Dinstar DAG-FXO VoIP Gateway

User Manual V2.0

Dinstar Technologies Co., Ltd.

Address: 9th Floor, Guoxing Building, Changxing Road, Nanshan District, Shenzhen, China

Postal Code: 518052

Telephone: +86 755 61919966

Fax: +86 755 2645 6659

Emails: sales@dinstar.com, support@dinstar.com

Website: www.dinstar.com

Preface

Welcome

Thanks for choosing **Dinstar DAG-FXO VoIP Gateway**! We hope you will make optimum use of this flexible, rich-feature VoIP gateway. Please read this document carefully before install the gateway.

About This Manual

This manual provides information about the introduction of the gateway and about how to install, configure or use the gateway.

This manual is written with reference to the default configurations of the **DAG-FXO VoIP Gateway**.

Intended Audience

This manual is aimed primarily at engineers who will install, configure or maintain the gateway.

- System Maintenance Engineer
- Network Engineer

Revision Records

Document Version 2.1 (2019/10/09)

- Document Name: Dinstar DAG-FXO VoIP Gateway User Manual V2.1
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- Revised Items: document layout, all contents, application scenario and network diagrams

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- Document Name: DAG Series FXO Voice Gateway User Manual V2.0

Contents

1 Product Description

1.1 Overview

Dinstar's DAG-FXO VoIP gateway is an access gateway based on IP network. It can provide low-cost, easy-to-use VoIP solutions and high-quality voice services for SOHO (Small Office-Home office), remote office, small enterprise and enterprise with multiple branches.

Based on standard SIP protocol, the gateway is compatible with IP PBX, soft-switch and SIP-based platforms.

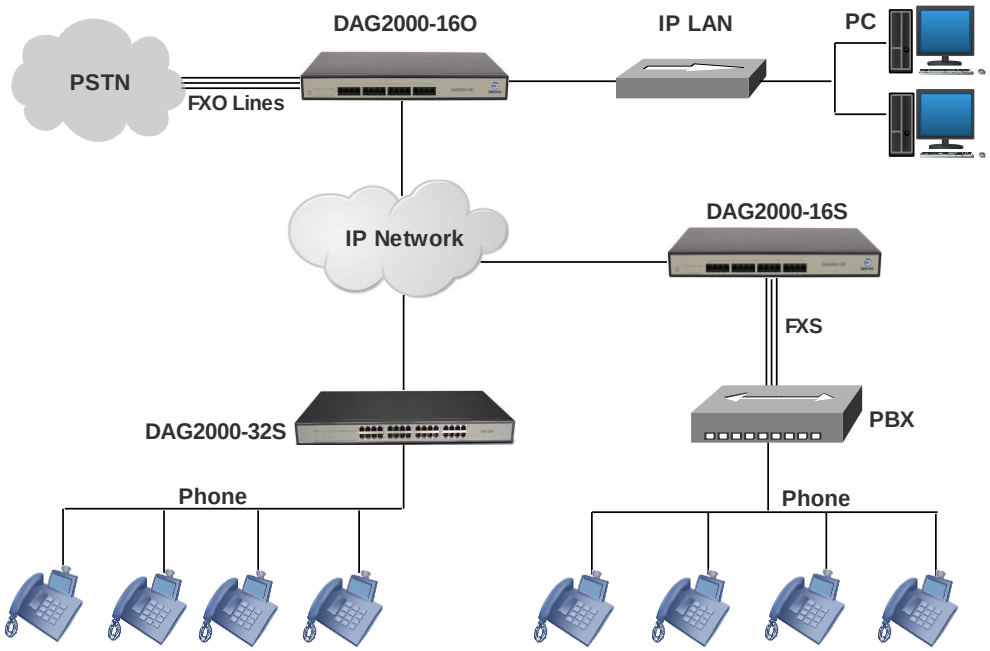
Dinstar provides FXO VoIP gateways with following models:

- DAG1000-4O
- DAG1000-8O
- DAG2000-16O

This manual takes DAG2000-16O as the example to introduce the functions and configurations of Dinstar's FXO VoIP gateway, since the functions and configurations of all models are almost the same.

1.2 Application Scenario

Figure 1-1 Application Scenario of DAG2000-16O VoIP Gateway:



1.3 Product Appearance

Figure 1-2 Image of DAG1000-4O

Figure 1-3 Image of DAG1000-8O

Figure 1-4 Images of DAG2000-16O

1.4 Ports, Connectors and Indicators

1.4.1 DAG1000-4O

Figure 1-5 Indicators f DAG1000-4O

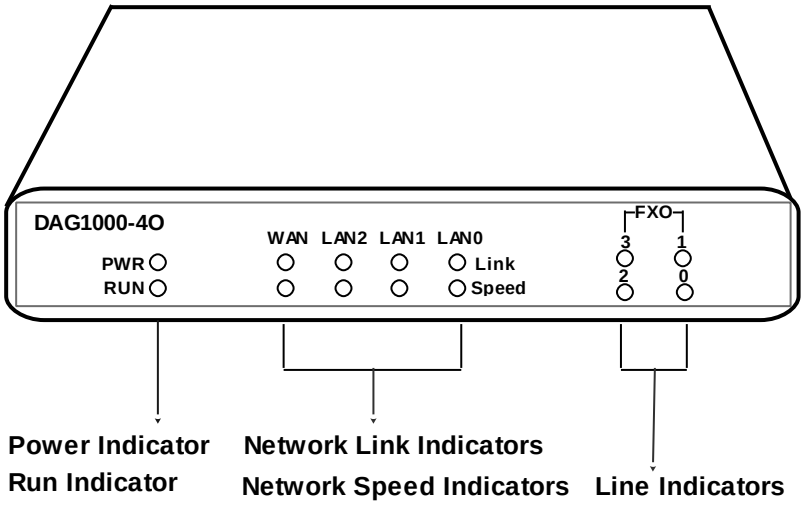


Figure 1-6 Ports of DAG1000-4O

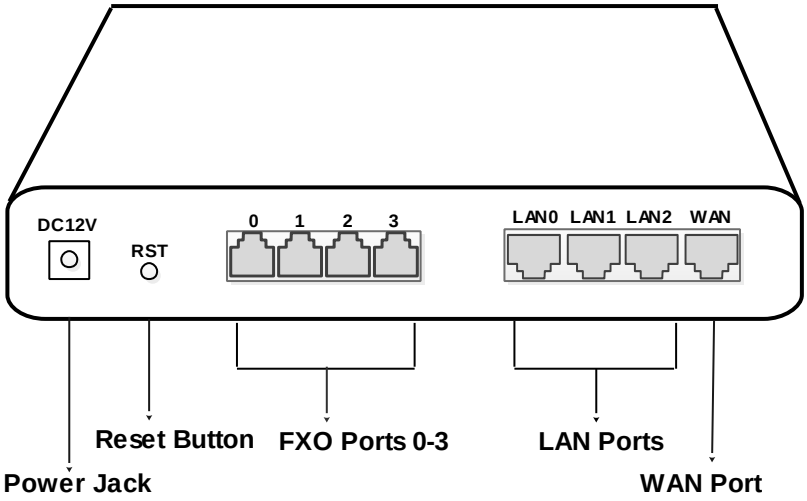


Table 1-1 Description of Ports

Port Name	Description
Power Jack	To connect DC 12V power supply
WAN/LAN Port	To connect to IP network over a DSL modem, router or LAN switch
FXO Ports 0-3	To connect to PSTN

Reset Button	Used to reset the gateway to factory default setting

1.4.2 DAG1000-8O

Figure 1-7 Indicators f DAG1000-8O

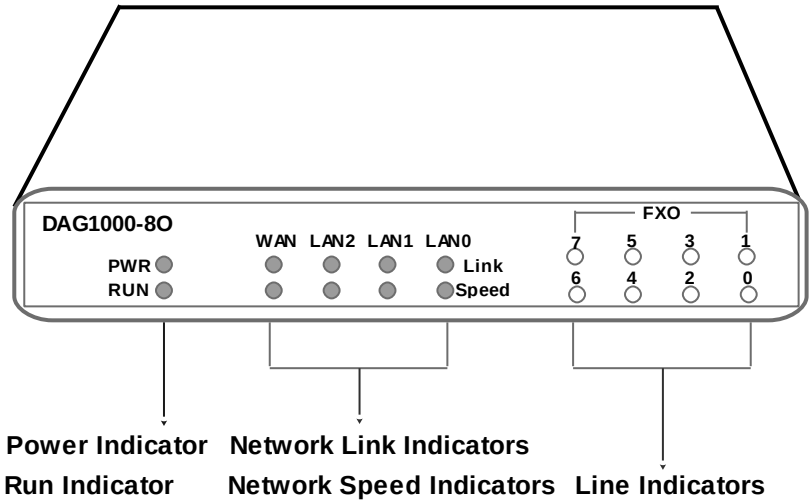


Figure 1-8 Ports of DAG1000-8O

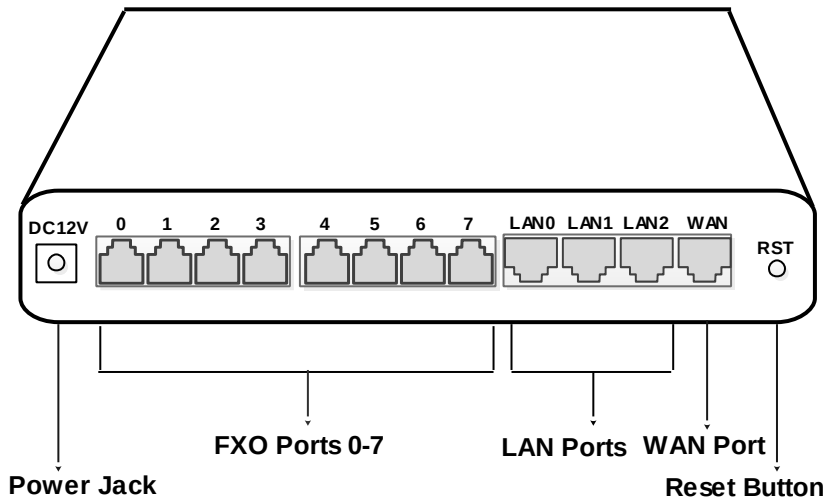


Table 1-2 Description of Ports

Port Name	Description
Power Jack	To connect DC 12V power supply
WAN/LAN Port	To connect to IP network over a DSL modem, router or LAN switch
FXO Ports 0-7	To connect to PSTN

Reset Button	Used to reset the gateway to factory default setting

1.4.3 DAG2000-16O

Figure 1-9 Indicators f DAG2000-16O

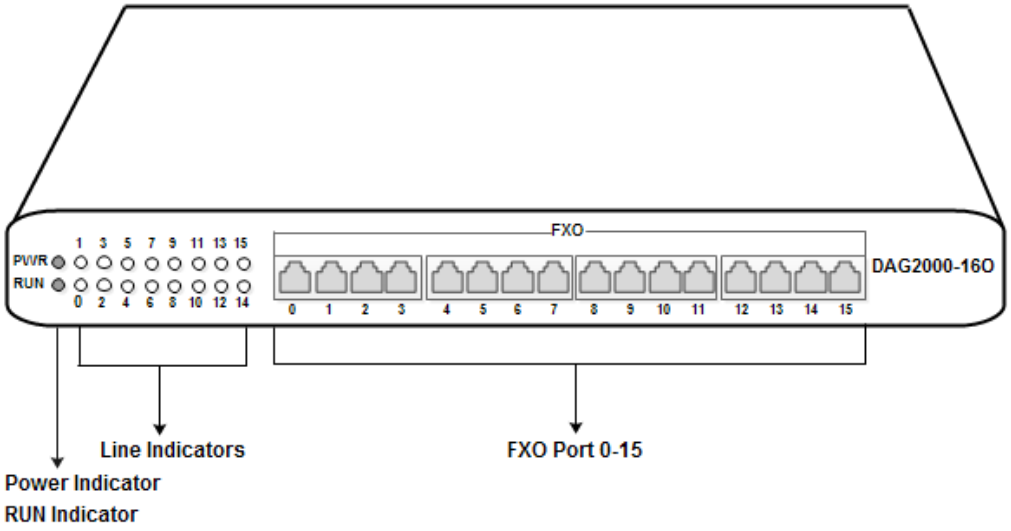


Figure 1-10 Ports of DAG2000-16O

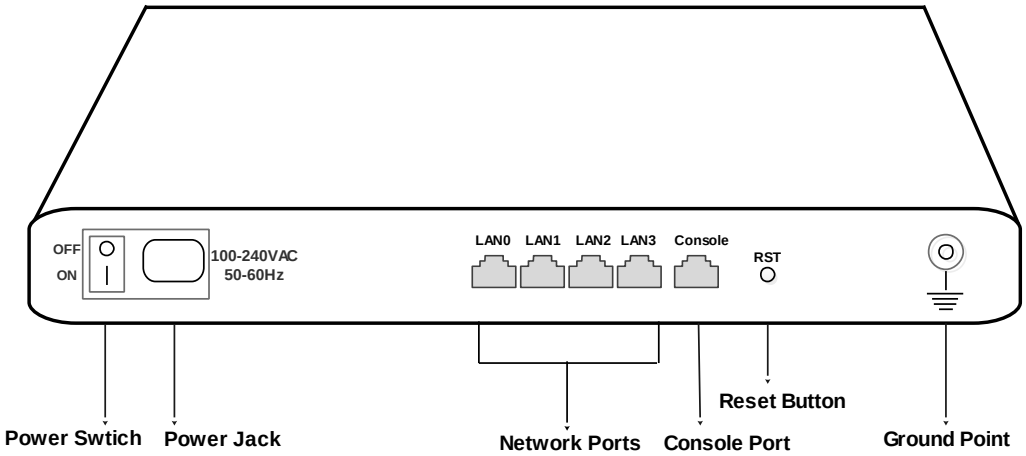


Table 1-3 Description of Ports

Port/Connector	**Description**
Power Switch	Turn on or turn off power supply
Power Jack	To connect 100-240V AC power supply
Reset Button	Used to reset the gateway to factory default setting

Console Port	Used to connect to serial line for the purpose of carrying out maintenance-related configurations
LAN Ports	To connect to IP network over a DSL modem, router or LAN switch
FXO Ports 0-15	To connect to PSTN
Ground Point	Used to connect to a ground wire

1.5 Description of Indicators

Indicator	**Status**	**Description**
PWR	On green	The DAG-FXO gateway is powered on.
	On dull	There is no power supply or power supply is abnormal.
RUN	Slow flash	The gateway is running properly.
	Quick flash	SIP account is registered successfully.
	No flash/ On dull	The gateway is running improperly.
FXO	On green	The corresponding FXO port is currently occupied by a call.
	On dull	No call occupies the corresponding FXO port currently.
WAN/LAN (Link)	On green	The gateway is properly connected to network.
	On dull	The gateway is not connected to network or the connection is improper.
WAN/LAN(Speed)	On green	The network speed is 100Mbps.
	On dull	The network speed is 10Mbps.

1.6 Functions and Features

1.6.1 Protocol standard supported

- SIP V2.0 (RFC 3261,3262,3264)
- SDP (RFC 2327)
- REFER (RFC 3515)
- RTP/RTCP (RFC 1889,1890)
- STUN (RFC 3489)

- ARP/RARP (RFC 826/903)
- SNTP (RFC 2030)
- DHCP/PPPoE
- TFTP/HTTP/HTTPS
- DNS/DNS SRV (RFC 1706/RFC 2782)
- VLAN 802.1P/802.1Q
- Differentiated Server

1.6.2 Voice and Fax Parameters

- G.711A/U law, G.723.1, G.729AB
- Comfortable Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation (G.168)
- Adaptive Dynamic Jitter Buffer
- Voice and Fax Gain Control
- Modem
- T.38/Pass-through
- DTMF Mode: Signal/RFC2833/INBAND

1.6.3 Supplementary Services

- Busy Tone Detection
- Detection for No Current in FXO Line
- Detection for Voice Interruption
- One-stage Dialing
- Two-stage Dialing
- Polling on FXO ports under PSTN
- Polarity Reversal
- Fake Billing Correction (FAS)
- DC/AC Impedance Configuration
- Calls Detection (Bellcore Type 1&2, ETSI, DTMF)
- Port Group
- IP Trunk
- Voice Mail

- Direct IP Call
- Redundant Registration Server
- 32 Inbound Routes
- 32 Outbound Routes
- Manipulation of Inbound and Outbound numbers

2 Basic Operation

2.1 Callout and Callin via FXO Port

2.1.1 Call Out

One-stage Dialing: After the DAG-FXO 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 go 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 PSTN's dialing tone. After that, you will be able to dial any number of PSTN lines.

2.1.2 Call In

Dial the number of a PSTN line connected to a FXO port of DAG-FXO gateway, and then you will hear a secondary 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 a PSTN line connected to a FXO port of DAG-FXO gateway, then the gateway will automatically route the call to designated extension number or telephone number according to preset hotline number.

2.1.3 Direct IP Calls

DAG series device with FXO port allow two parties directly call through IP address. The user need only a simulation with the FXO port unit equipment linked together and set up calls not registered.

Elements necessary to completing a direct IP call：

(1) Both DAG serial and other VoIP Device, have public IP addresses;

(2) Both DAG serial and other VoIP Device are on the same LAN using private IP addresses;

(3) Both DAG serial and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Operation Process：

1）Pick up the analog phone then dial “*47”

2）Enter the target IP address.

【Note】: No dial tone will be played between step 1 and step 2

Examples:

If the target IP address is 192.168.0.160, the dialing convention is ***47,** then **192*168*0*160.** Followed by pressing the “#” key or wait 3 seconds. Complete signaling interactive soon after, he was called the unit can be heard ringing.

【Note】: You cannot make direct IP calls between FXO0 to FXO1 since they are using same IP. It only supports the default destination port 5060.

