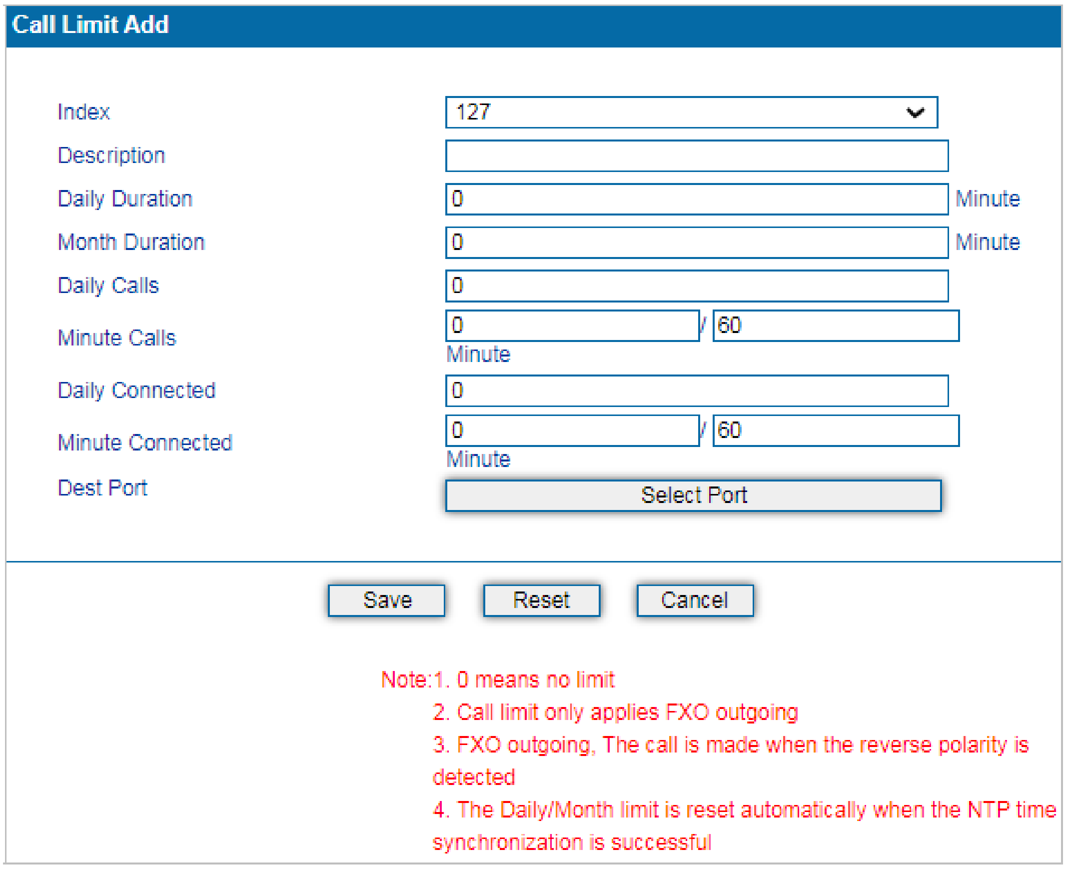


Table 4.11.6 Explanation of call limit

Parameter	Explanation
Index	The index of call limit
Description	The description of this call limit; it is used to identify the limiting.
Daily Duration	The maximum duration of a daily call
Month Duration	The maximum duration of a monthly call
Daily Calls	The times of daily calls
Minute Calls	The times of a minute calls
Daily Connected	The times of daily connected calls
Minute Connected	The times of mi connected calls The times of calls made in minute
Dest Port	Select the port that needs to be call limit

Note:

1) 0 means no limit
2) The call limit only affects the outgoing call from the FXO port
3) The day/month limit will be automatically reset when the NTP time synchronization is successful.

4.12 Manipulation

Number manipulation refers to the change of a called number or a caller number during calling process when the called number or the caller number matches the preset rules.

4.12.1 IP → Tel Called

On the **IP → Tel Called** page, you can set rules for manipulating the called number of IP → Tel calls.

Figure 4-12-1 Add IP →Tel Called Number Manipulation

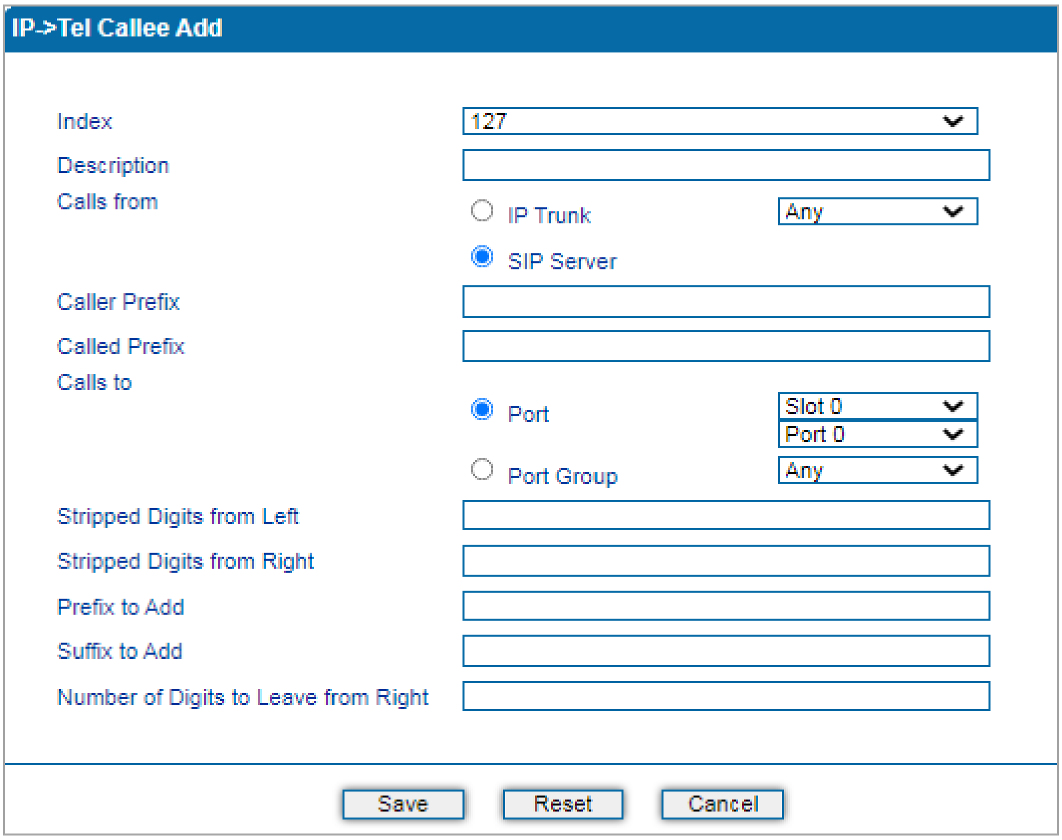


Table 4.12.1 Explanation of Parameters for IP →Tel Called Number Manipulation

Parameter	Explanation
Index	The index of this manipulation; range is from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority
Description	Description of this manipulation; it is used to identify this manipulation.
Calls From	Determine the calls come from IP trunk or SIP server
Caller Prefix	Set a prefix for caller number. The prefix's length is less than or equal to that of the caller number, which helps to match the caller number of this call. If caller number is 2001, the caller prefix can be 200 or 2. "any" means match any caller number.
Called Prefix	Set a prefix for called number. The prefix's length is less than or equal to called number, which helps to match the called number. If called number is 008675526456659, the called prefix can be 0086755 or 00. "any" means match any called number.
Calls to	Determine the call is routed to a port or a port group.
Stripped Digits from Left	The number of digits which are lessened from the left of the called number

Stripped Digits from Right	The number of digits which are lessened from the right of the called number
Prefix to Add	The prefix added to the called number after its digits are lessened.
Suffix to Add	The suffix added to the called number after its digits are lessened.

4.12.2 Tel → IP/Tel Caller

On the **Tel → IP/Tel Caller** page, you can set rules for manipulating the caller number of Tel → IP/Tel calls.