2.2 Call Features

DAG (FXO) support all traditional and senior phone function.

DAG (FXO) support all traditional and senior phone function.

Table 2.2-1 Feature Codec

Feature Codec	Operation Instructions
***158#**	View the LAN port IP address
***159#**	View the WAN port IP address
***114#**	Inquire port account
150	Set the way of obtain IP address
157	Set network method
152	Set IP address
153	Set Subnet mask
156	Set default gateway IP address
***193#**	Obtain IP address through DHCP again
***160*1#**	Open WAN port to access web

***166*000000#**	Factory reset
***111#**	Restart device
***#**	Call hold
47	IP address call
***51#**	Enable call waiting
***50#**	Disable call waiting
87	Blind transfer
72	Enable Unconditional Call Forward
***73#**	Disable Unconditional Call Forward
90	Enable Busy Call Forward
***91#**	Disable Busy Call Forward
92	Enable No Answer Call Forward
***93#**	Disable No Answer Call Forward
***78#**	Enable DND
***79#**	Disable DND
***200#**	Access Voice mail
Flash/Hook	Switch between incoming calls, If not in session, flash/hook will switch a new channel for new call.

2.3 Sending and Receiving Fax

The DAG-FXO gateway supports four fax modes:

- T.38 (FoIP)
- Pass-Through
- Modem
- Adaptive

2.3.1 T. 38 and Pass-Through

T.38 is the preferred fax mode because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used.

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.

2.4 Local IVR Operation

2.4.1 Inquire IP address

Connect a PSTN line to one of the FXO ports of the DAG-FXO 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 voive prompt, dial *158# to inquire the IP address of LAN port and dial *159# to inquire the IP address of WAN port.

2.4.2 Factory Reset

Connect a PSTN line to one of the FXO ports of the DAG-FXO 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 voive prompt, dial *166*000000#, and you will hear “successful setting”, then hang up the phone and the gateway is reset to factory defaults.

2.4.3 Set IP Address

Before configuration, please ensure:

- The gateway is power on;
- Device has been connected to network;
- Telephone is connected to FXO port of the DAG1000-4S/8S gateway.

Configure dynamic IP address by DHCP:

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

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.

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

Then configure IP address and subnet mask as follow:

• Configure IP address

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

• Configure subnet mask

Pick up the phone, dial *153*255*255*0*0# and then hang up the phone.

• Configure gateway IP address

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

• Query the IP address of the DAG1000-4S/8S gateway:

Pick up the phone, 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.

3 Web Configuration

3.1 WEB Login

The default IP addresses of LAN port of the device is 192.168.11.1. Connect the DAG FXO gateway to the network according to the following network topology, and dial *158 to query the IP address of the gateway.

3.1.1 Login

Modify the IP address of the PC to make it at the same network segment with the DAG FXO device, since the default IP address of the gateway is 192.168.11.1.

Take Windows 7 as an example, the IP address of PC is changed into 192.168.11.10：

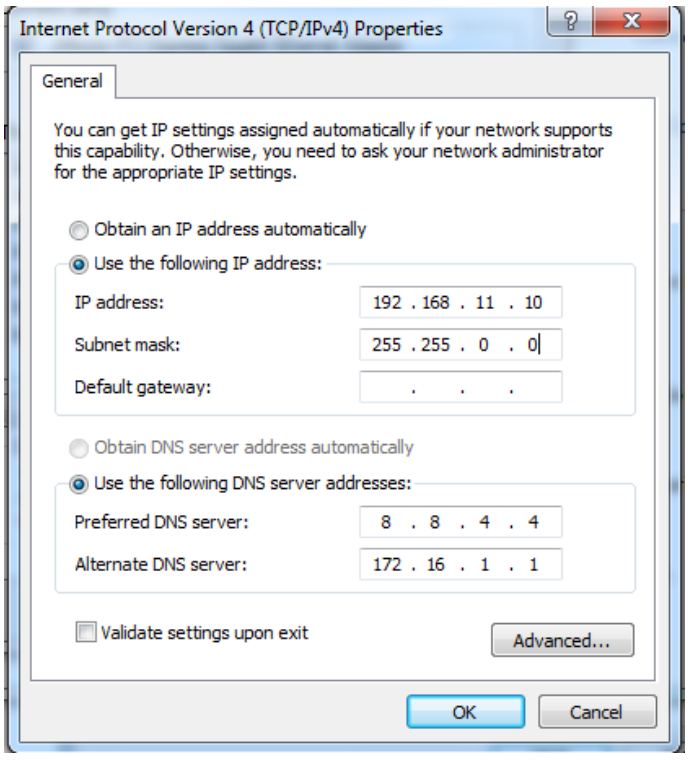


Figure 3.1-1Modify IP address

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

3.1.2 Login WEB

Open a web browser and enter the IP address of LAN port of the DAG FXO gateway (the default IP of LAN port is 192.168.11.1; if this IP address has been changed, please enter the new IP address). Then the login GUI will be displayed. Both the default username and password are admin.

It is advised to modify the username and password for security consideration.

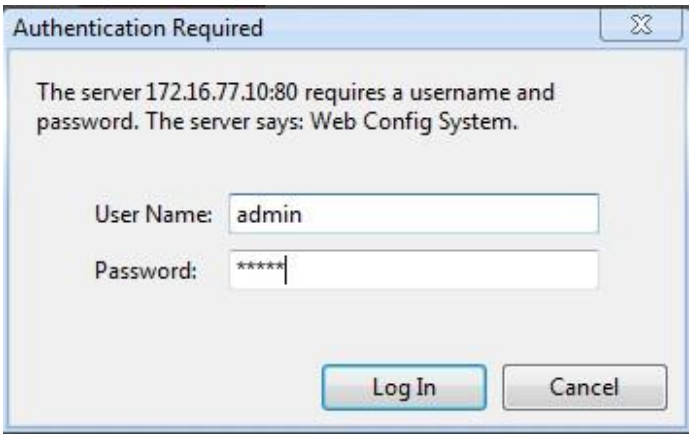


Figure 3.1-1 DAG FXO Login Interface

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.

Figure 3.1-2 DAG Configure Interface

3.2 Navigation Tree

DAG series voice gateway web configuration interface mainly includes navigation tree and the right configuration interface. Choose navigation tree in order to entry into the configuration interface.

Figure 3.2-1 Navigation Tree

When device is in bridge mode, navigation tree won't display "routing configuration" items and the following "DHCP service", "DMZ host", "forward rules" and "static routing" and "ARP" etc.

3.2.1 System Information

System information interface shows the run information as following figure 4.3.1 below:

Figure 3.3-1 System Information

System information as follow:

Table 4.3-1 System Information Description

Device ID	A unique ID of each device. This ID is used for warranty and cloud server authentication.
MAC address	Hardware address of the DAG FXO gateway
IP Address	The IP address of LAN port of the gateway **DHCP: Obtain IP address automatically**. DAG FXO gateway is regarded as a DHCP client, which sends a broadcast request and looks for a DHCP serve to answer. Then the first discovered DHCP server automatically assigns an IP address to the DAG FXO gateway from a defined range of numbers.

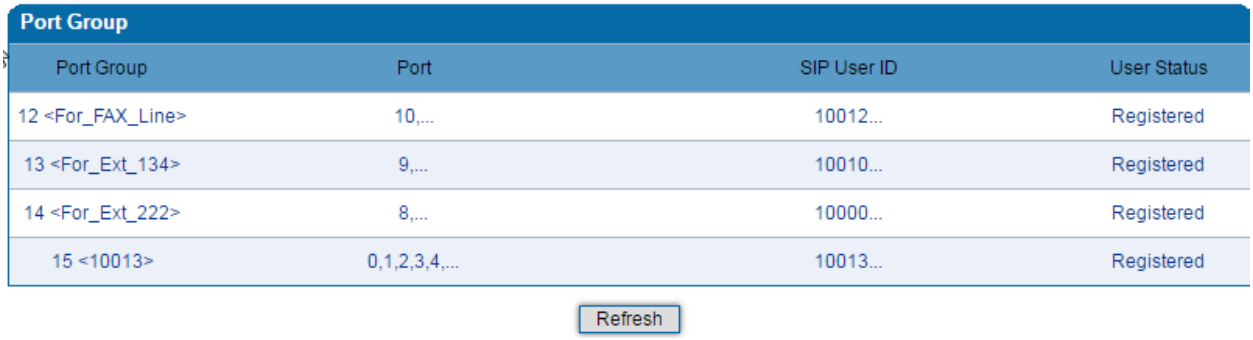
	Static IP Address: Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a **dynamic IP address,** which is assigned *ad hoc* at the start of each session, normally changing from one session to the next. If you choose static IP address, you need to fill in the following information: ● IP Address: the IP address of LAN port of the DAG FXO gateway; ● Subnet Mask: the netmask of the router connected the DAG FXO gateway; ● Default Gateway: the IP address of the router connected the DAG FXO gateway; **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 address of DNS server and default gateway information is displayed.
Cloud Register Status	Whether the DAG FXO gateway is registered to cloud or not.
System Uptime	The running time of the DAG FXO gateway since it is powered on.
NTP Status	Succeed: the DAG FXO gateway is sync to NTP server successfully; Failed: the DAG FXO gateway fails to be sync to NTP server. Then you should check network connection and the NTP server.
Network Traffic Statics	Total bytes of message received and sent by network port.
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 gateway. 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 DAG FXO gateway

3.2.2 Port Status

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
8	FXO	---	---	OnHook	Idle
9	FXO	---	---	OnHook	Idle
10	FXO	---	---	Offline	Idle
11	FXO	---	---	Offline	Idle
12	FXO	---	---	Offline	Idle
13	FXO	---	---	Offline	Idle
14	FXO	---	---	Offline	Idle
15	FXO	---	---	Offline	Idle



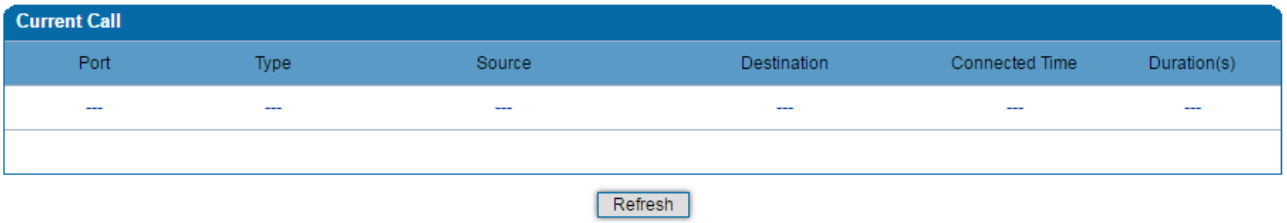
Port Group

Port Group	Port	SIP User ID	User Status
12 <For_FAX_Line>	10,...	10012...	Registered
13 <For_Ext_134>	9,...	10010...	Registered
14 <For_Ext_222>	8,...	10000...	Registered
15 <10013>	0,1,2,3,4,...	10013...	Registered

Refresh

Figure 3.3-2 Port and Port group registration information

3.2.3 Current Call



Current Call

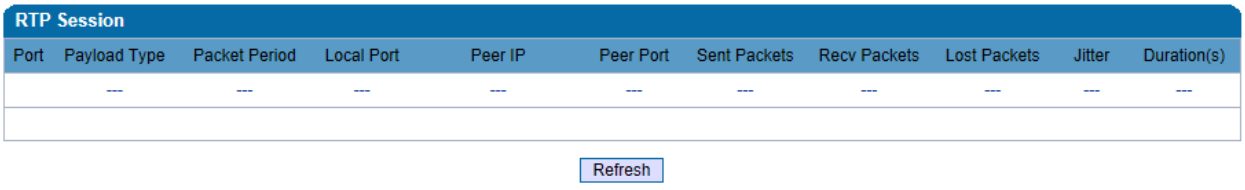
Port	Type	Source	Destination	Connected Time	Duration(s)
---	---	---	---	---	---

Refresh

Figure 3.3-3 Current Call Statistics

The above interface shows the the current call information. If there are no calls currently, no information will be displayed.

3.2.4 RTP Session



RTP Session

Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Sent Packets	Recv Packets	Lost Packets	Jitter	Duration(s)
---	---	---	---	---	---	---	---	---	---	---

Refresh

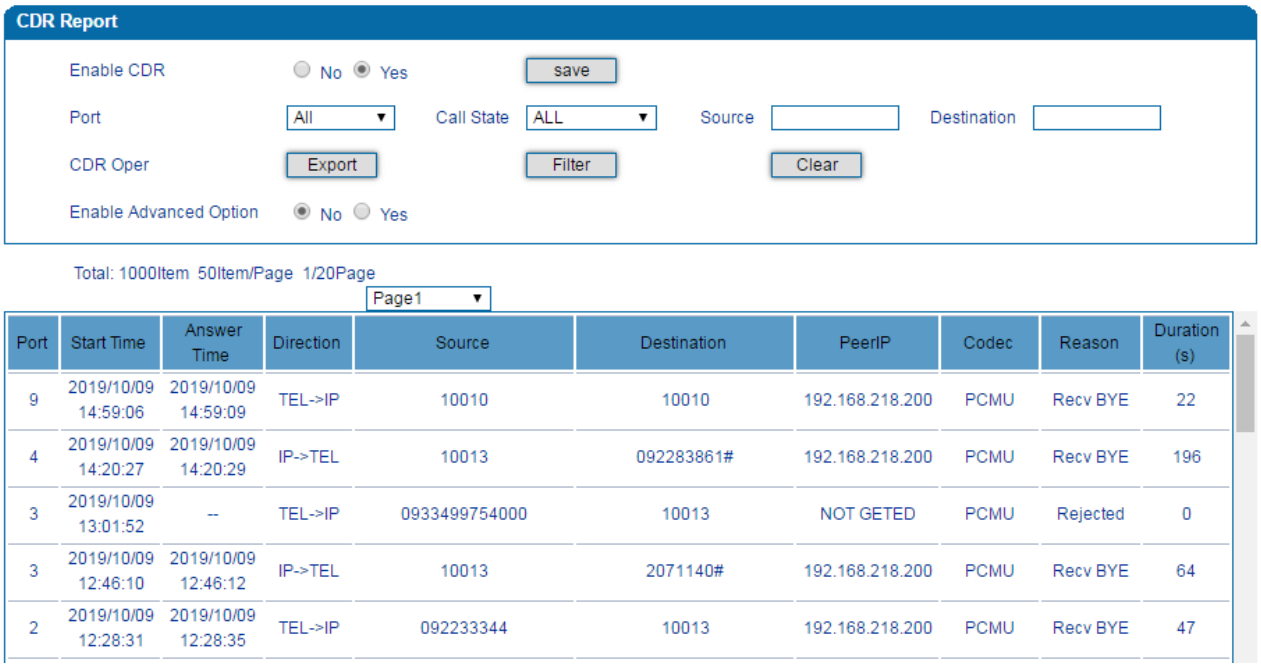
Figure 3.3-4 RTP Session Statistics

The above interface shows real-time RTP conversation flow data information, includes:

Port, voice codec, packet period, local port, peer IP, peer port, sent packets, receive packets, lost packets, jitter and duration.

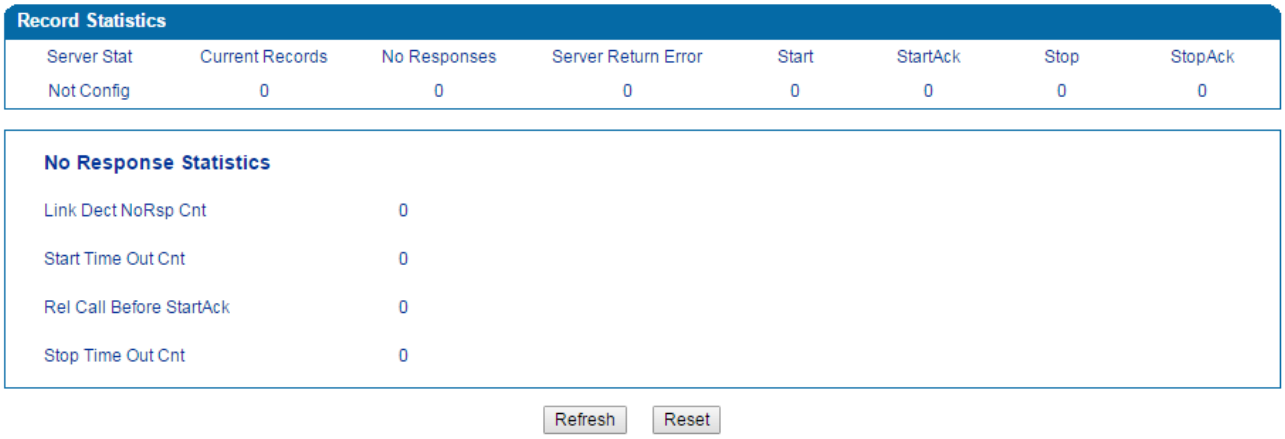
3.2.1 CDR

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.