Figure 4-12-2 Add Tel → IP/Tel Caller Number Manipulation

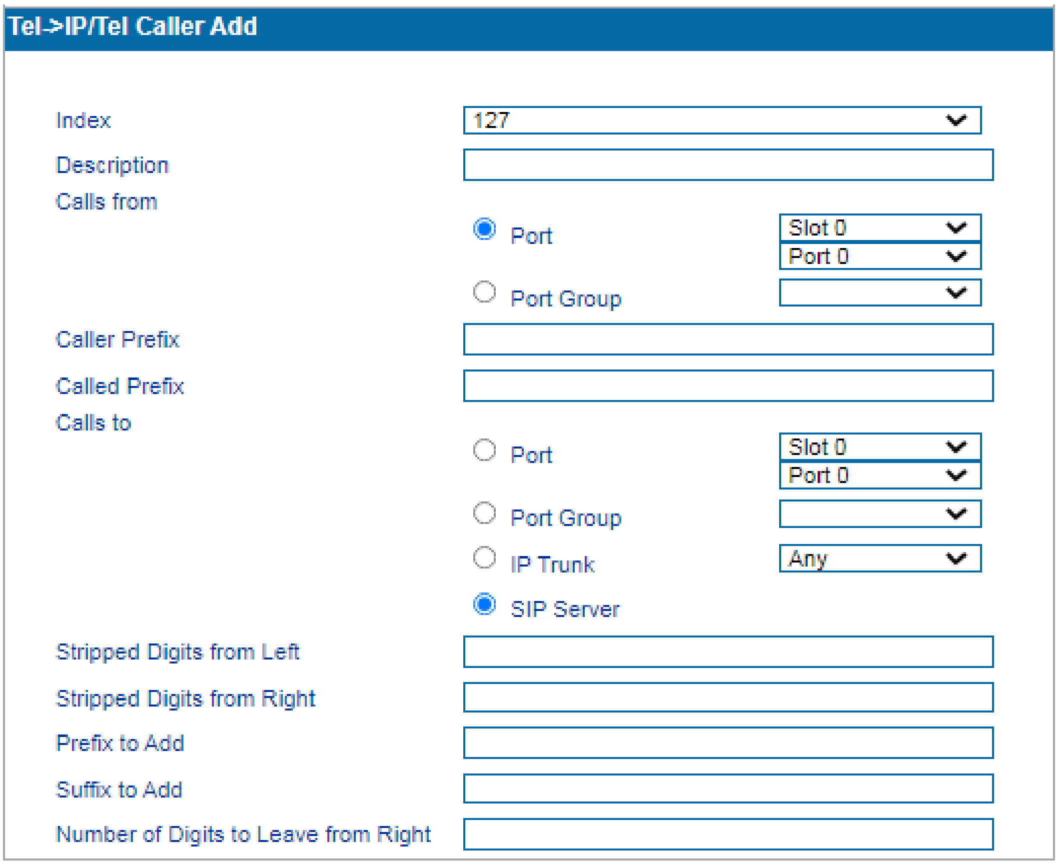


Table 4.12.2 Explanation of Parameters for IP → Tel Called Number Manipulation

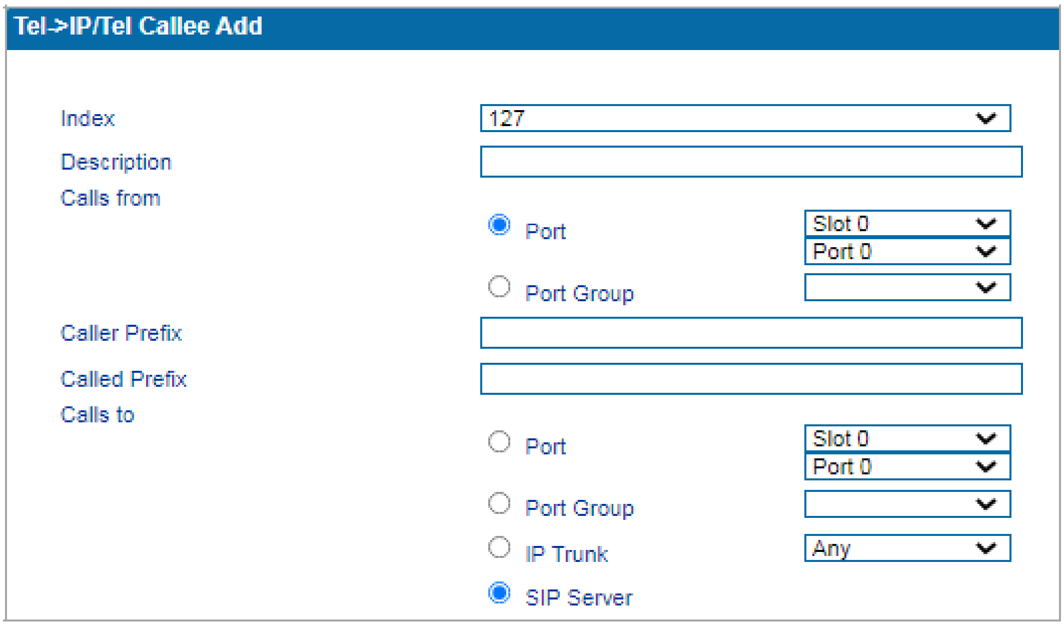
Parameter	Explanation
Index	The index of this manipulation; range is from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority
Description	Description of this manipulation; it is used to identify this manipulation.
Calls From	Determine the calls come from a port or a port group.

Caller Prefix	Set a prefix for caller number. The prefix’s length is less than or equal to that of the caller number, which helps to match the caller number of this call. If caller number is 2001, the caller prefix can be 200 or 2. ‘any’ means match any caller number.
Called Prefix	Set a prefix for called number. The prefix’s length is less than or equal to called number, which helps to match the called number. If called number is 008675526456659, the called prefix can be 0086755 or 00. ‘any’ means match any called number.
Calls to	Determine the call is routed to a port, a port group, an IP trunk or a SIP server.
Stripped Digits from Left	The number of digits which are lessened from the left of the caller number
Stripped Digits from Right	The number of digits which are lessened from the right of the caller number
Prefix to Add	The prefix added to the caller number after its digits are lessened.
Suffix to Add	The suffix added to the caller number after its digits are lessened.

4.12.3 Tel → IP/Tel Called

On the **Tel → IP/Tel Called** page, you can set rules for manipulating the called number of Tel → IP/Tel calls.

Figure 4-12-3 Add Tel → IP/Tel Called Number Manipulation



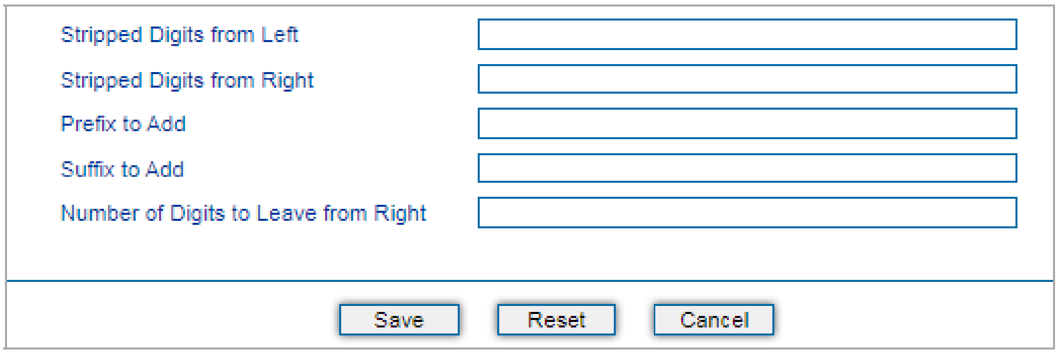


Table 4.12.3 Explanation of Parameters for Tel → IP/Tel Callee Number Manipulation

Parameter	Explanation
Index	The index of this manipulation; range is from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority
Description	Description of this manipulation; it is used to identify this manipulation.
Calls From	Determine the calls come from a port or a port group.
Caller Prefix	Set a prefix for caller number. The prefix's length is less than or equal to that of the caller number, which helps to match the caller number of this call. If caller number is 2001, the caller prefix can be 200 or 2. 'any' means match any caller number.
Callee Prefix	Set a prefix for called number. The prefix's length is less than or equal to called number, which helps to match the called number. If called number is 008675526456659, the called prefix can be 0086755 or 00. 'any' means match any called number.
Calls to	Determine the call is routed to a port, a port group, an IP trunk or a SIP server.
Stripped Digits from Left	The number of digits which are lessened from the left of the called number
Stripped Digits from Right	The number of digits which are lessened from the right of the called number
Prefix to Add	The prefix added to the called number after its digits are lessened.
Suffix to Add	The suffix added to the called number after its digits are lessened.

4.13 Management

4.13.1 TR069

TR069 is short for Technical Report 069, which provides a commonly-used framework and protocol for next-generation network devices. As an application-level protocol on top of IP TR069 has no limitation to access ways of network devices.

Under the network management model of TR069, ACS (Auto-Configuration Server) works as a management server, responsible for managing CPEs (Custom Premise Equipment).

ACS URL (auto-configuration server URL address) is provided by service provider. The ACS URL generally starts with http:// or https://

Username and password are used for ACS authentication.

Figure 4-13-1 Configure TR069 Parameter

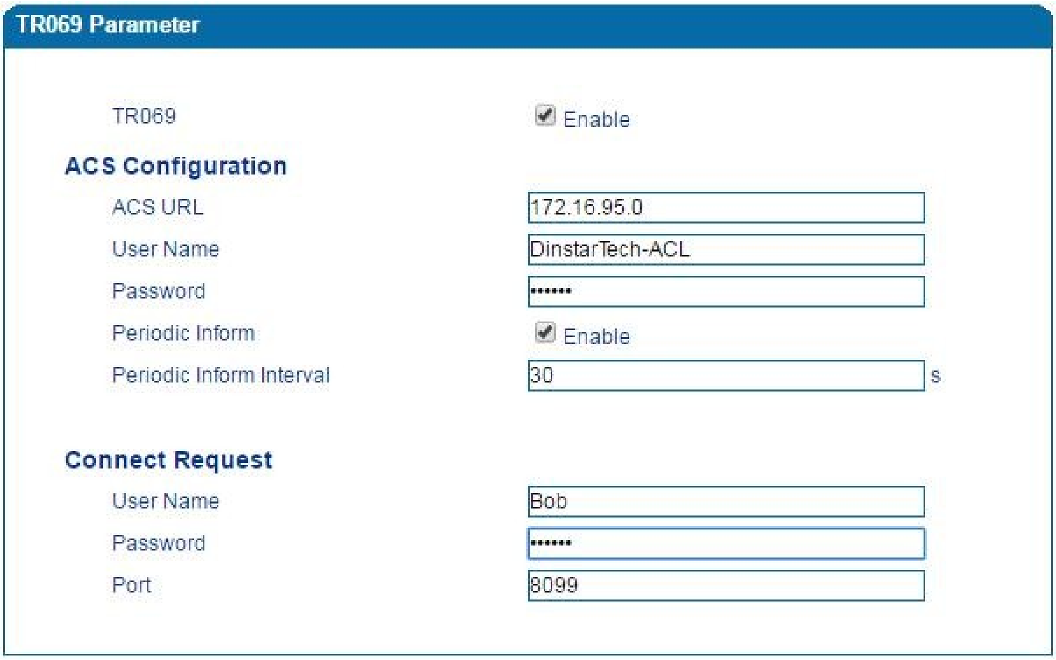


Table 4.13.1 Explanation of TR069 Parameters

Parameter	Explanation
TR069	Choose whether to enable TR069; it is 'not enable' by default.
ACS URL	The IP address or domain name of ACS, which is provided by service provider.
Username(ACS)	Username of ACS, which is provided by service provider.
Password(ACS)	Password of ACS, which is provided by service provider.
Periodic Inform	Choose whether to enable 'Periodic Inform'; if it is enabled, ACS will connect to CPE every 30 seconds (if the interval is set as 30 seconds).
Periodic Inform	The interval set for periodic connection between ACS and CPE.