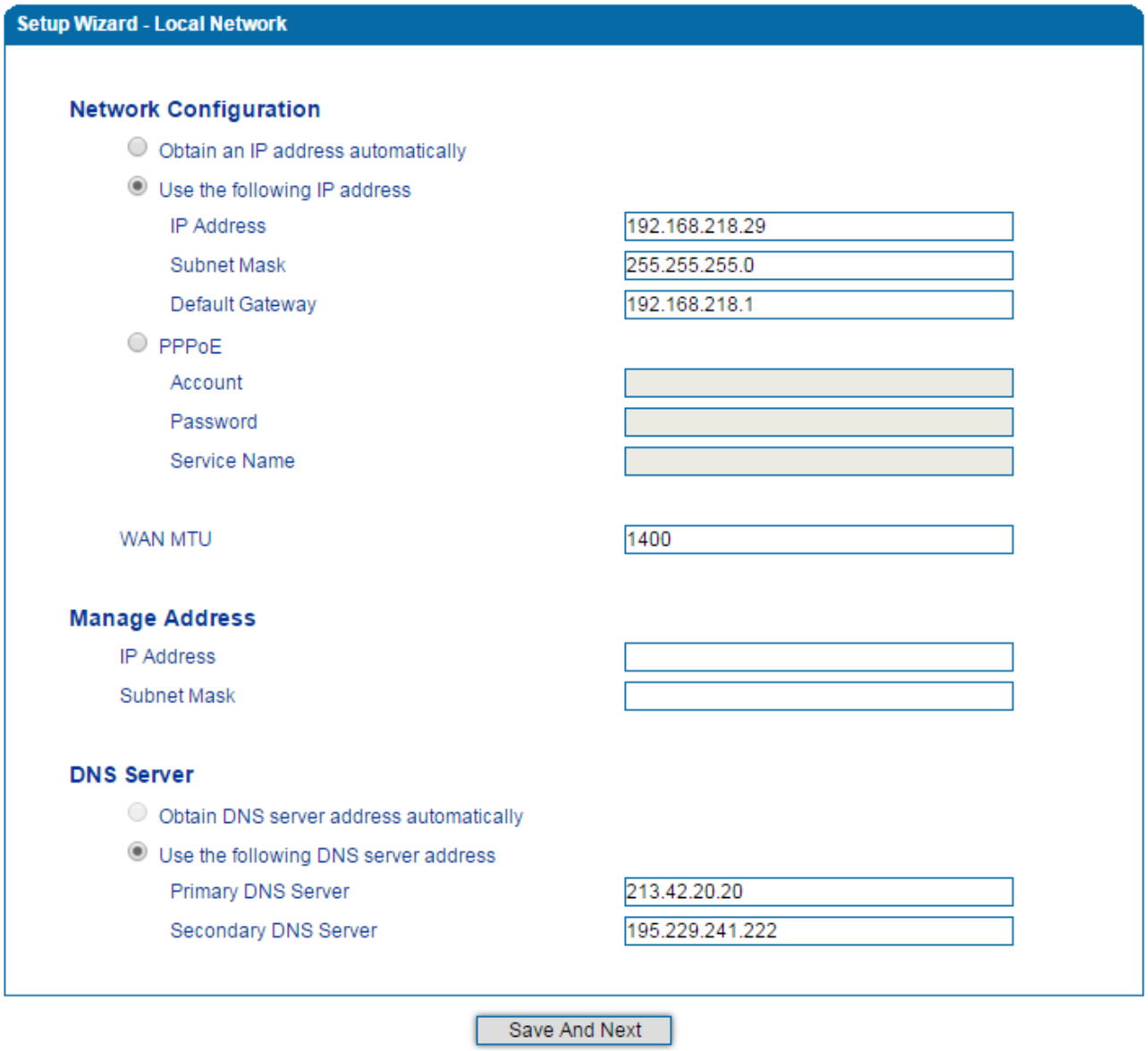
On the **Status & Statistic → CDR** interface, details of all calls through the DAG FXO gateway are displayed. The CDR function can be enabled on this interface.

3.2.1 Record Statistics



3.3 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 DAG FXO device.



3.4 Network Configuration

3.4.1 Local Network

DAG has two kinds of work mode: route and bridge. When DAG is set rout mode, the DAG will work as small router and NAT function has enabled. In this situation, WAN port is normally connect to uplink router/switch or

ADSL MODEM, LAN port used to connect local computer or other network device(such as Ethernet switches, Hubs etc); When DAG is set bridge mode, WAN and LAN port are the same. The DAG just work as two ports or four ports Ethernet switch.

When it set to bridge mode, only need to configure WAN port IP address and DNS. If set to route mode, default LAN port IP will display and it can be change by users.

Note: DAG2000-16O just supports bridge mode. DAG1000-4/8O supports bridge and route mode.

Bridge mode

Network configuration of bridge mode:

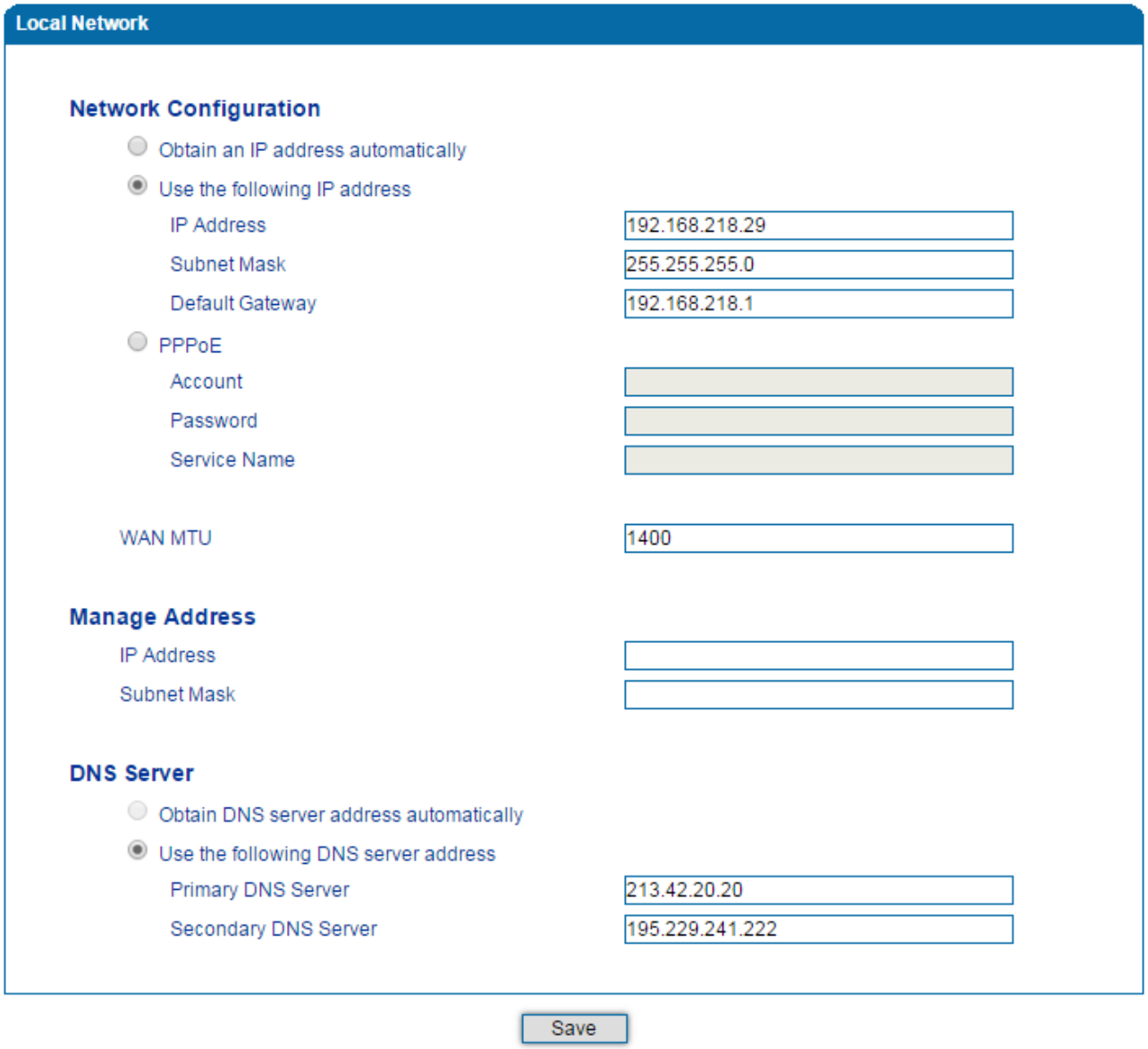


Figure 3.5-1 Local network

- When “Obtain IP address automatically” is selected, the gateway will obtain IP address by DHCP.
- When “Use the following IP address” is selected, user needs to configure a static IP address.

- When “PPPoE” is selected, user needs to fill in the account and password offered by ISP.

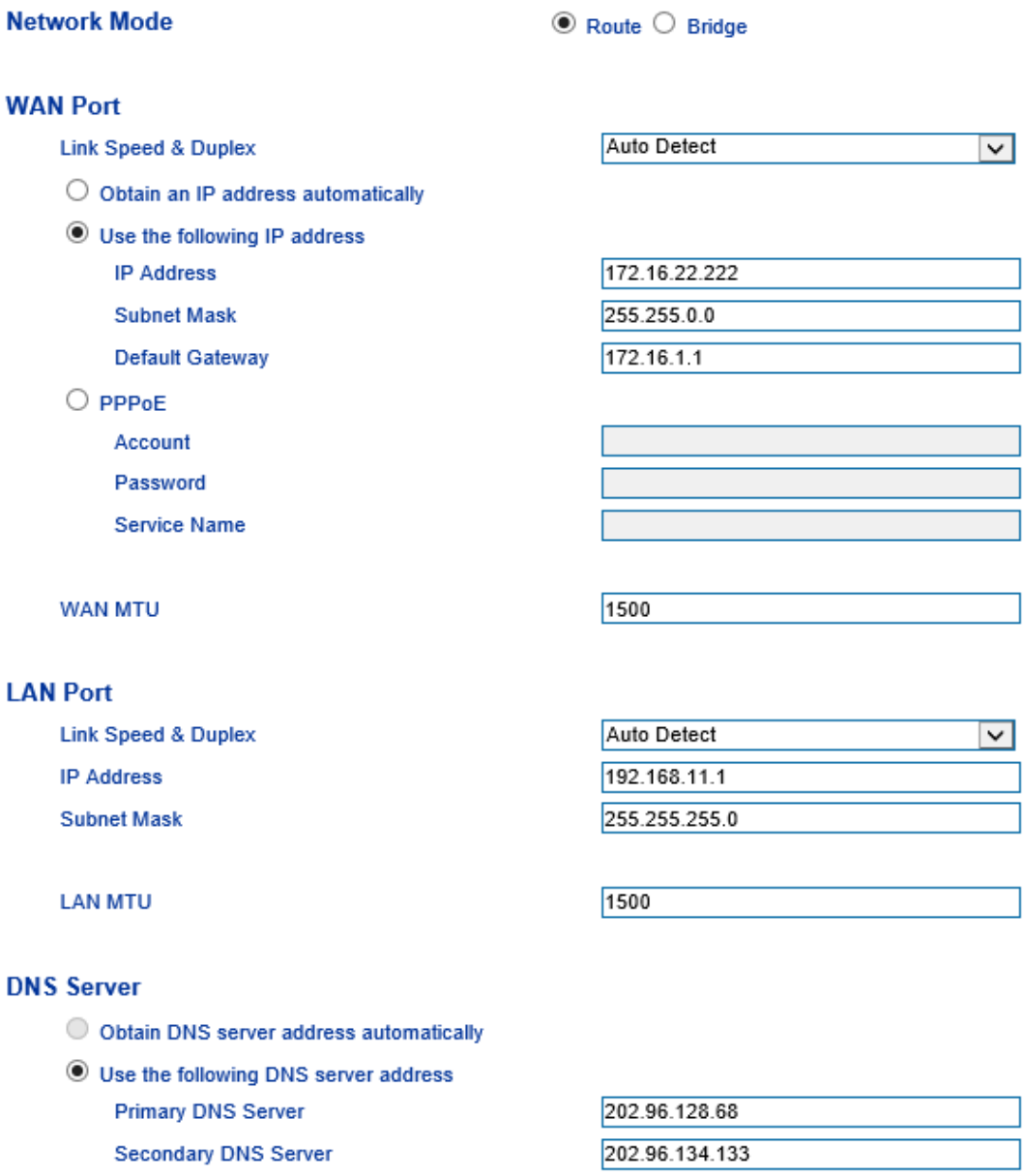
【Notes】:

1） If DHCP is selected to obtain IP address, please ensure DHCP server in the network works normally.

2） After the configurations are finished, please restart the gateway for the configurations to take effect.

Route Mode

Network configuration of route mode:



Notes: The following settings are available on route mode only!

3.4.2 VLAN Parameter

In order to control the impacts brought by broadcast storms, user can divide VLANs into three groups, namely VLAN1, VLAN2 and VLAN3. There are kinds of VLAN, including data VLAN, voice VLAN and management VLAN. Different kind of VLAN has different messages.

- **802.1Q**

The IEEE 802.1Q standard defines the architecture for Virtual Bridged LANs, the services provided in Virtual Bridged LANs and the protocols and algorithms involved in the provision of those services.

No Quality of Service mechanisms are defined in this standard, but an important requirement for providing QoS is included in this standard, e.g. the ability to regenerate user priority of received frames using priority information contained in the frame and the User Priority Regeneration Table for the reception Port.

- **802.1P**

IEEE 802.1P standard, describes important methods for providing QoS at MAC level. IEEE 802.1p is in fact quite good. Lower priority level packets are not sent, if there are packets in queued in higher level queues. IEEE 802.1p describes no admission control protocols. It would be possible to give Network Control priority to all packets and the network would be easily congested.

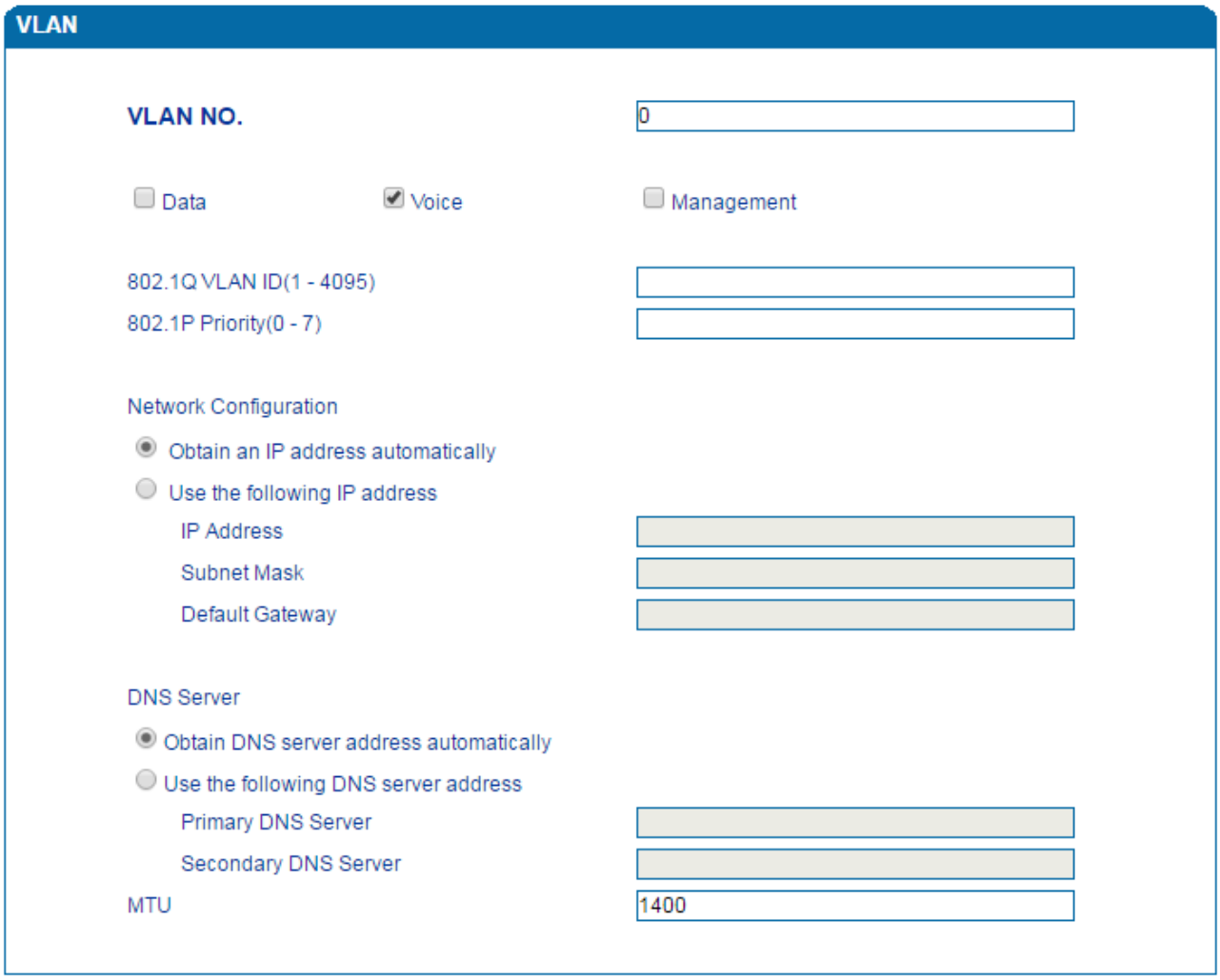


Figure 3.7-3 VLAN parameter configuration

Explanations of the parameters in VLAN interface:

VLAN1/VLAN2/VLAN3	The gateway supports three VLANs at most. Please enable VLAN according to actual needs.
Data/Voice/Management，	If the checkboxes on the right of data, voice and management of VLAN1 are selected, it means data messages, voice messages and management messages are subject to the network setting, 802.1Q VLAN1 ID and 802.1P Priority of VLAN1.
802.1Q VLAN ID(0-4095)	Set an ID to identify a VLAN based on 802.1Q protocol.
802.1p Priority（0-7）	Set the priority of a VLAN based on 802.1P protocol.
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】：User needs to restart the gateway for the configurations to take effect.

【Note】：restart the device to take configuration effect.

3.4.3 DHCP Option (Routing mode)

On the following interface, you can choose the source of DHCP. Network interfaces include Data VLAN, Manage VLAN and Voice VLAN, while DHCP sources include TFTP Server, SIP Server and Static Route.