Interval	
Username (CPE)	Username of CPE
Password (CPE)	Password of CPE
Port	The port to connect CPE and ACS

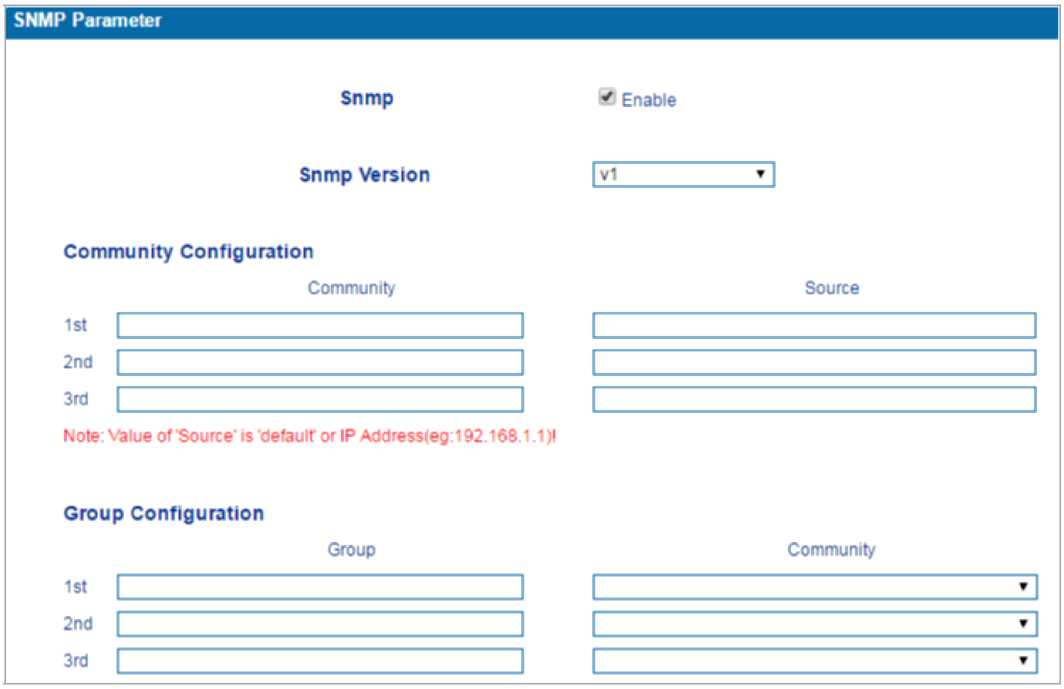
4.13.2 SNMP

SNMP (Simple Network Management Protocol) is an Internet-standard protocol for collecting and organizing information about managed devices on IP networks and for modifying that information to change device behavior. Devices that typically support SNMP include routers, switches, servers, workstations, printers, modem racks and more.

SNMP is widely used in network management for network monitoring. SNMP exposes management data in the form of variables on the managed systems organized in a management information base which describe the system status and configuration. These variables can then be remotely queried (and, in some circumstances, manipulated) by managing applications.

Three significant versions of SNMP have been released. SNMPv1 is the original version of the protocol. More recent versions, SNMPv2c and SNMPv3, feature improvements in performance, flexibility and security.

Figure 4-13-2 Configure SNMP Parameters



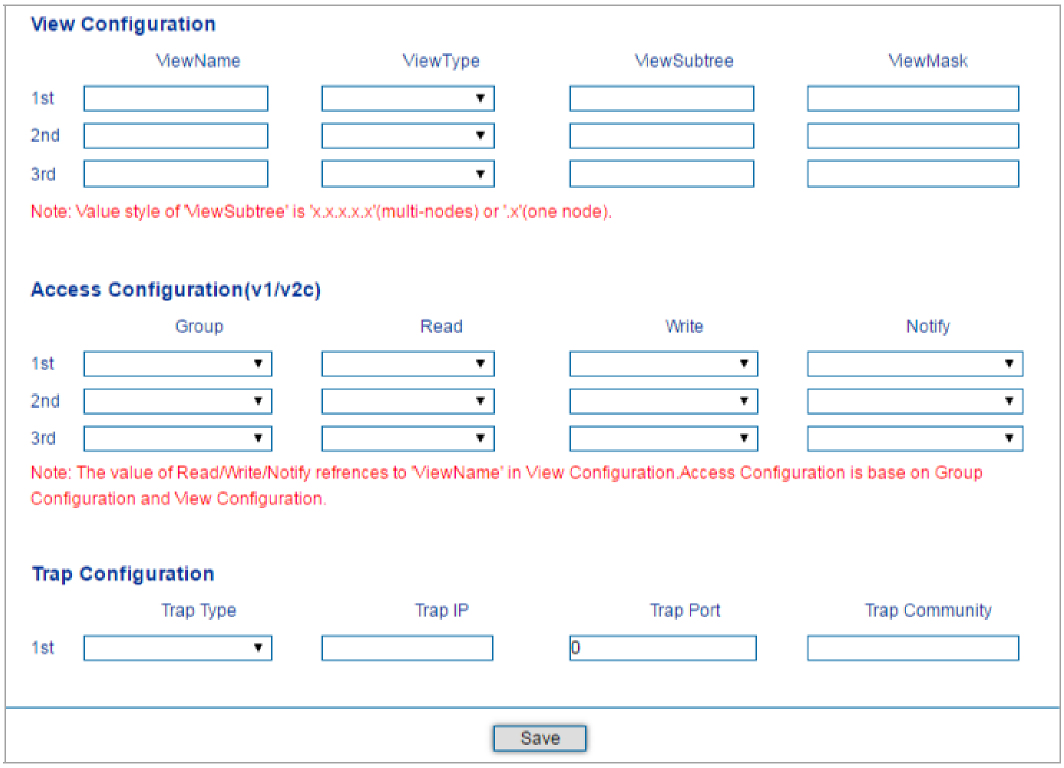


Table 4.13.2 Explanation of SNMP Parameters

Parameter	**Explanation**
SNMP	The FXO GW device supports three versions of SNMP, namelyV1、V2C and V3.
Community Configuration	Community configuration exists in V1 and V2C. **Community:** fill in a community name used to read through SNMP protocol; it is a character string. **Source:** The IP address of SNMP server. SNMP server cannot identify the packets sent from FXO GW unless the community configured in FXO GW matches with the community configured in SNMP server.
Group Configuration	Group configuration exists in V1 and V2C and V3. **Group:** fill in a group name which is used to identify the group; it's a character string. **Community:** fill in a community which means this community has joined in the group. In the following, access permission of read, write and notify is configured for each group.

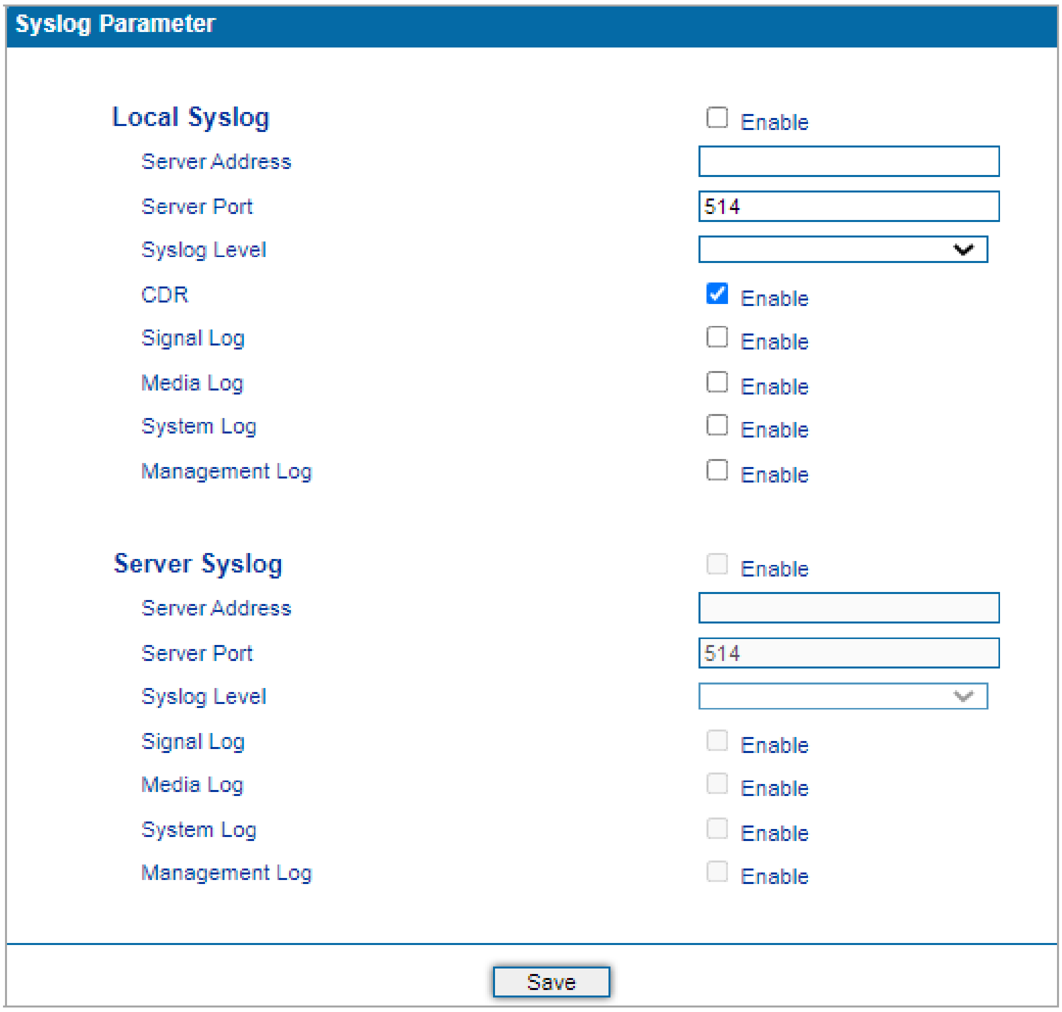
View Configuration	View configuration exists in V1, V2C and V3. **ViewName:** fill in a view name which is used to identify this view. **ViewType:** choose 'Included' or 'Excluded'. 'Included' means the view includes the OID of the corresponding ViewSubtree, while 'Excluded' means the OID of the corresponding ViewSubtree is excluded from this view. **ViewSubtree:** fill in the OID of the view subtree. **ViewMask:** it is used to withdraw a row of a table, such as an Ethernet port.
Access Configuration	Access configuration exists in V1, V2C and V3, under which permission of read, write or notify is configured for a community group. **Group:** choose a group name that has been configured. **Read:** Choose a 'read' view for the group. **Write:** Choose a 'write' view for the group. **Notify:** Choose a 'notify' view for the group.
Trap Configuration	Trap configuration exists in V1, V2C and V3, which is aimed to send trap alarm. **Trap Type:** Choose V1, V2C and Inform. **Trap IP:** the IP address of the destination SNMP server where trap alarm is sent. **Trap Port:** the port of the destination SNMP server, which will receive trap alarm. **Trap Community:** the community configured in the destination SNMP server.
User Configuration	User configuration exists in V3. When V3 transmits SNMP packets in an encryption way, this item needs to be configured. **User:** fill in a user name used to authenticate. **AuthType:** choose MD5 or SHA as authentication type. **AuthPassword:** the password used to authenticate. **Privacy Type:** Choose DES, AES or AES 128 as encryption type. **Privacy Password:** the encryption password.

4.13.3 Syslog

Syslog is a standard for message logging. It allows separation of the software that generates messages, the system that stores messages, and the software that reports and analyzes messages. It also provides a means to notify administrators of problems or performance.

Syslog levels include: EMERG, ALERT, CRIT, ERROR, WARNING, NOTICE, INFO and DEBUG.

Figure 4-13-3 Configure Syslog Parameters



When the FXO GW device registers to SIM Cloud server, local syslog will be changed to non-configurable and all logs will be stored on the Cloud server.

4.13.4 Provision

Provision is used to make the FXO GW device automatically upgrade with the latest firmware stored on an http server, an ftp server or a tftp server. Please refer to the Instruction for Using Provision.

Figure 4-13-4 Provision

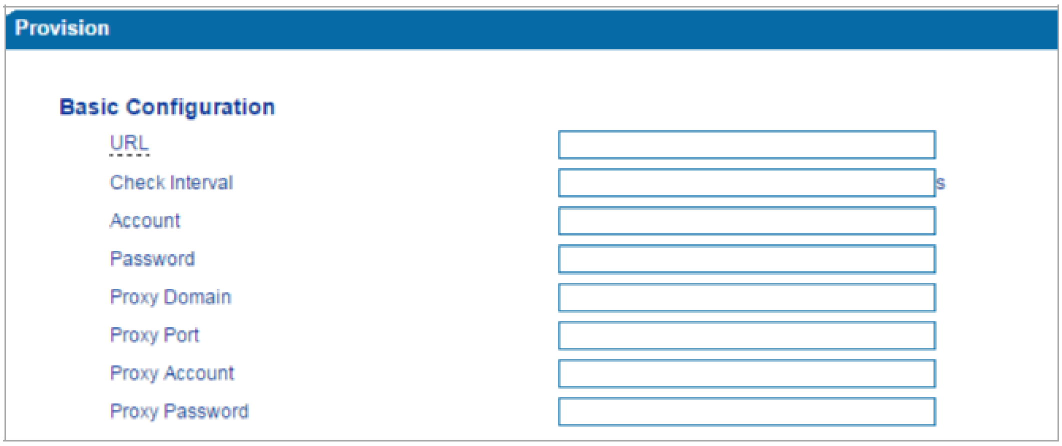


Table 4.13.4 Explanation of Provision Parameters

Parameter	**Explanation**
URL	URL of provisioning server, support HTTP, TFTP, FTP
Check Interval	The interval to check whether there is new firmware version on the provisioning server
Account	Account for logging in provisioning server
Password	Password for logging in provisioning server

4.13.5 Cloud server

You can register the FXO GW device to cloud server, and then the device can be managed by the cloud server.

Figure 4-13-5 Configure Cloud Server

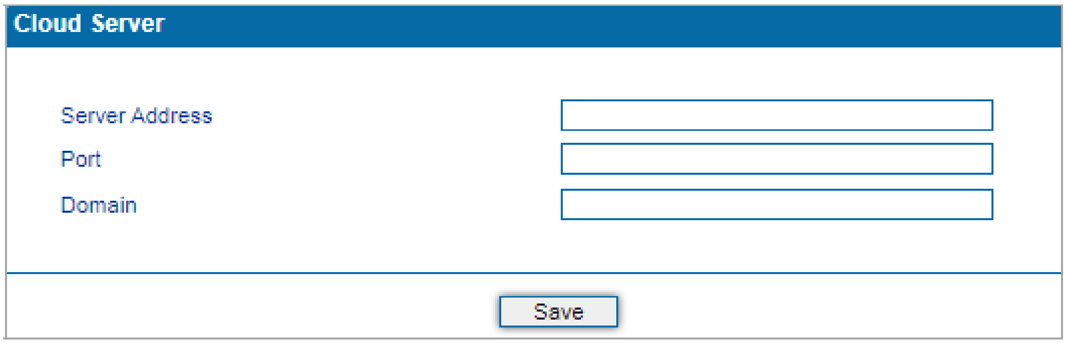


Table 4.13.5 Explanation of Parameters for Cloud Server

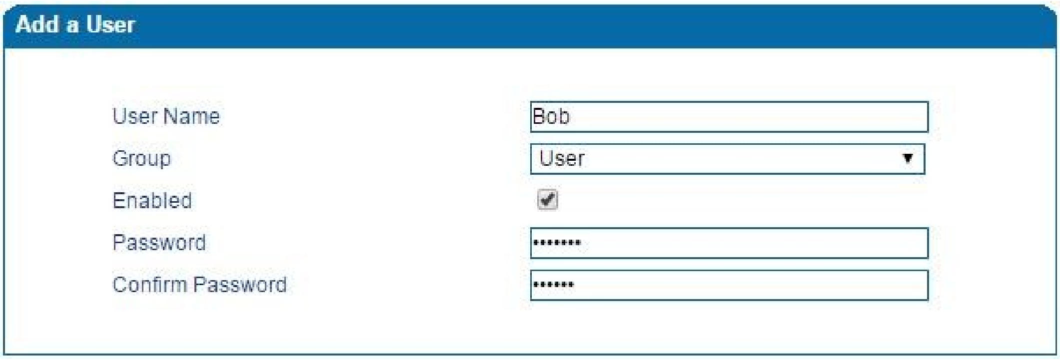
Parameter	**Explanation**

Server Address	The IP address of the cloud server
Port	The listening port of the cloud server
Domain	The domain name of the cloud server
Join the remote management system	Choose whether to join the remote management system of the cloud server.

4.13.6 User Manage

On the **Management** → **User Manage** page, the administrator of the FXO GW device can classify users in different groups, and set login username and password for each user.

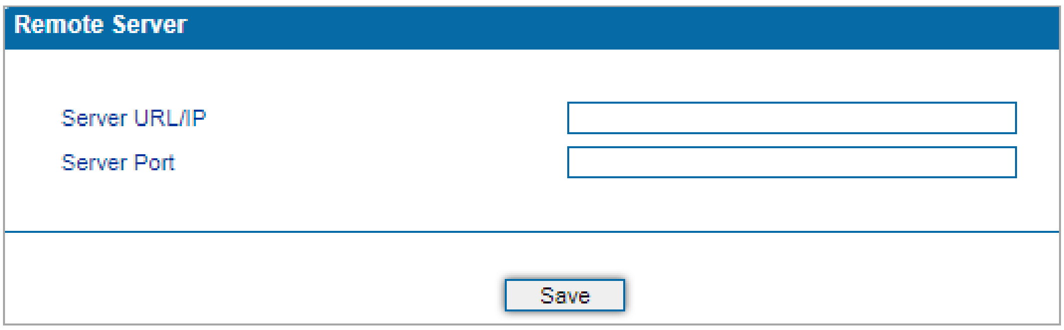
Figure 4-13-6 Modify Username and Password



4.13.7 Remote Server

In case that you need remote technical support, technical support engineers can connect your device with a service server on the **Management** →**Remote Server** page, so as to better help you to solve problems.