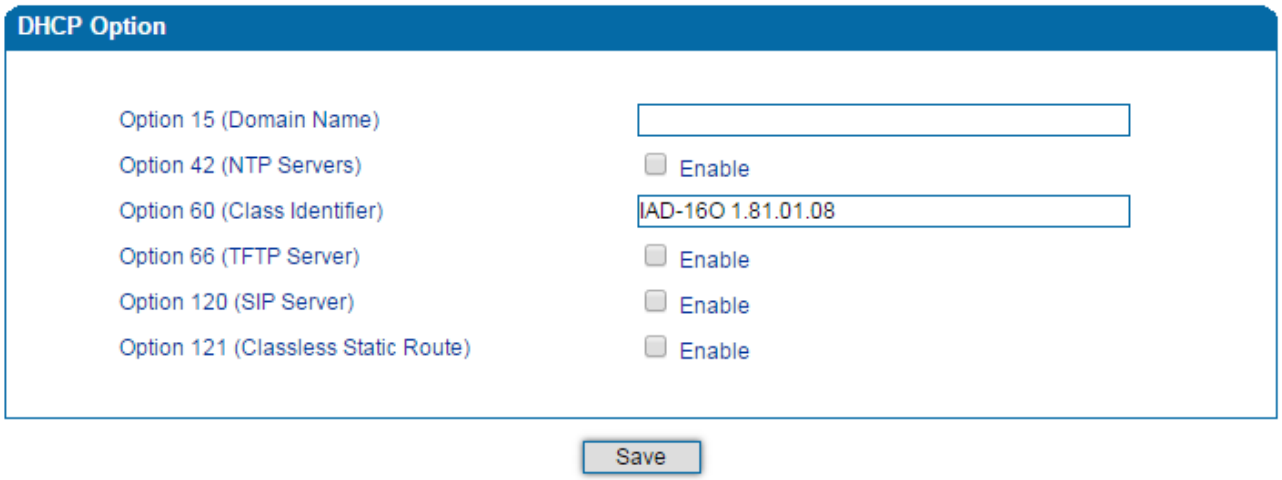
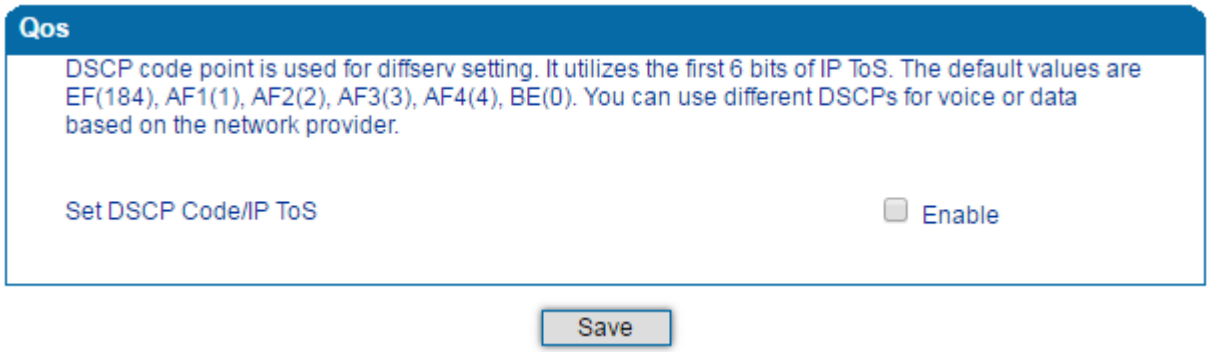


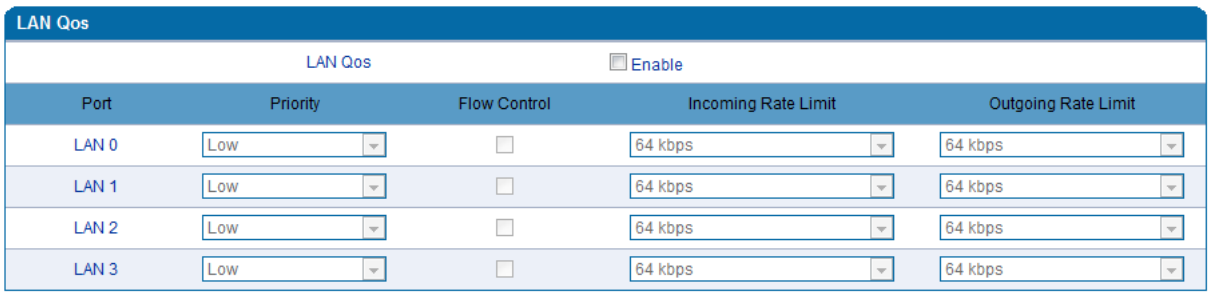
Figure 3.5-4 DHCP Option Interface

3.4.4 QoS



3.4.5 LAN QoS

On the **LAN Qos** interface, user can set the priority of each LAN port and limit the incoming rate or outgoing rate of each LAN port.



3.4.6 ARP

ARP is address resolution protocol. ARP helps user 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. ARP is a tool that converts IP address into MAC address.

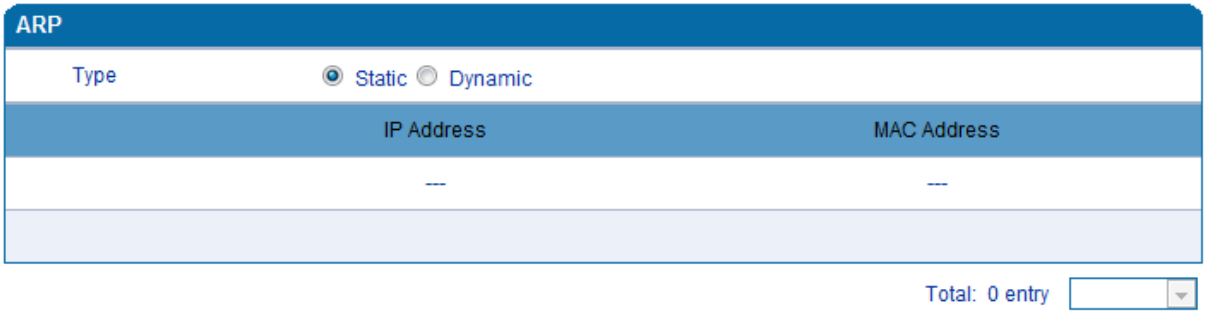


Figure 3.7-9 ARP Parameters

3.5 SIP Server

Introduction of SIP Server:

1）SIP server is the main component of VoIP network and is responsible for establishing all the 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.

2）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.

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

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

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

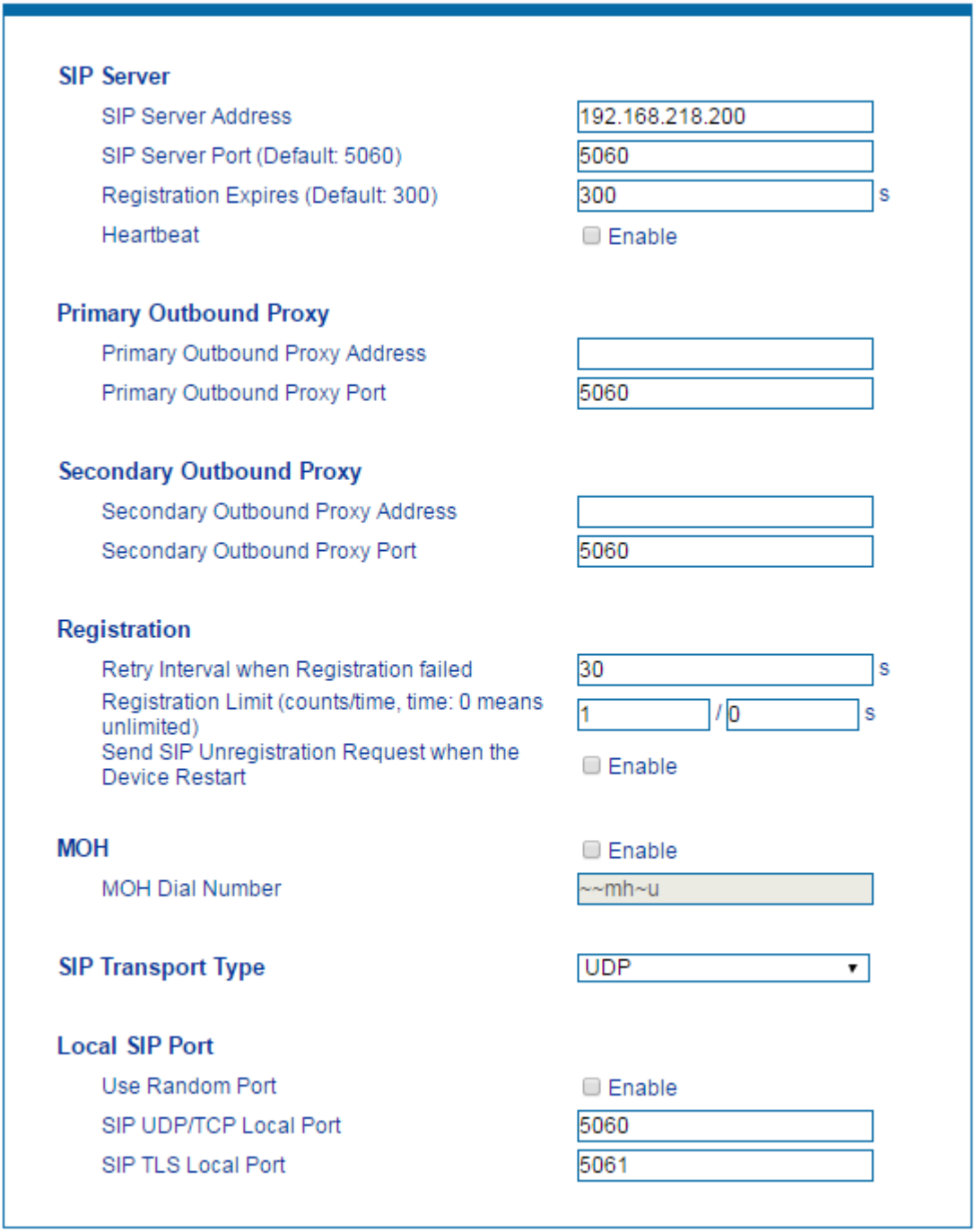


Figure 3.8-1 Configuration Interface for SIP Server

Explanation for SIP parameters:

Primary SIP Server Address	The IP address or domain name of the primary SIP server. They are provided by VoIP service provider.
Primary SIP Server port	The Service port of the primary SIP server. It is 5060 by default.
Registration Expires	It is used to avoid excessively frequent registrations. When the time that is set expires, terminals will send register request to the

	primary SIP server. The time is 1800s by default.
Heartbeat	Heartbeat is used to check the connection between terminal and SIP server.
Secondary SIP Server address	The IP address or domain name of the backup SIP server. They are provided by VoIP service provider.
Secondary SIP Server port	Service port of the backup SIP server. It is 5060 by default.
Registration Expires	It is used to avoid excessively frequent registrations. When the time that is set expires, terminals will send register request to the backup SIP server. The time is 1800s by default.
Secondary SIP heartbeat	Heartbeat is used to check the connection between terminal and SIP server.
Outbound Proxy Address	Outbound proxy IP address or domain name provided by VoIP service provider.
Outbound Proxy Port	Default outbound proxy port is 5060.
Retry Interval when Registration failed	The retry interval time after a registration fails. Default: 30s
Registration times per second	The maximum number of registrations in a second. 0 means no limitation for registrations.
SIP Transport Type	The way of SIP-based transmission. It can be UDP, TCP and Auto. Default: UDP.
Use Random Port	The SIP port for providing services for terminal is chosen by random.
SIP Local Port	Default SIP local service port is 5060.

3.6 IP Profile

On the following interface, you can configure IP profiles which can be used on the Port configuration, IP trunk configuration as well as call &routing configuration.

IP Profile - Add

Index	1
Description	

SIP Server

SIP Server Address	192.168.218.200
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 (Default: 5060)	5060

Secondary Outbound Proxy

Secondary Outbound Proxy Address	
Secondary Outbound Proxy Port (Default: 5060)	5060

MOH

	Enable
MOH Dial Number	~~mh~u

Digit Map

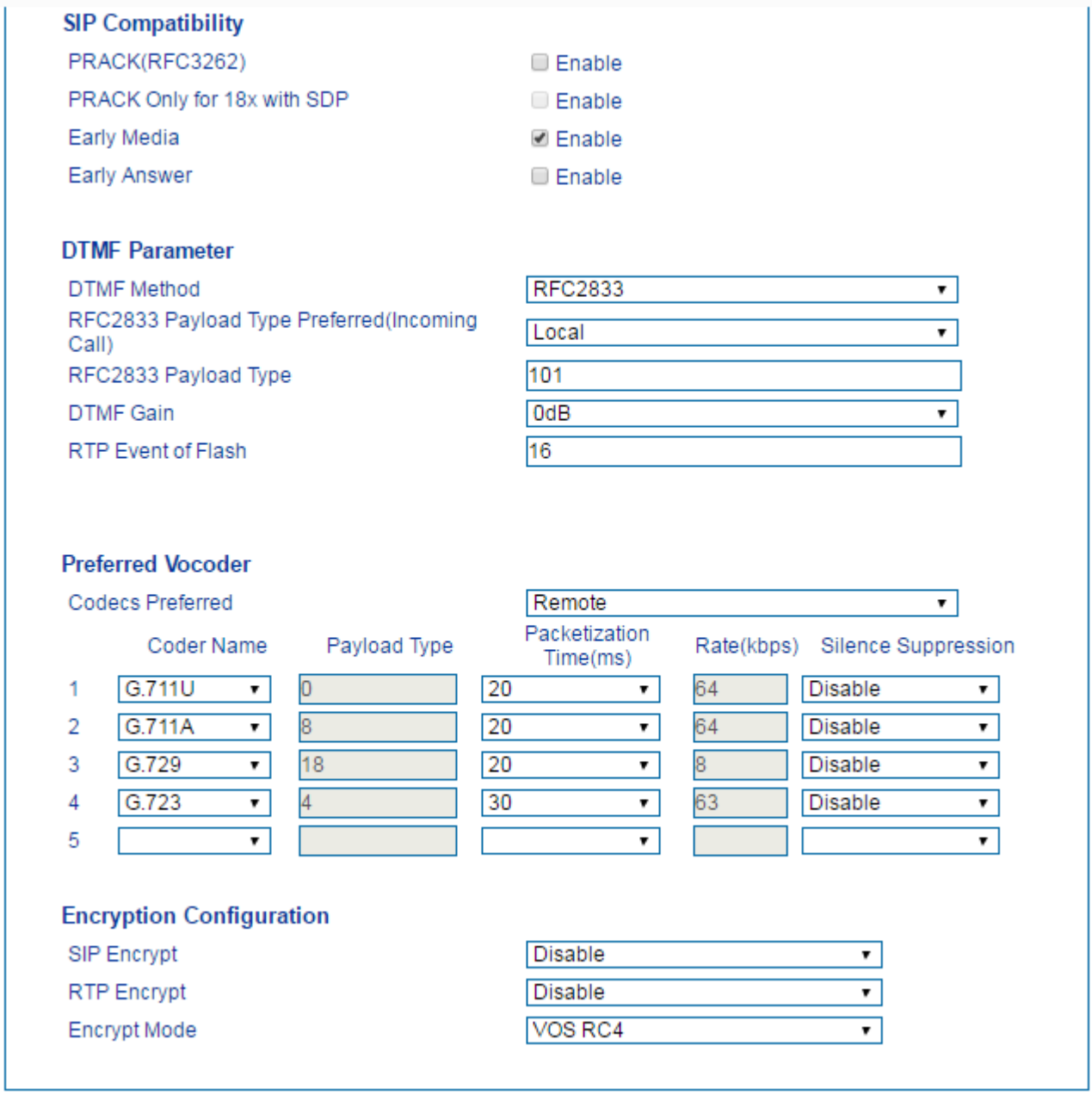
Match Failed(When the registration is successful)	Send to the server

Digit Map

```
[*#]T|[*#][*#]|*x.T|**x.#|[*#]xx#|*#xx#|[*#][0-9*#]x[0-9*].x#|x.#|x.T
```

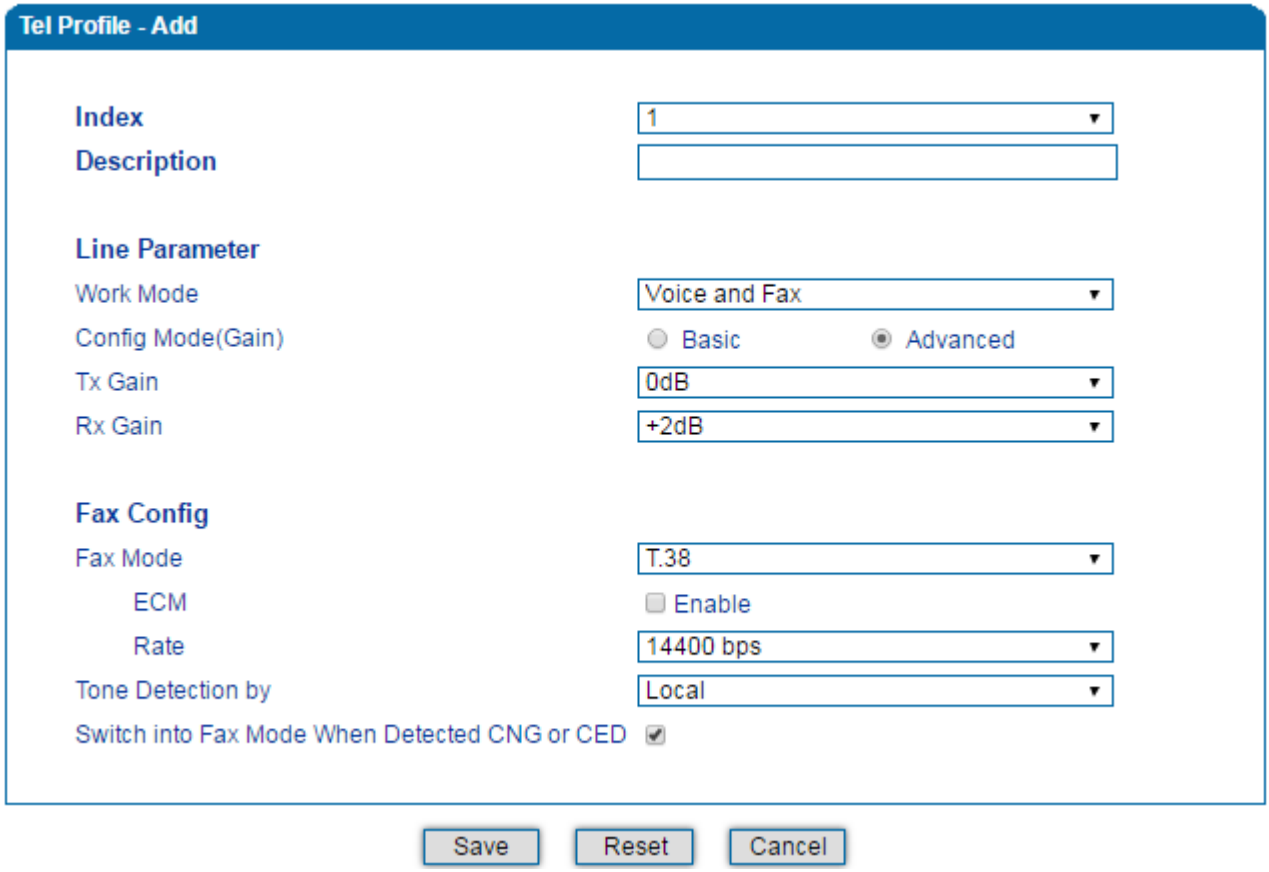
Service Parameter

Echo Cancel Tail	64 ms



3.7 TEL Profile

On the following interface, you can configure Tel profiles which can be used on the Port configuration, IP trunk configuration as well as call &routing configuration.