Figure 4-13-7 Configure Remote Server



4.13.8 Record Parameter

After configuring the record server, you can upload the call voice of the FXO port of the device to the record server. The record configuration page is shown in Figure 4-13-8:

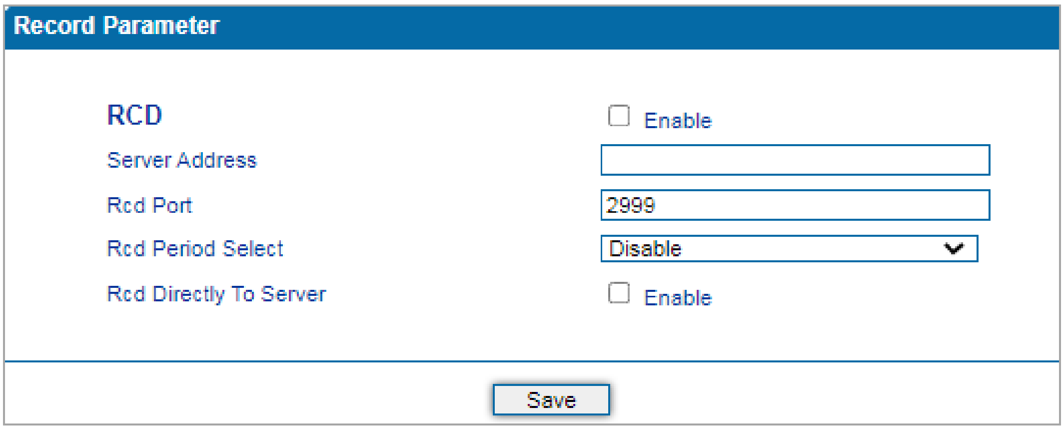


Table 4.13.8 Explanation of Record Parameters

Parameter	Explanation
RCD	Enable or disable the record function
Server Address	Set the record server address, IP address or domain name
Rcd Port	Recording server port (The default is 2999)
Rcd Period Select	Only record within the set time range, support 3 recording periods
Rcd Directly To Server	In the NAT environment, the recording can be directly sent to the public network

4.13.9 Radius Parameter

After the Radius server is configured, you can log into the gateway after successful Radius authentication. The Radius server configuration page is shown in Figure 4-13-9:

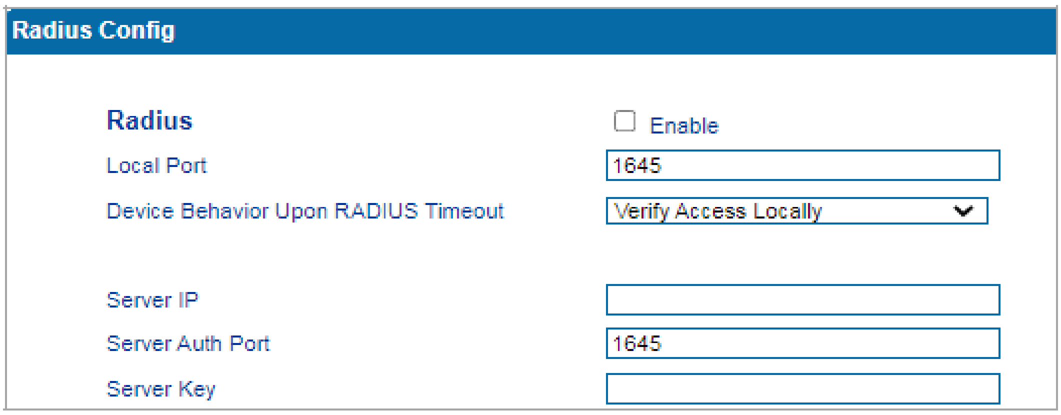


Table 4.13.9 Explanation of Parameters for Radius Config

Parameter	Explanation
Radius	Enable or disable Radius

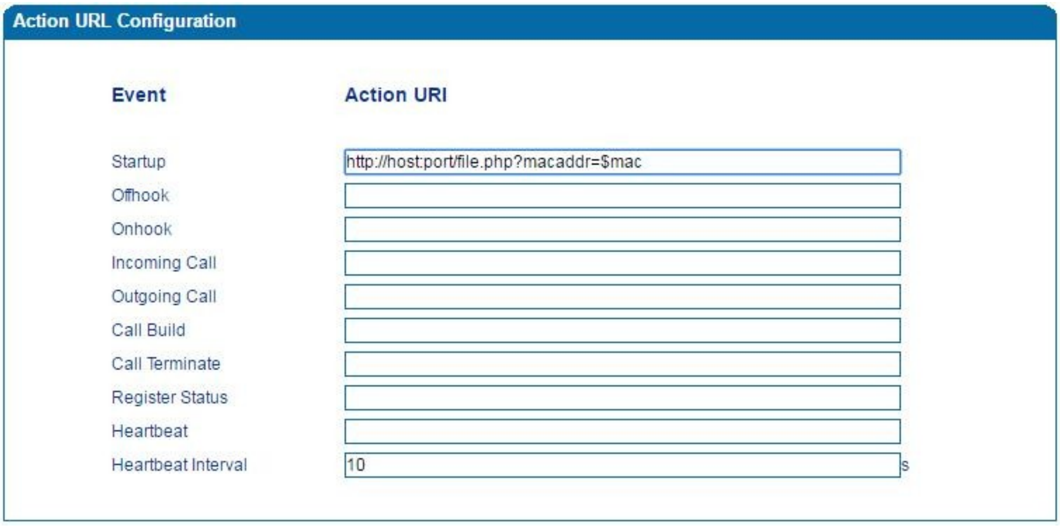
Local Port	The port of the local Radius client
Device Behavior Upon RADIUS Timeout	Processing after Radius authentication timeout. Verify Access Locally : After the timeout, when verifying the user name and password of the local Web login is successful, the login is successful; Deny Access : No matter what, the login is refused
Server IP	The IP address of the Radius server
Server Auth Port	The port of the Radius server
Server Key	The key of the Radius server

4.13.10 Action URL

Action URL is a means of allowing VoIP platform/VoIP server to learn about the statuses of the FXO GW device. This is realized by GET request over the HTTP protocol. During the transmission of status, some data (such as device ID, mac address, called/caller number, IP address) carried in GET request can also be reported to VoIP platform/VoIP server.

The data that can be carried in GET request, please refer to the notes on the **Management → Action URL** page.

Figure 4-13-10 Configure Action URL



Event: Statuses of FXO GW device, which will be reported to VoIP platform/VoIP server.

Action URL: for example, http://host:port/file.php?macaddr=$mac, **among** which 'host' means the HTTP server's IP address or domain name, 'port' means the http server's listening port, 'file.php' means the script that will process this request, and '$mac' means the parameter carried in the request when this request is sent out.

Heartbeat: heartbeat packets are sent to URL by the FXO GW device, used to examine the connection between the FXO GW device and HTTP/HTTP server.

4.13.11 SIP PNP

The gateway can restore the profile and upgrade the software version through SIP PNP. The SIP PNP process is as follows:

1) The gateway sends request packet for SIP subscription to the multicast at intervals

2) The gateway receives the Notify message and reads the URL address of the deployment server in the message

3) Initiate the Provision to the URL to restore the profile or upgrade the software version

Figure 4-13-11 Configure SIP PNP

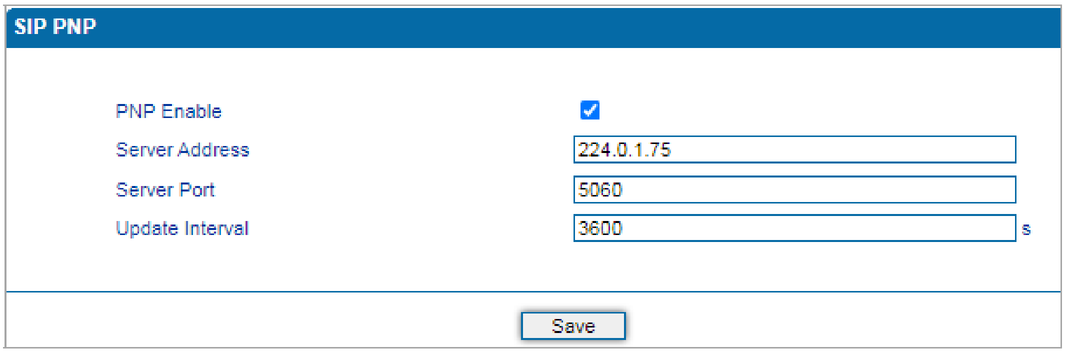
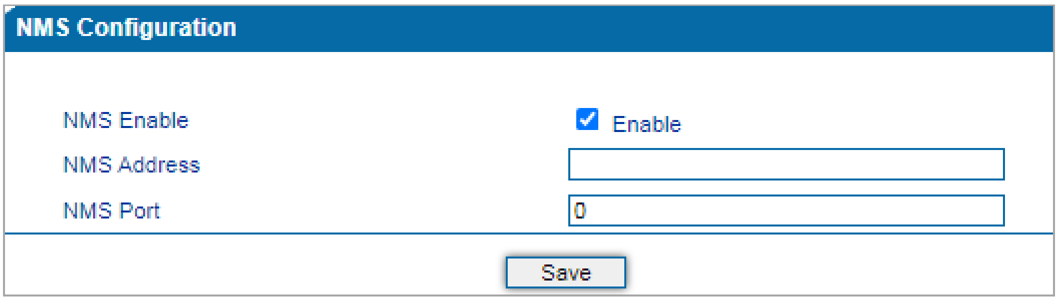


Table 4.13.11 Explanation of Parameters for SIP PNP

Parameter	Explanation
PNP Enable	Enable or disable PNP
Server Address	SIP PNP server IP address, and the default is the multicast address 224.0.1.75
Server Port	SIP PNP server port, and the default is 5060
Update Interval	Send subscription messages periodically, and 3600s by default

4.13.12 NMS Configuration

NMS is a public/private cloud-based device management platform. After users register with the NMS, you can perform batch upgrades, status monitoring, and view alarm information for the devices. To apply for cloud management server(NMS) , please contact your local distributor.

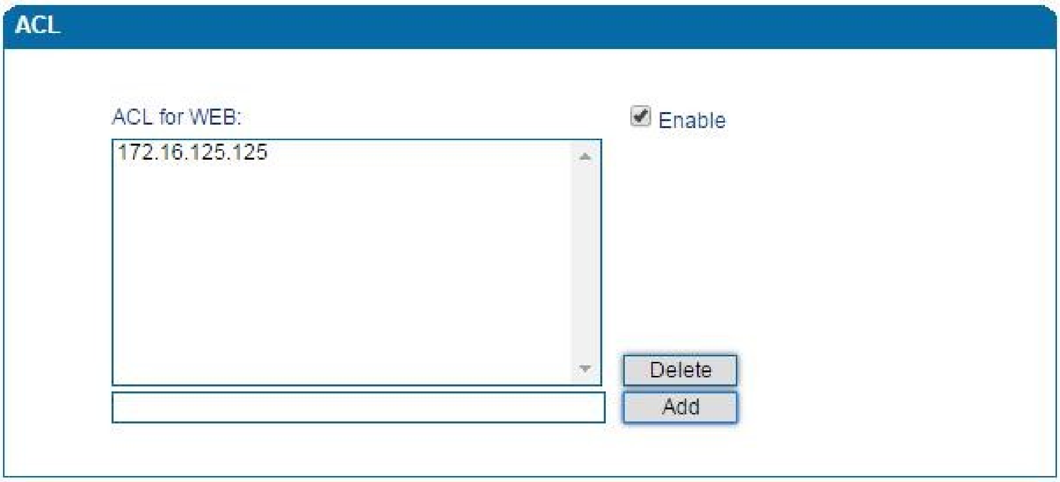


4.14 Security

4.14.1 WEB ACL

ACL (Access Control List) for Web is used to configure IP addresses that are allowed to access the Web Interface of the FXO GW device. The IP address list can't be null once ACL is enabled.