3.8 Port

On this interface, you can configure the port information.

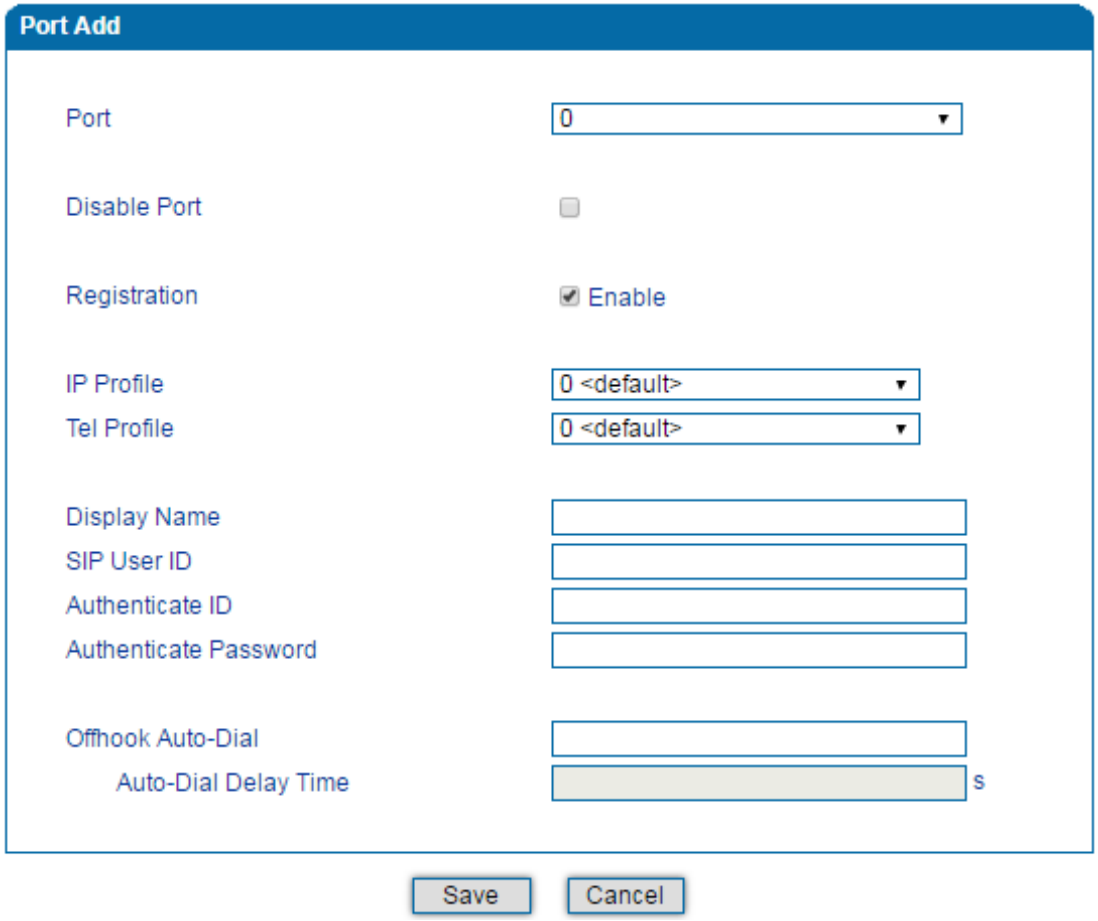


Figure 3.9-1 Port Configuration Interface

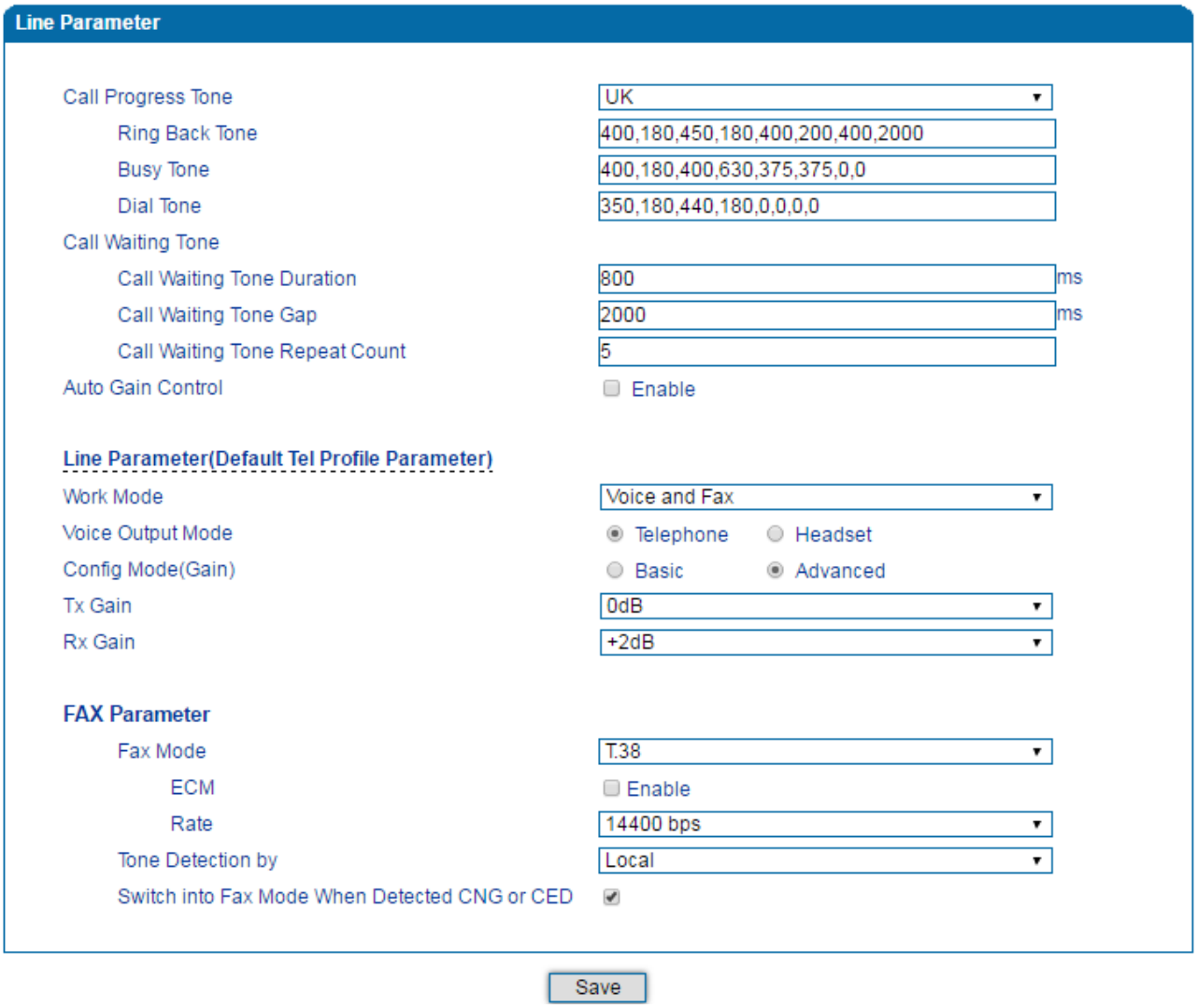
Explanations for port parameters:

Port	Port number
Disable port	Whether to disable port temporally
Registration	Whether to enable registration for the port
IP Profile	Choose an IP profile
Tel Profile	Choose a Tel profile
Display name	It is used to identify the SIP account
SIP User ID	User account information provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's authenticate ID used for authentication. It can be identical to or different from SIP User ID.
Authenticate password	SIP password which registers to soft switch/SIP server

Offhook 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.