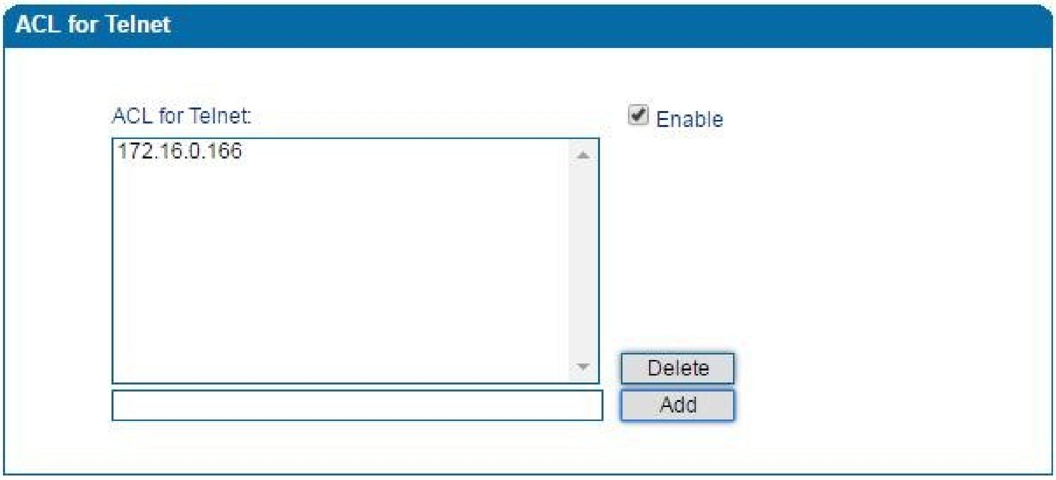
Figure 4-14-1 Add IP Address to Web ACL



4.14.2 Telnet ACL

ACL (Access Control List) for Telnet is used to configure IP addresses that are allowed to access the Telnet Interface of the FXO GW device. The IP address list can't be null once ACL is enabled.

Figure 4-14-2Add IP Address to Telnet ACL

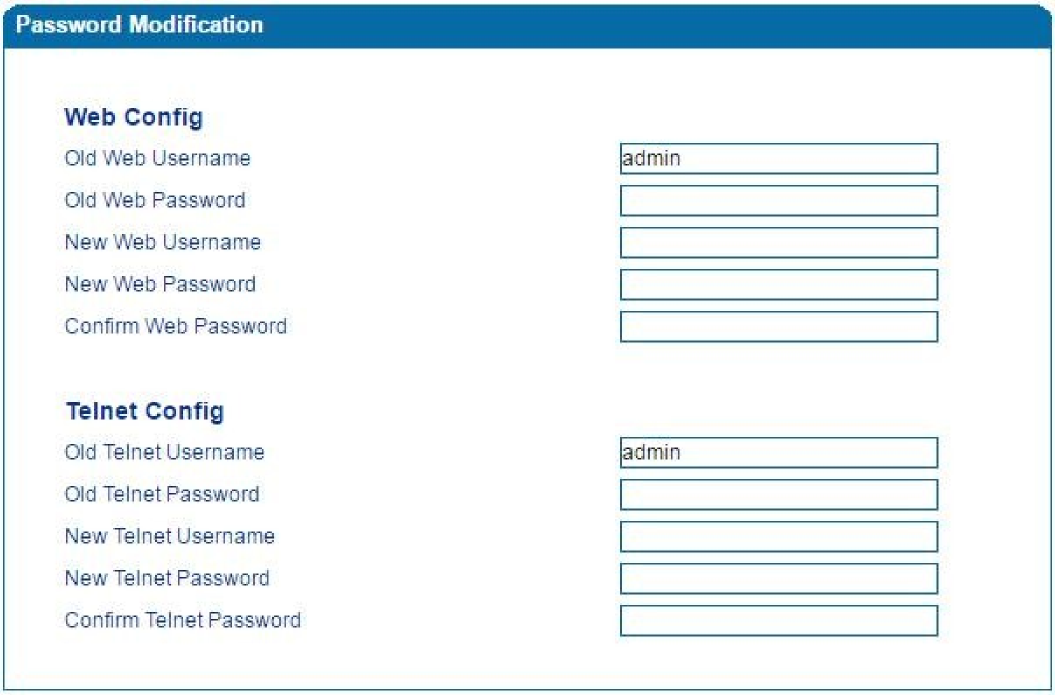


4.14.3 Passwords

You can configure or modify the username and password for logging in the Web interface and the Telnet interface of the FXO GW device on this page.

Note：Both the username and password of Web and Telnet are 'admin' and 'admin' by default. It is advised to modify them for security consideration.

Figure 4-14-3 Modify Username and Password

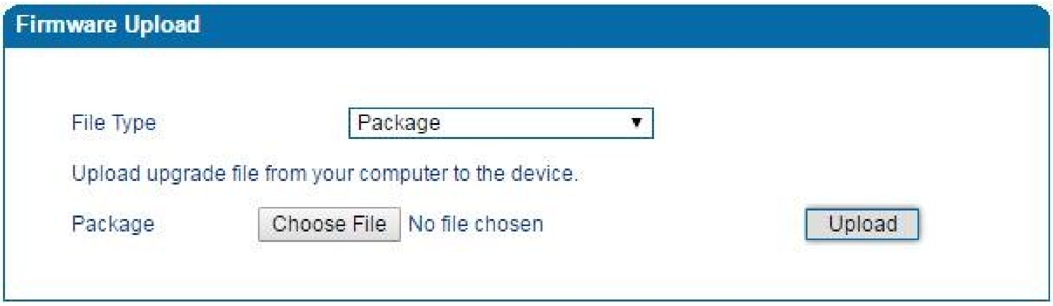


4.15 Tools

4.15.1 Firmware Upload

On the T**ools** → **Firmware** Upload page, you can upload a new firmware version from a local folder.

Figure 4-15-1 Upload Firmware



Steps of Firmware Uploading:

Step 1. Check the current firmware version on the Status & Statistics →System Information page.

Step 2. Prepare firmware package.

Step 3. Upload firmware, select the package from a specific folder on the computer and click the Upload button.

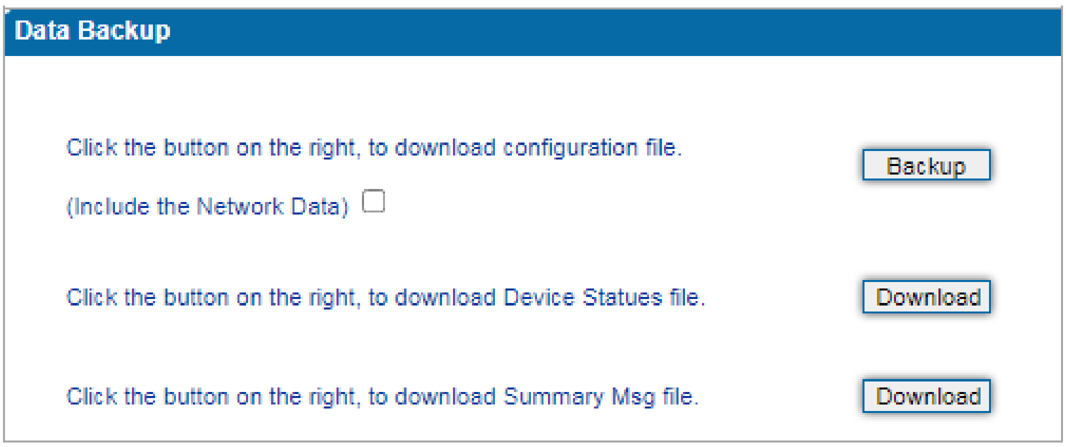
Step 4. Keep waiting until it prompts 'Software loaded successfully!'

Step 5. Reboot the device on the Tools → Device Restart page.

4.15.2 Data Backup

On the **Tools** → **Data Backup** page, you can download and backup configuration data, device status and summary messages on local computer.

Figure 4-15-2 Backup Data



4.15.3 Data Restore

On the **Tools** → **Data Restore** page, you can restore configuration data through uploading a data file from local computer. The restored configurations will take effect after the device is restarted.

Figure 4-15-3 Restore Data

4.15.4 FXO Test

FXO test consists of two parts: Impedance Test and Auto-detect Busy Tone.

- **Impedance Test**

The impedance test of FXO port is mainly for the technicians to perform impedance matching, and can only test the ports in the online state.