3.9 Advanced

3.9.1 Line Parameter



3.9.2 FXO Parameters

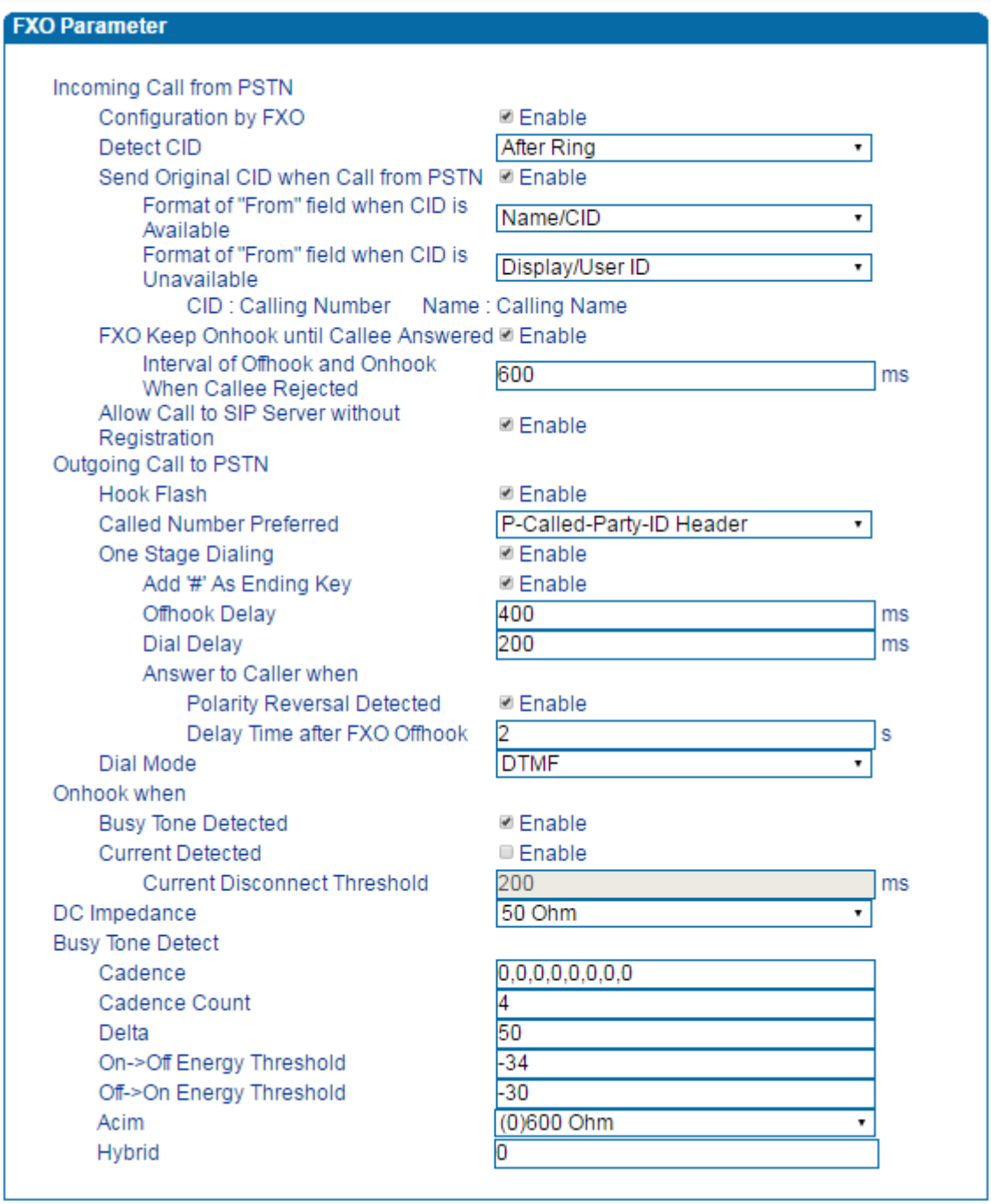


Figure 3.10-1 Configuration Interface for FXO Parameters

3.9.3 Media Parameter

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

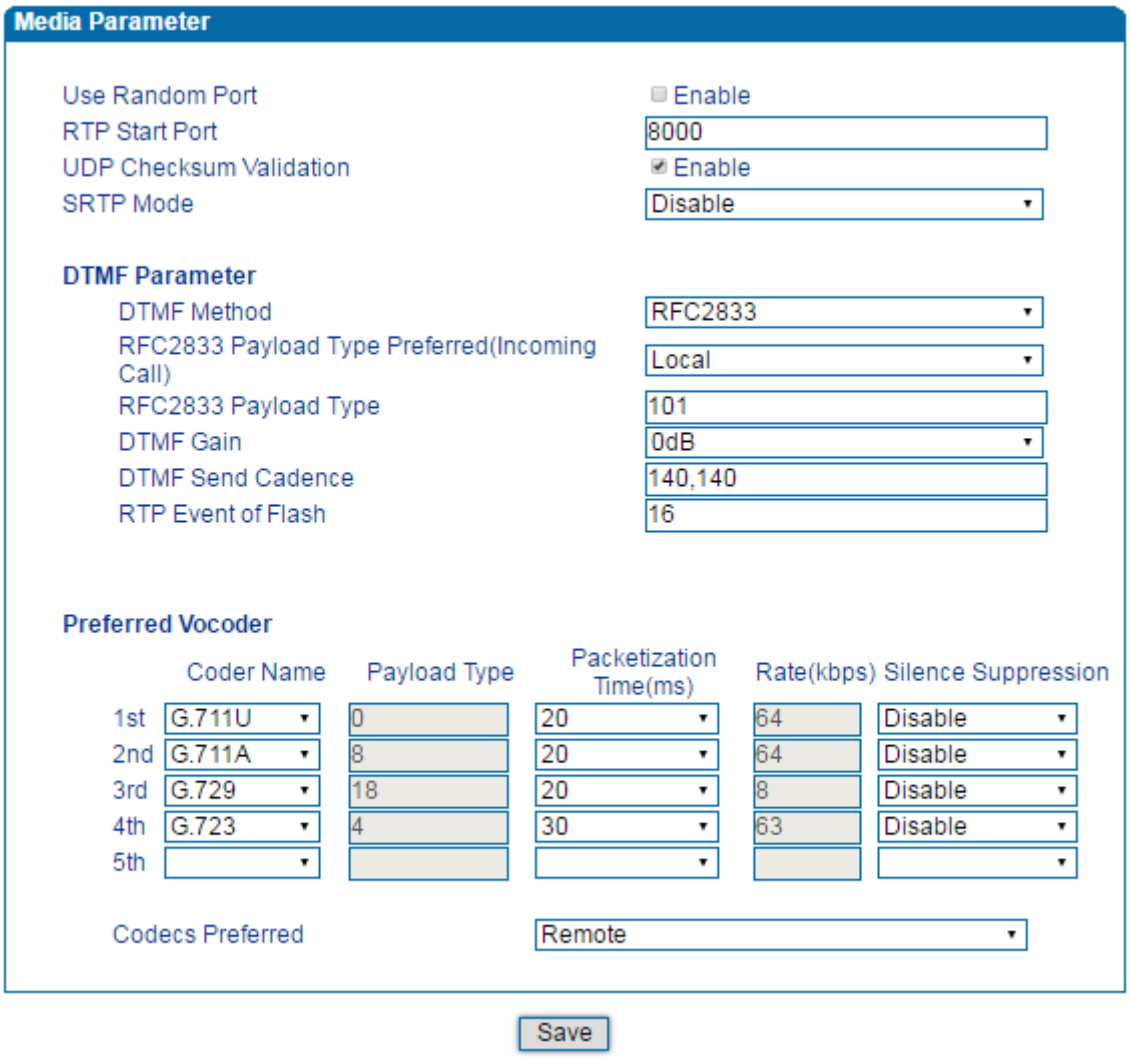


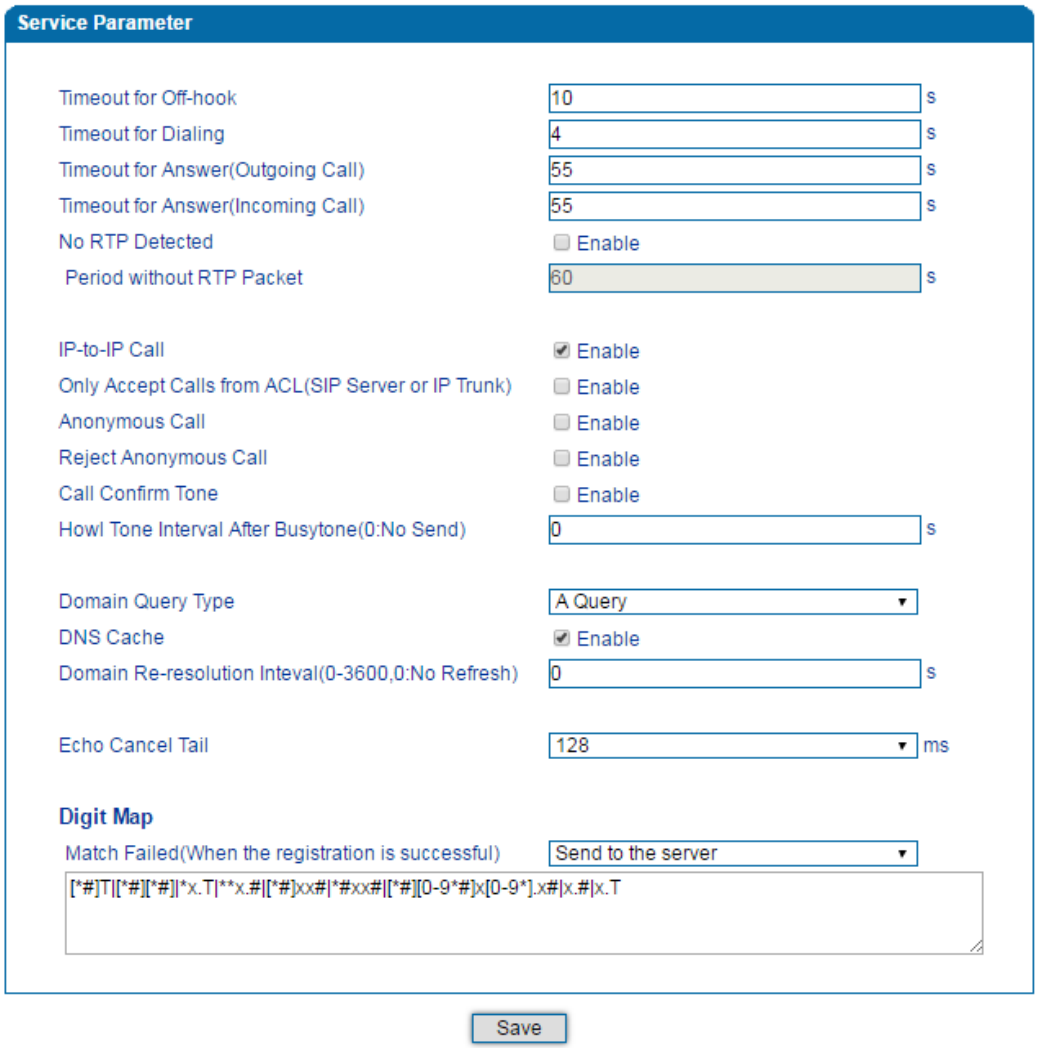
Figure 3.10-2 Configuration Interface for Media Parameters

Explanation of media parameters:

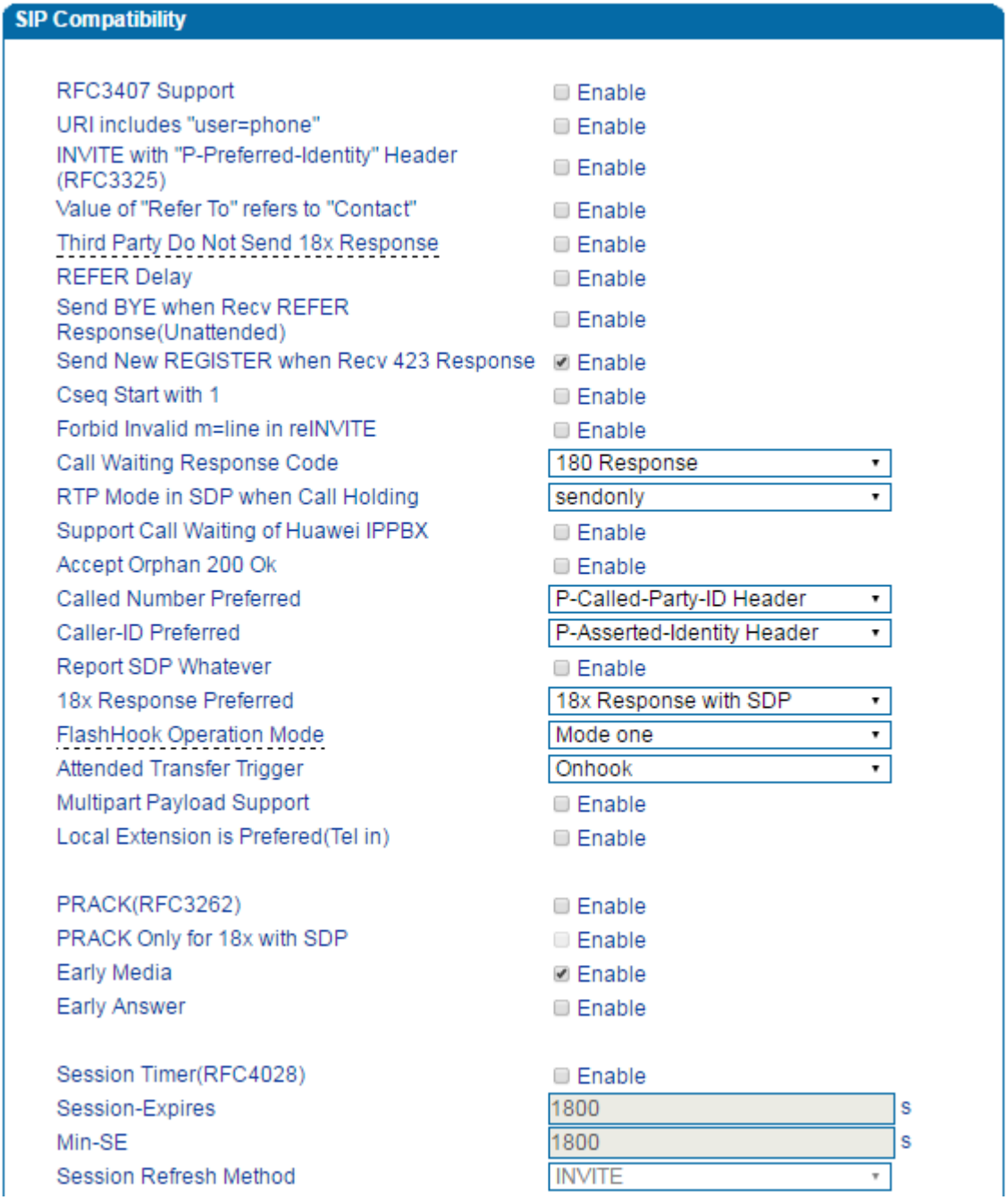
Use Random Port	If this parameter is enabled, the gateway will choose a port by random as the start port for RTP.
RTP Start Port	Default RTP start port is 8000
DTMF Method	Include SINGAL, INBAND and RFC2833
RFC2833 Payload Type	Payload value, default value is 101
DTMF Gain	Default value is 0 DB
DTMF Send Interval	The interval for sending DTMF signal. The default value is 200ms.
Send Flash Event	If this parameter is enabled, the gateway will send flash event to remote terminal, and thus user does need to handle it locally

Coder Name	The gateway supports G729, G711U, G711A and G723. 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.

3.9.4 Service Parameter



3.9.5 SIP Compatibility



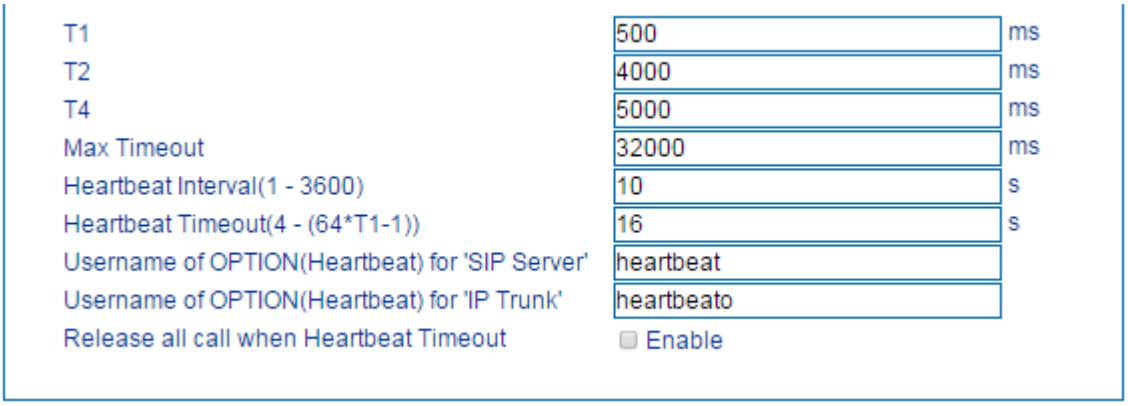


Figure 3.10-3 SIP Parameter Configuration Interface

Explanation of SIP parameters:

SUBSCRIBE for MWI (Message Waiting Indicator)	Whether to enable 'voicemail message waiting indicator'; it is realized in the way of NOTIFY
MWI Subscription Expires	MWI subscription expiry time; Default value is 3600s.
Voicemail User ID	The user ID for access to voicemail box
RFC3407 Support	Whether to enable RFC3407 support.
IP-to-IP Call	If this parameter is enabled, user can dial IP address through a phone to call destination gateway.
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).
Only Accept Call from ACL (SIP server or IP Trunk)	If this parameter is enabled, the gateway 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 value is 'not disable'.
# as ending Dial Key	'#' is used as the end mark for dialing.
# Escape	If this parameter is enabled, '#' is considered as a digit of the number that is dialed.
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	If this parameter is enabled, the third party will not send 18x response

Response	during a 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.
Implicit Subscribe	If this parameter is enabled, the gateway will accept implicit subscription.
CSeq Start with 1	If this parameter is enabled, the value of CSeq starts with '1'.
Forbid Invilad m=line in reINVITE	If this parameter is enabled, the gateway will prevent 'invalid m=line' from being carried in the SDP of re-INVITE.
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 gateway will support call waiting of Huawei IPPBX.
Accept Orphan 200 OK	If this parameter is enabled, the gateway will support different 'to-tag 200 OK' in a INVITE session
Domain Query Type	There are two modes: A QUERY and SRV QUERY. Default is 'A QUERY'.
Domain Re-resolution Interval	Default 0: forbidden
DNS cache	If this parameter is enabled, the gateway will cache the DNS query results.
Early Media	Support the receiving of Early Media.
PRACK(RFC3262)	Support reliable transmission of provisional response
PRACK Only for 18x with SDP	Send PRACK only when there's SDP in 18x response
Early Answer	If this parameter is enabled, SDP will be contained in 18x
Session Timer (RFC4028)	Whether to enable 'session timer', default value is ' no'.
Session-Expires	The Session-Expires header field conveys the session interval for a SIP session.
Min-SE	Min-SE header field indicates the minimum value for the session interval.
T1	T1 timer of SIP protocol, default is 500ms
T2	T2 timer of SIP protocol, default is 400ms
T4	T4 timer of SIP protocol, default is 500ms

Max Timeout	The max timeout of sending or receiving, default is 32s
Heartbeat Interval	Default is 10s.
Heartbeat Timeout	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

3.9.6 NAT Parameter

If NAT Traversal is enabled, the IP address of SIP extension in LAN (local-area network) will be turned into the outbound IP address of the public network, thus making NAT traversal possible.

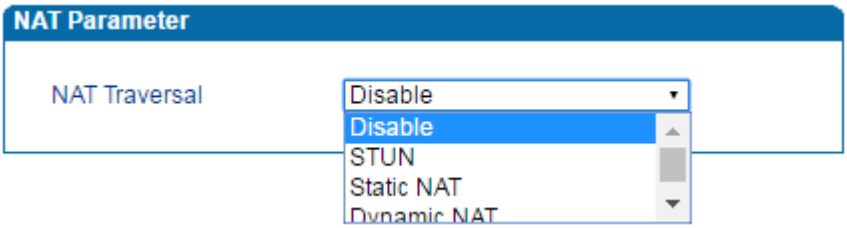


Figure 3.10-4 Configuration Interface for NAT Parameter

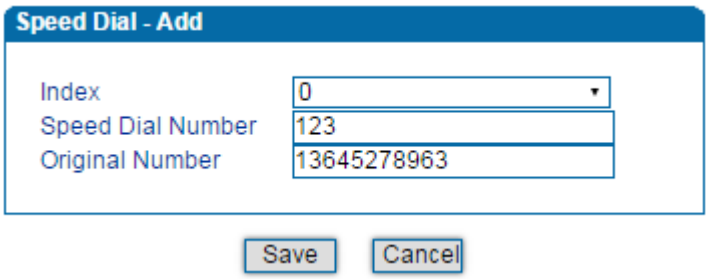
STUN（Simple Traversal of UDP over NATs）: STUN is a lightweight protocol that allows applications to discover the presence and types of NATs as well as firewalls between them and the public Internet. It also provides the ability for applications to determine the IP addresses allocated to them by the NAT.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

Static NAT: the IP address of the device in the LAN will be changed into the outbound IP address of the public network which is a static IP. The NAT IP will be carried in the SIP messages sent by the device afterwards. If you selected static NAT, you need to enter an IP address (that is the outbound IP address of the public network) manually.

Dynamic NAT: the IP address of the device in the LAN will be changed into the outbound IP address of the public network that is a dynamic IP. The device will check the dynamic IP automatically and will update the IP in the SIP messages sent by itself.

3.9.7 Speed Dial

This function is used under the "two-stage dialing" application scenario. When an inbound call comes into the FXO port of the DAG device, you will hear a PSTN's dialing tone and then you can dial the speed-dial number to let the call goes out. In this case, the original number is replaced by the speed-dial number to save time and trouble.



3.9.8 Feature Code

Please make reference to the Feature Codes interface and the following table.

Inquiry LAN port IP address	Dial*158# to obtain device WAN port IP address
Inquiry WAN port IP address	Dial*159# to obtain device WAN 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.
Network Work Mode	*157*0#, set network work mode to routing mode; *157*1#, set network work mode to bridge mode
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
Renew DHCP	*193#, set dynamic IP again
Access Web by Wan in Rout Mode	Allow access web through WAN port：*160*1#; don't allow access web through WAN port：*160*0#
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 Activate	*72*+ phone number#, transfer the call from the phone 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

3.9.9 System Parameter

System parameters include: STUN, NTP, Provision, EB parameter and Telnet.

1）STUN：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.

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

3）Provision：provision is used to make the gateway automatically upgrade with the latest firmware stored on an http server an ftp server or a tftp server.

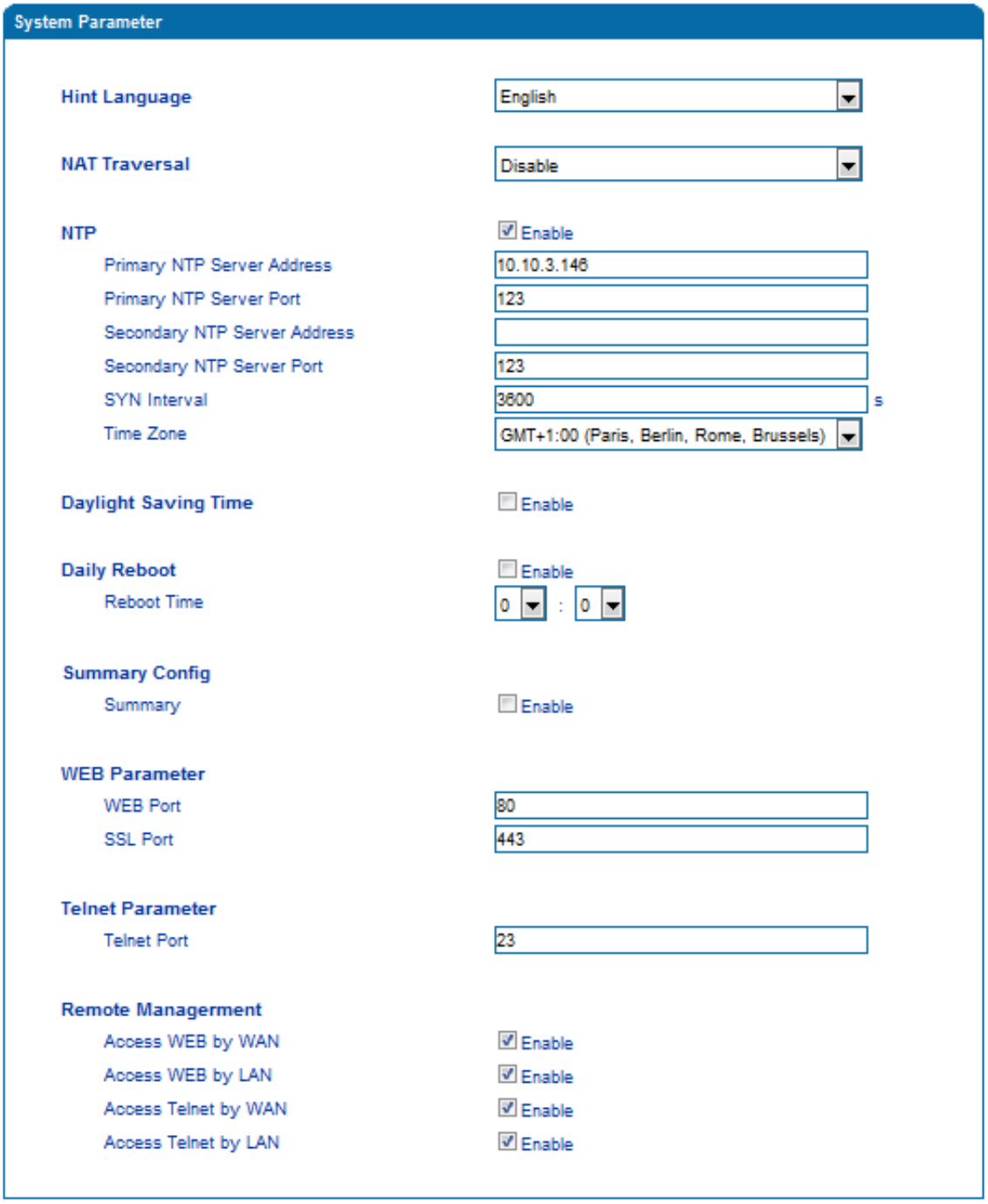


Figure 3.10-7 Configuration Interface for System Parameters

Explanation for related parameters:

Hint Language	IVR language of the gateway
NAT Traversal	User can choose ‘Disable’, ‘ STUN’, ‘static NAT’ and ‘dynamic NAT’.
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 18.145.0.30
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 DAG FXO. Default value is 3600s.
Time Zone	The time zone of the gateway; Default configuration is United States central time, Chicago.
Daylight Saving Time	Enable or disable daylight saving time
Daily Reboot	Whether to enable daily reboot
Reboot time	The time to reboot the gateway daily
WEB Port	The web port of the gateway; Default port is 80
Telnet port	Listening port of telnet service; Default port is 23
Access WEB by WAN	Enable or disable 'Access web service from WAN'
Access WEB by LAN	Enable or disable 'Access web service from LAN'
Access Telnet by WAN	Enable or disable 'telnet service from WAN'
Access Telnet by LAN	Enable or disable 'telnet web service from LAN'

3.10 Call & Routing

3.10.1 Wildcard Group

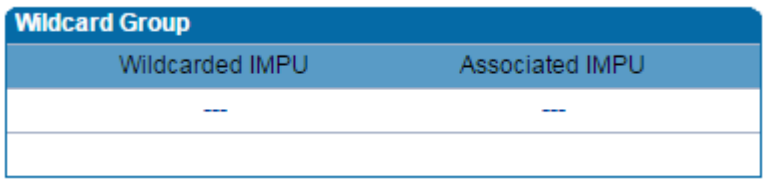


Figure 3.11-1 Wildcard Group

3.10.2 Port Group

On the **Port Group** interface, you can group several ports together and then set a strategy for port selection of the group. 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 and so on.

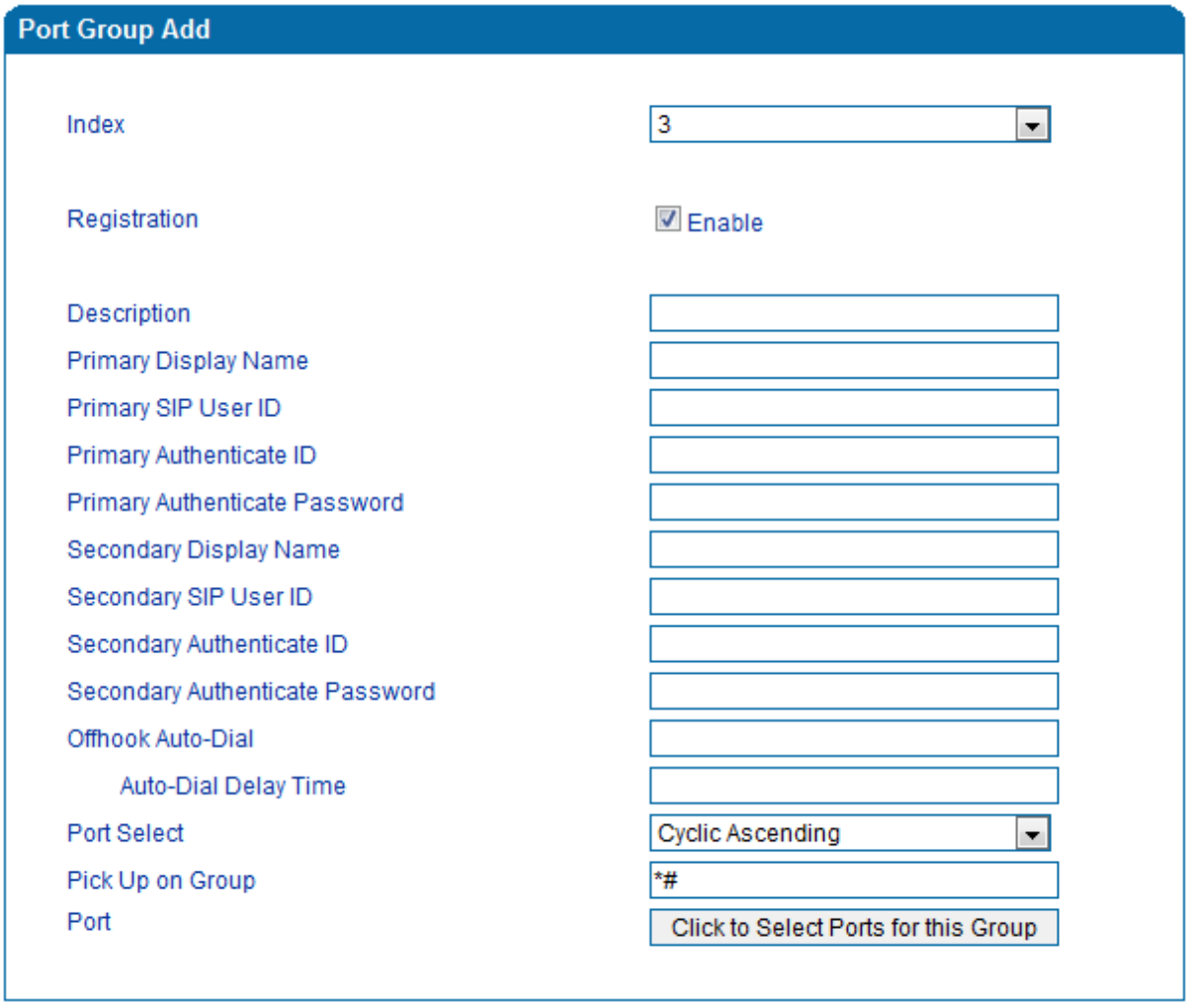


Figure 3.11-2 Configuration Interface for Port group

Explanation of related parameters

Index	The NO. of the port group ; It uniquely identifies a route, range from 0-7
Description	The description of the port group; it is used to identify the port group
Primary/Secondary Display Name	Port group display, 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
Primary/Secondary SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Primary/Secondary Authenticate ID	SIP service subscriber's authentication ID, it can be identical to or different from SIP User ID.
Primary/Secondary Authenticate Password	Password of SIP user ID
Offhook Auto-Dial	To enter offhook auto-dial number
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 gateway always selects a port from the minimum number. • Cyclic ascending: the gateway 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 gateway always selects a port from the maximum number. • Cyclic descending: the gateway 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
Pickup UP on group	When one port rings, user can dial '*#' to pick up the call from other ports under the same port group.
Port	Select ports for this port group

3.10.3 IP Trunk

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

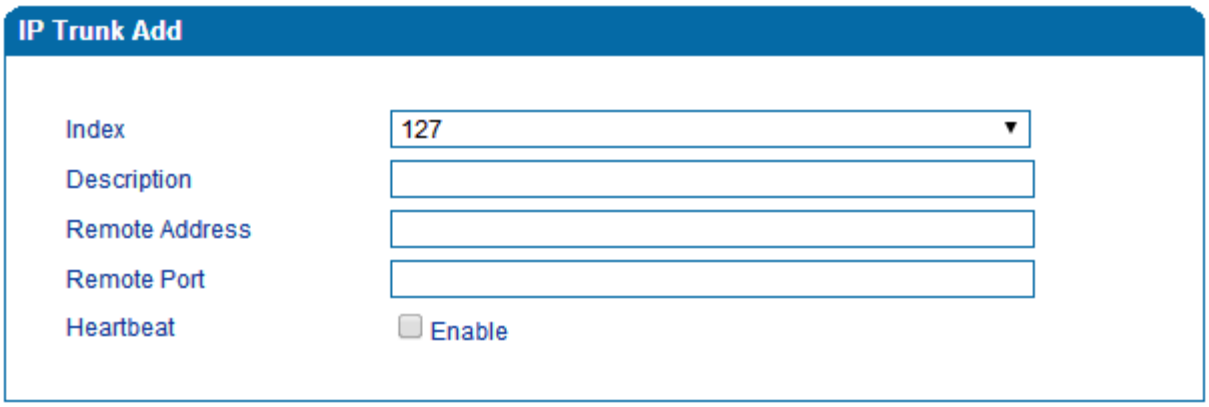


Figure 3.11-3 IP Trunk Configuration Interface

Explanation of related parameters:

Index	The No. of the IP trunk; 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 peer device
Remote Port	SIP port of peer device
Heartbeat	Whether to enable the 'Heartbeat' function for the IP trunk. Default value is ' not enable'. If heartbeat is enabled, the gateway will send "OPTION" to peer device.

3.10.4 Routing Parameter

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

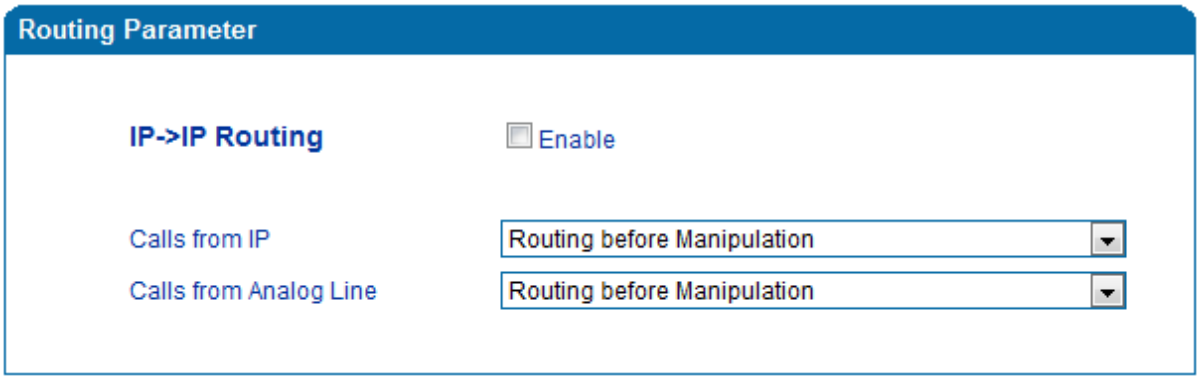


Figure 3.11-4 Configuration Interface for Routing Parameter

3.10.5 IP -> Tel Routing

Calls from the IP network to the GSM network is defined as IP → Tel calls, and those calls need to be routed by IP → Tel routing.

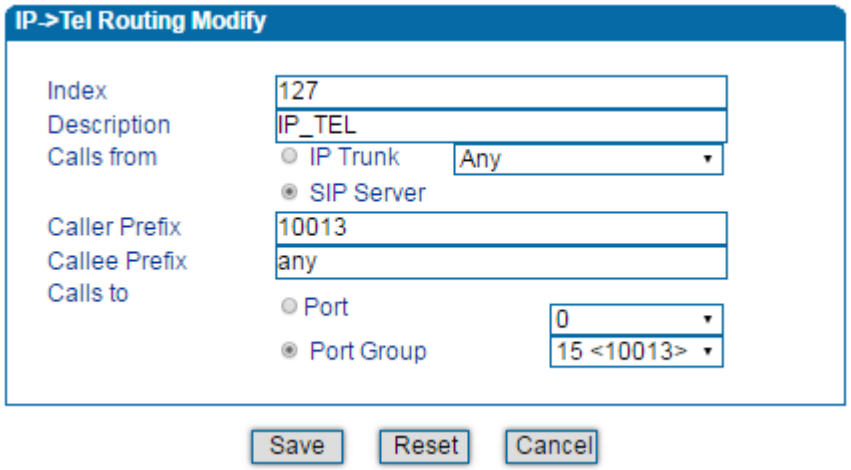


Figure 3.11-5 Configuration Interface for IP-Tel Routing

Explanation of related parameters:

Index	IP →Tel Routing priority: from 0 to127; 0 is the highest priority.
Description	It is used to identify the IP → Tel routing
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

3.10.6 Tel-IP/Tel Routing

Calls from the GSM network to the IP network are defined as Tel → IP calls, and those calls need to be routed by Tel → IP routing. Calls from a GSM network to an onother GSM network are defined as Tel → Tel calls, and those calls need to be routed by Tel → Tel routing.

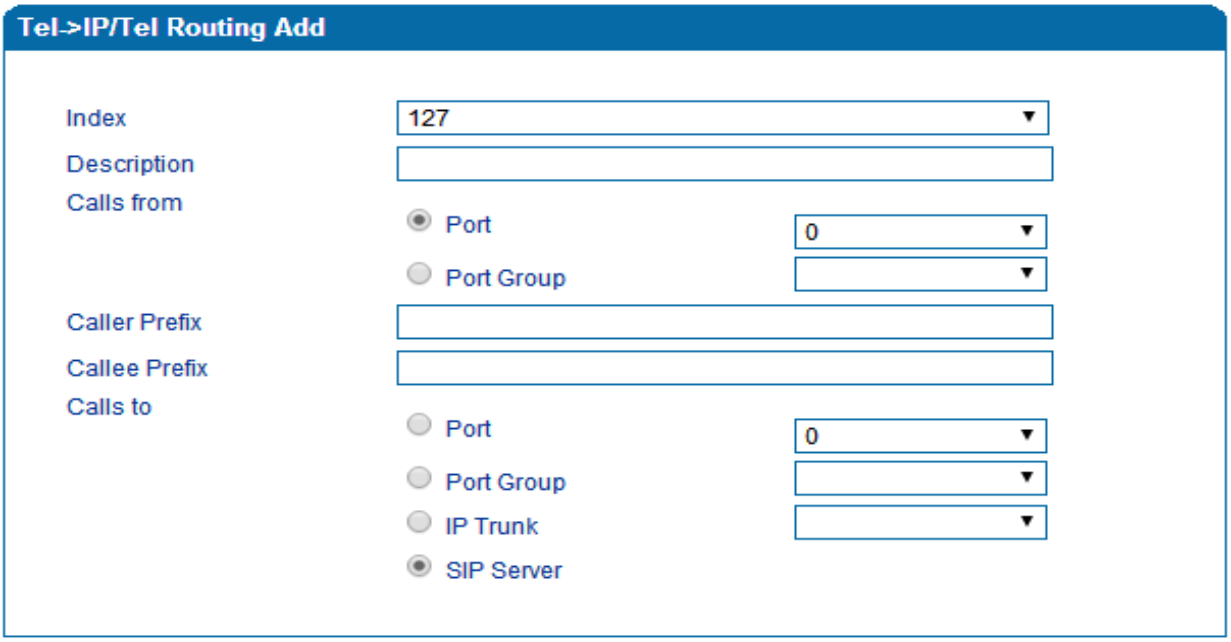


Figure 3.11-6 Configuration Interface for Tel-IP/Tel Routing

Explanation of related parameters:

Index	The index of this Tel →IP/Tel routing, from 0 to 127. Each index cannot be used repeatedly. Routing priority: 0 is the highest priority.
Description	It is used to identify the routing
Calls From	Tel →IP 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	Calls are routed to a port, port group, IP trunk or SIP server

Internal Call Function: When there is no SIP server for registration, calls can be given from one FXO port to another FXO port of a same access gateway of DAG1000, DAG2000, DAG2500 and DAG3000.

3.11 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.

3.11.1 IP -> Tel Callee

On the following interface, you can set a manipulation rule to change callee numbers of IP → Tel calls.