Figure 4-15-4.1 Impedance Test

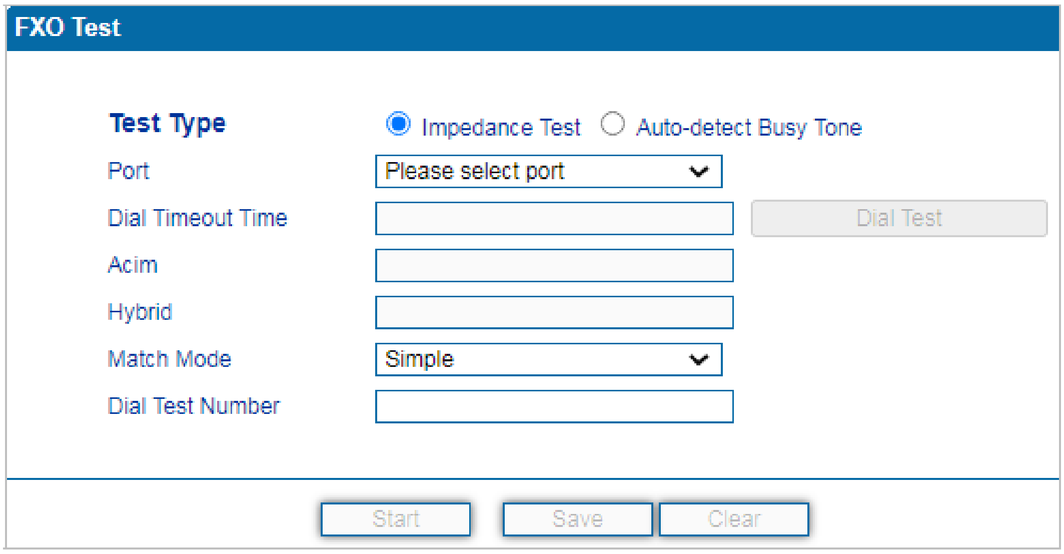


Table 4.15.4-1 Explanation of Parameters for Impedance Test

Parameter	Explanation
Test Type	Choose a type to test
Port	Choose a port to test

Dial Timeout Time	Set the dialing timeout time. If you are not sure, you can also perform a "Dial Test" first (see step 2 for details)
Acim	Display the current impedance value of the FXO port (displayed value, cannot be modified)
Hybrid	Display the current hybrid parameters of the FXO port (displayed value, cannot be modified)
Match Mode	Matching mode: Simple, Standard and Exact (The higher the mode, the higher the accuracy and the longer it takes).
Dial Test Number	Fill in the test number

Steps of impedance test:

1) Navigate to Tools> FXO Test> Impedance Test
2) Fill in the dial timeout time (if you don't know the dial timeout time, you can perform the dial timeout test first (about 10 seconds), after selecting the online port to be tested, click "Dial test", and the timeout time will be displayed after the test is completed)
3) Select the matching mode, test port, and test number, etc., and click "Start" (different modes, time and accuracy are also different, the simple mode is about 15 minutes, the standard is about 30 minutes, and the exact is about 45 minutes);
4) After the test is completed, the Acim and Hybrid values will be displayed.

Note:

1) The dial test number can be configured by itself, but it cannot be the same as the service number.
2) If you do not click to save the result, after restarting, the dialing timeout time, dialing test number and impedance value will be invalid.
3) Please do not leave this page before the test is completed to avoid errors.

Auto-detect Busy Tone

Busy tone detection can only select the online port . The testing steps are as follows:

Figure 4-15-4.2 Auto-detect Busy Tone

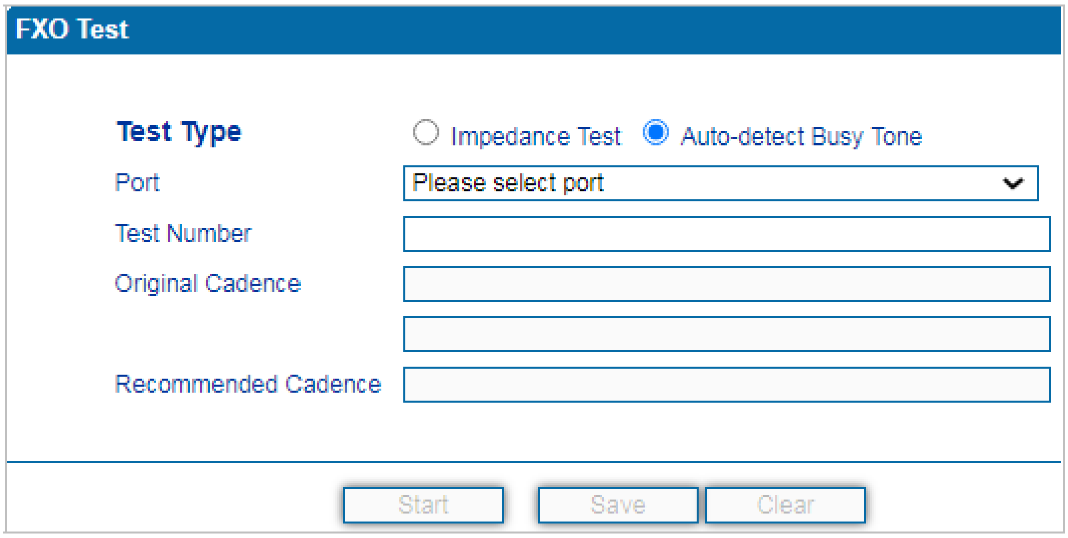


Table 4.15.4-2 Explanation of Parameters for Auto-detect Busy Tone

Parameter	Explanation
Test Type	Choose a type to test
Port	Choose a port to test
Test Number	The destination number for busy tone detection (see step 2 for details)
Original Cadence	The original busy tone cadence captured during the detection
Recommended Cadence	Recommended busy tone cadence after detection

Steps of Auto-detect Busy Tone:

1) Navigate to Tools > FXO Test > Auto-detect Busy Tone

2) Select the online port to be tested and fill in the test number (Make sure that the busy tone service has opened for this number. Its advised to use a PSTN line to connect telephone for test. If this parameter is null, it means no number is dialed)

3) Click ‘Start’, it will take about 1 minute, please do not leave this page

4) After the test is completed, the original cadence and recommended cadence are displayed, Please save the result after finishing, otherwise you can clear the results and retest.

4.15.5 Ping Test

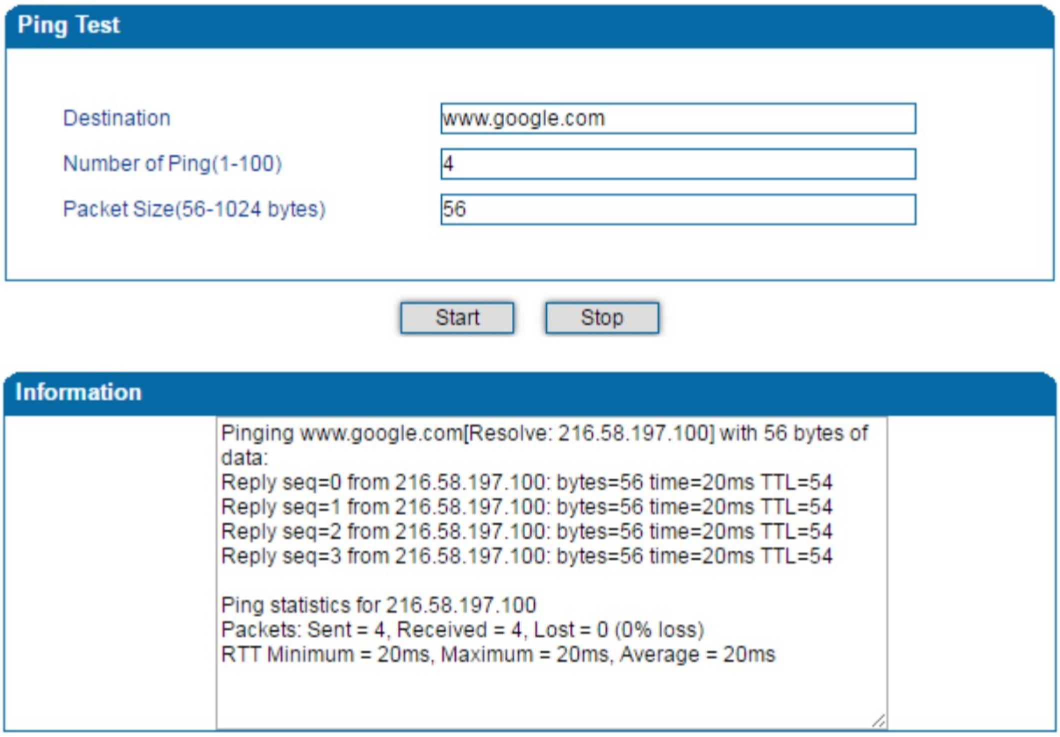
Ping is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

1. Enter the IP address or domain name of a network, a website or a device in the input box of Ping, and then click **Start**.

2. If related messages are received, it means the network works normally; otherwise, the network is not connected or is connected faultily.

Figure 4-15-5 Execute Ping Test



4.15.6 Tracert Test

Tracert is short for traceroute, used to track a route from one IP address to another.

Instruction for using Traceroute:

1) Enter the IP address or domain name of a destination device in the input box of Traceroute, and then click **Start**.

Figure 4-15-6 Execute Tracert Test

Destination: the IP address or domain name of a destination device that needs to be tracked.

Max Hops: the maximum hops for searching the above IP address or domain name. For example, if 'max hops' is set as 30, and the configured IP address or domain name cannot be reached within 30 hops, it's thought that the IP address or domain name cannot be searched.

2) View the route information from the returned message.

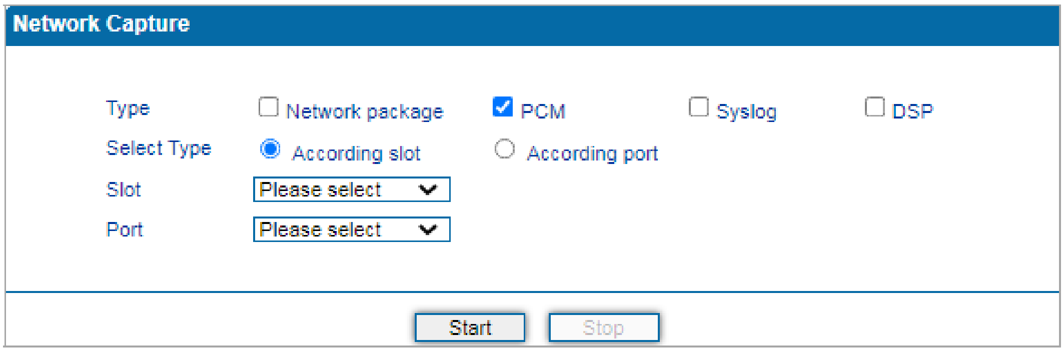
4.15.7 Network Capture

Network capture is an important diagnostics tool for maintenance. It is used to capture data packages of the available network ports.

PCM Capture:

PCM capture helps to analysis voice stream between analog phone and DSP chipset.

Figure 4-15-7.1 Capture PCM Packages



- Click "*Start*' to enable PCM capture
- Dialing out through the device, start talking a short while then hang up the call.
- Click '*Stop*' to disable network capture
- Save the file to local computer

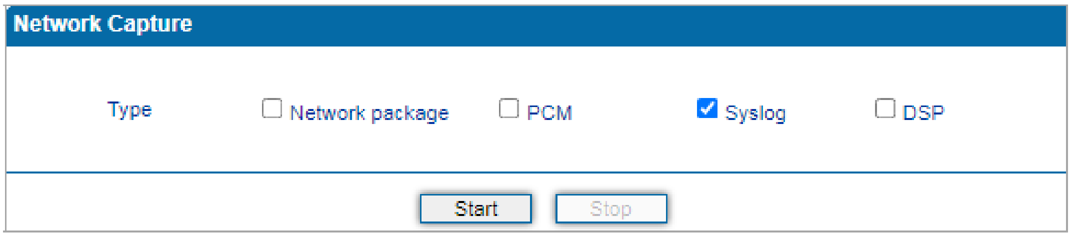
The captured package is named 'capture(x).pcap', among which x is the serial number of the capturing and will be added 1 in next time. The sample of PCM capture as below:

No.	Time	Source	Destination	Protocol	Length	Info	
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021	Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021	Ch: 0xFFFF, Seq: 11 (From Host)
4	1.320893	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0e00	Ch: 0x0003, Seq: 0 (From Host)
5	1.321022	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
6	1.321129	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0e00	Ch: 0x0003, Seq: 1 (From Host)
7	1.329890	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0e01	Ch: 0x0003, Seq: 1 (From Host)
8	1.330010	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
9	1.330093	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0e01	Ch: 0x0003, Seq: 2 (From Host)
10	1.330472	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0802	Ch: 0x0003, Seq: 2 (From Host)
11	1.330566	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
12	1.330639	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0802	Ch: 0x0003, Seq: 3 (From Host)
13	1.330820	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0803	Ch: 0x0003, Seq: 3 (From Host)
14	1.330903	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
15	1.330989	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0803	Ch: 0x0003, Seq: 4 (From Host)
16	1.337791	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9010	Ch: 0x0003, Seq: 4 (From Host)
17	1.337996	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
18	1.338033	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9010	Ch: 0x0003, Seq: 5 (To Host)
19	1.338369	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9000	Ch: 0x0003, Seq: 5 (From Host)
20	1.338460	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
21	1.338564	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9000	Ch: 0x0003, Seq: 6 (To Host)
22	1.343521	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8084	Ch: 0x0003, Seq: 6 (From Host)
23	1.343627	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
24	1.343725	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8084	Ch: 0x0003, Seq: 7 (To Host)
25	1.344060	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8001	Ch: 0x0003, Seq: 7 (From Host)

Syslog Capture:

Syslog capture is another way to obtain syslog which is the same as remote syslog server and filelog. The captured file is saved as pcap format so that it can be opened in some of capturing software like Wireshark, Ethereal software etc.

Figure 4-15-7.2 Capture Syslog Packages



- Click "Start' to enable syslog capture
- Dialing out through the device, start talking a short while then hang up the call.
- Click 'Stop' to disable syslog capture
- Save the capture to local computer

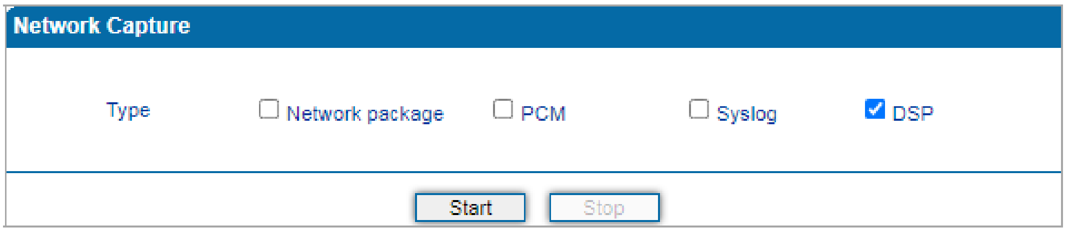
The capture package is named 'capture(x).pcap', among which x is the serial number of capturing and will be added 1 in next time. The sample of syslog capture as below:

No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r
3	0.013432	172.16.222.22	1.1.1.1	Syslog	595	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 2> [DEBUG] <<---*** message from 172.16.222.22/5060,cryp
4	0.013750	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 3> [DEBUG] <<--- from 172.16.222.22/5060,crypt:FALSE, Ph
5	0.014036	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 4> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r
6	0.014512	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 5> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
7	0.014806	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 6> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
8	0.028396	172.16.222.22	1.1.1.1	Syslog	662	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 7> [DEBUG] <<---*** message from 172.16.222.22/5060,cryp
9	0.028759	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 8> [DEBUG] <<--- from 172.16.222.22/5060,crypt:FALSE, Ph
10	0.029052	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 9> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
11	0.030017	172.16.222.22	1.1.1.1	Syslog	233	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 10> [DEBUG] sip-->app: msgtype:ST_SIP_SERVER_CONN \r\n ca
12	0.331167	172.16.222.22	1.1.1.1	Syslog	983	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 11> [DEBUG] <<---*** message from 172.16.222.127/5060,cry
13	0.331498	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 12> [DEBUG] <<--- from 172.16.222.127/5060,crypt:FALSE, P
14	0.331959	172.16.222.22	1.1.1.1	Syslog	907	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 13> [DEBUG] INVITE sip:10086@172.16.222.22:5060 SIP/2.0\r
15	0.332307	172.16.222.22	1.1.1.1	Syslog	122	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 14> [DEBUG] get route entry 31\r\n
16	0.332584	172.16.222.22	1.1.1.1	Syslog	111	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 15> [DEBUG] lPort:3\r\n
17	0.332848	172.16.222.22	1.1.1.1	Syslog	124	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 16> [DEBUG] get route, to port:3\r\n
18	0.333315	172.16.222.22	1.1.1.1	Syslog	526	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 17> [DEBUG] sip-->app: localindex:69, msgtype:SIP_CALL_IN
19	0.333603	172.16.222.22	1.1.1.1	Syslog	173	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 18> [DEBUG] --->> to 172.16.222.127/5060 crypt:FALSE Phon
20	0.333877	172.16.222.22	1.1.1.1	Syslog	386	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 19> [DEBUG] SIP/2.0 100 Trying\r\nVia: SIP/2.0/UDP 172.16.
21	0.346687	172.16.222.22	1.1.1.1	Syslog	131	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 20> [DEBUG] RTP: alg:0, pkt:20, band:-1\r\n
22	0.347453	172.16.222.22	1.1.1.1	Syslog	120	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 21> [DEBUG] dial tick:102433\r\n
23	7.232839	172.16.222.22	1.1.1.1	Syslog	533	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 22> [DEBUG] <<---*** message from 172.16.222.127/5060,cry
24	7.233513	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 23> [DEBUG] <<--- from 172.16.222.127/5060,crypt:FALSE, P
25	7.233959	172.16.222.22	1.1.1.1	Syslog	457	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 24> [DEBUG] CANCEL sip:10086@172.16.222.22:5060 SIP/2.0\r
26	7.234596	172.16.222.22	1.1.1.1	Syslog	287	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 25> [DEBUG] sip-->app: localindex:69, msgtype:SIP_CALL_BY

DSP Capture:

DSP capture helps to analyze voice stream inside DSP chipset. The DSP chipset will handle RTP from IP network as well as voice stream from analog phone.

Figure 4-15-7.3 Capture DSP Packages



- Click Start to enable DSP capture
- Dialing out through the device, start talking a short while then hang up the call.
- Click Stop to disable DSP capture
- Save the capture to local computer

The captured package is named 'capture(x).pcap', among which x is the serial number of the capturing and will be added 1 in next time. The sample of RTP capture as below:

No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 2 (From Host)
2	0.007246	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
3	0.007260	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 5 (From Host)
4	2.994581	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 3 (From Host)
5	2.997308	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
6	2.997316	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 6 (From Host)
7	5.992790	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 4 (From Host)
8	5.997282	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
9	5.997290	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 7 (From Host)
10	7.691428	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9010 Ch: 0x0003, Seq: 3 (From Host)
11	7.691552	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
12	7.691715	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9010 Ch: 0x0003, Seq: 1 (To Host)
13	7.701379	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9000 Ch: 0x0003, Seq: 4 (From Host)
14	7.701494	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
15	7.701622	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9000 Ch: 0x0003, Seq: 2 (To Host)
16	7.709662	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8084 Ch: 0x0003, Seq: 5 (From Host)
17	7.709798	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
18	7.709902	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8084 Ch: 0x0003, Seq: 3 (To Host)
19	7.710238	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8001 Ch: 0x0003, Seq: 6 (From Host)
20	7.710328	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
21	7.710496	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8001 Ch: 0x0003, Seq: 4 (To Host)
22	7.716241	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8018 Ch: 0x0003, Seq: 7 (From Host)
23	7.716352	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
24	7.716465	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8018 Ch: 0x0003, Seq: 5 (To Host)
25	7.716711	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x805b Ch: 0x0003, Seq: 8 (From Host)

4.15.8 Factory Reset

Click 'Apply' to restore configurations of the device to the factory default settings.

Figure 4-15-8 Reset Device to Factory Default Setting

4.15.9 Device Restart

For some configurations or changes to the FXO GW device, you are required to restart the device for the configurations or changes to take effect.

Figure 4-15-9 Restart Device

5 Glossary

Abbr.	Full Name
ACD	Automatic Call Distribution
DNS	Domain Name System
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
RTP	Real Time Protocol
POE	point-to-point protocol over Ethernet
VLAN	Virtual Local Area Network
ARP	Address Resolution Protocol
CID	Caller Identity
DND	Do NOT Disturb
DTMF	Dual Tone Multi Frequency
NTP	Network Time Protocol
DMZ	Demilitarized Zone
STUN	Simple Traversal of UDP over NAT
PSTN	Public Switched Telephone Network