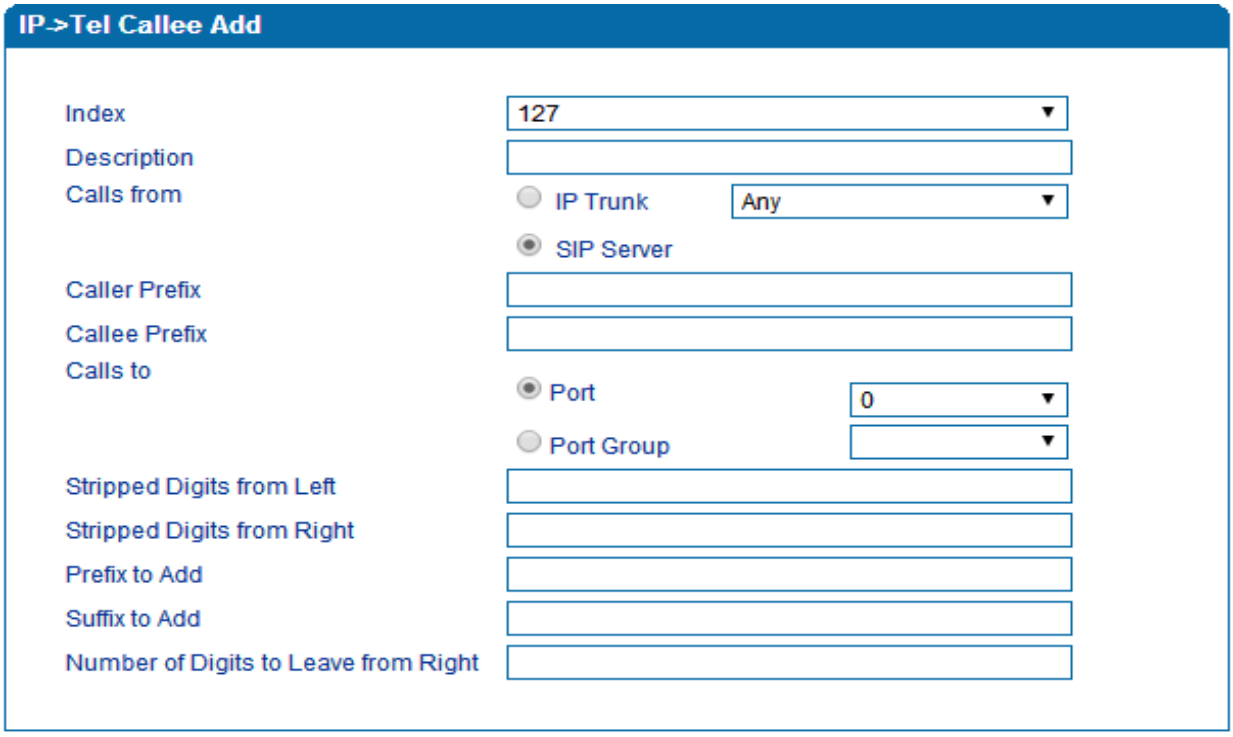


Figure 3.12-1 Add IP -> IP Callee

Index	The index of this manipulation, from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority
Description	Name of this IP ->Tel manipulation name
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 routing. 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 routing. If called number is 008675526456659, the called prefix can be 0086755 or 00., "any" means match any called number
Calls to	Determine the port or port group to which the call is routed.
Stripped Digits from Left	The number of digits which are lessened from the left of the callee number

Stripped Digits from Right	The number of digits which are lessened from the right of the callee number
Prefix to Add	The prefix added to the callee number after its digits are lessened.
Suffix to Add	The suffix added to the callee number after its digits are lessened.
Number of Digits to Leave from Right	The number of the retained digits which. are counted from the right of the callee number

3.11.2 Tel -> IP/Tel Caller

On the following interface, you can set a manipulation rule to change caller numbers of Tel → IP/Tel calls.

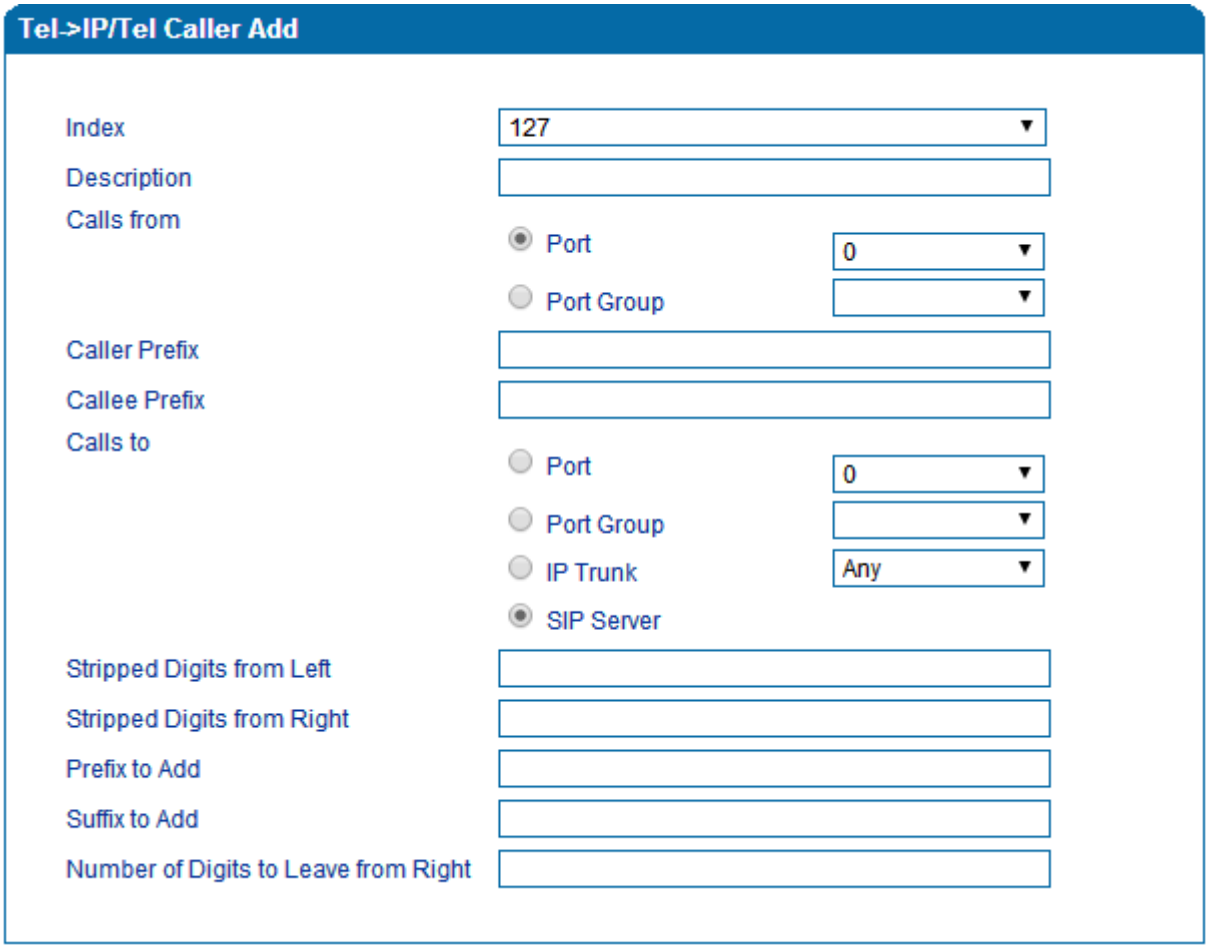


Figure 3.12-2 Add Tel -> IP Caller

Configuration parameters are the same with those of 'IP->Tel Callee'.

3.11.3 Tel-IP/Tel Callee

On the following interface, you can set a manipulation rule to change callee numbers of Tel → IP/Tel calls.

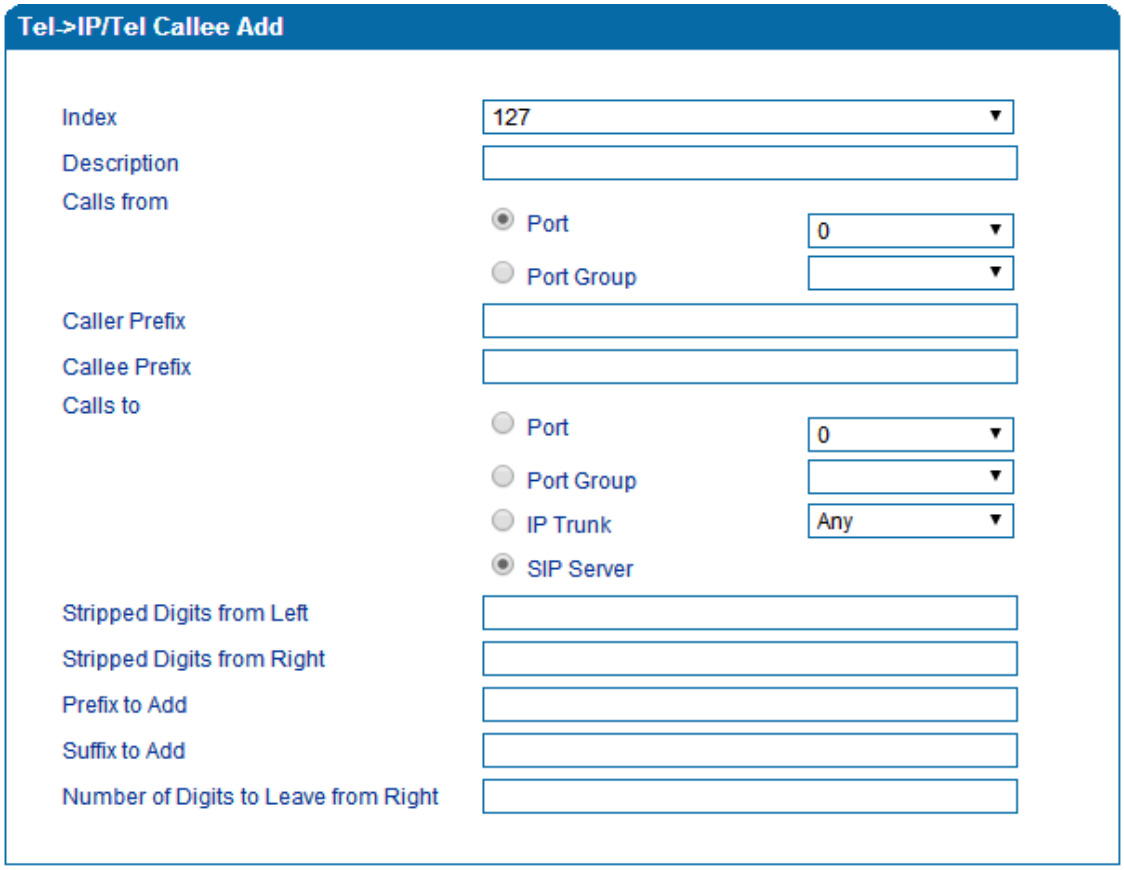


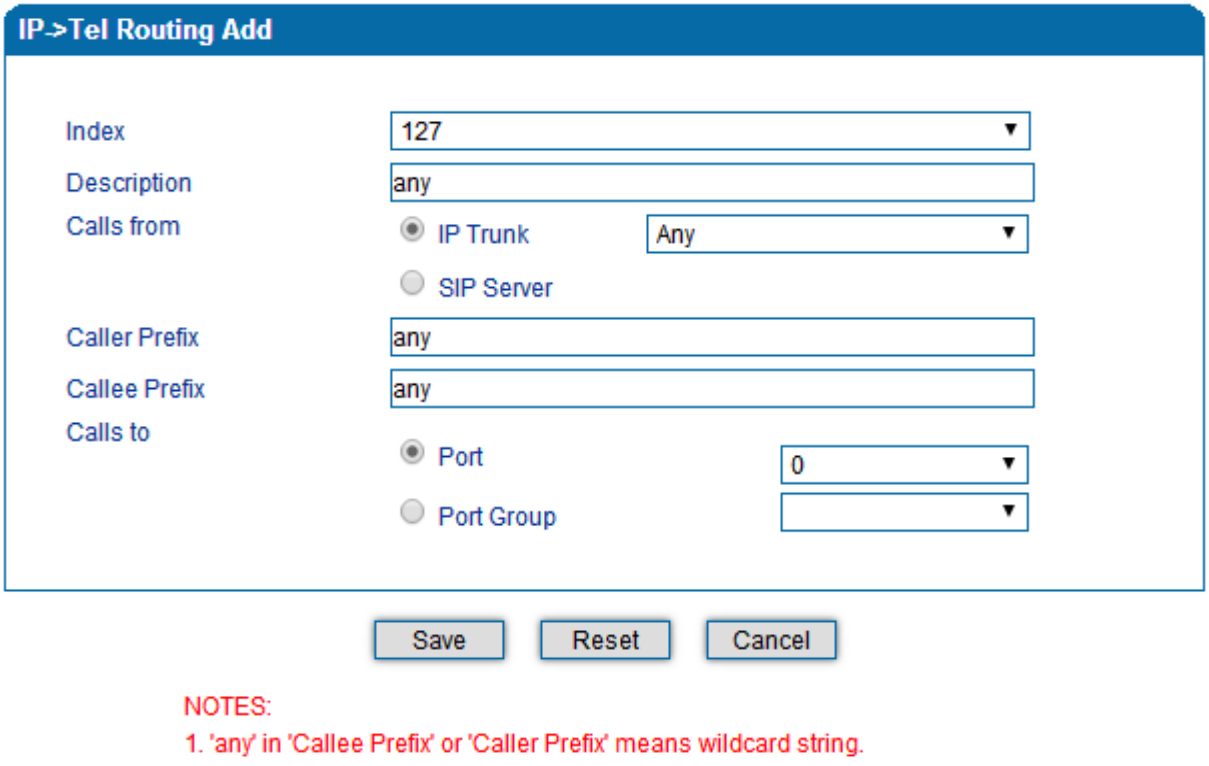
Figure 3.12-3 Add Tel-IP Callee

Configuration parameters are the same with those of 'Tel->IP Caller'.

3.12 Routing rule examples

3.12.1 Route any calls from any IP to specific port

After enter the Web interface, click **Call & Routing → IP-Tel Routing** in the navigation tree on the left, and then click **Add** to create a new routing rule.

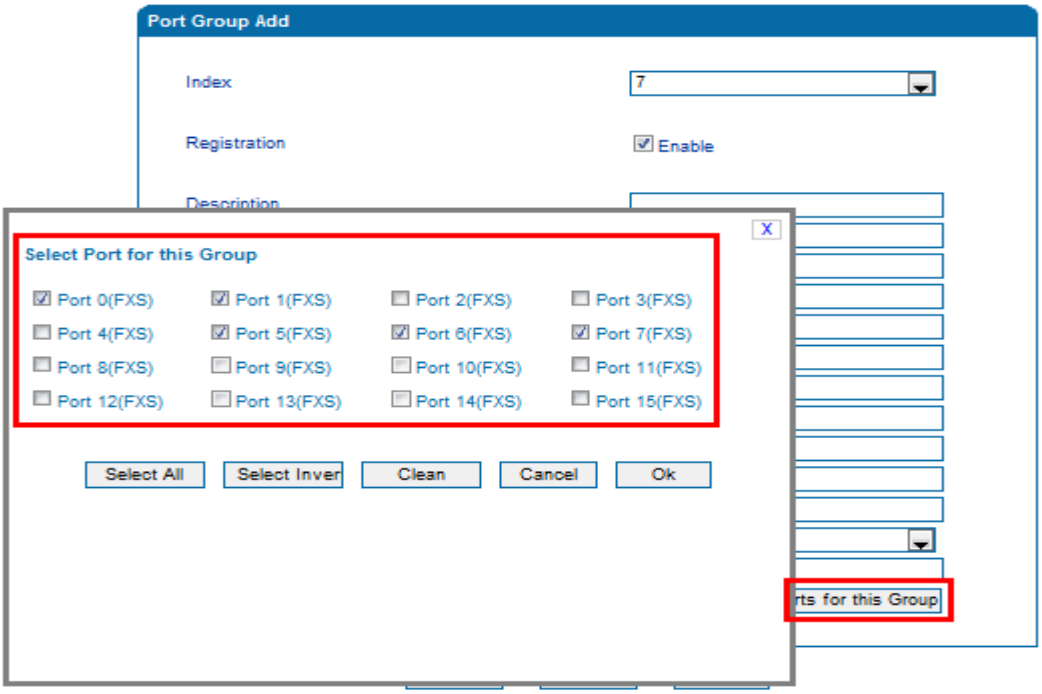


In the example above, all calls will be routed to port 0 when the routing rule is matched.

3.12.2 Route any calls from any IP to specified port group

- Create port group

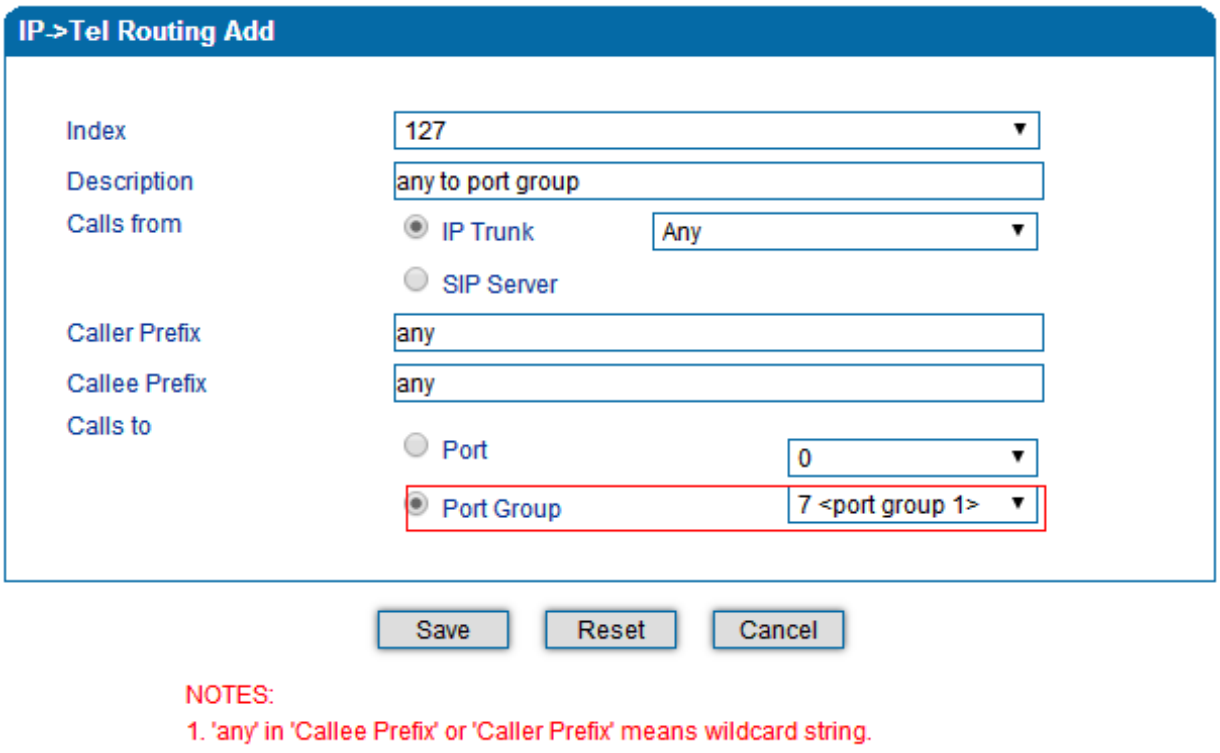
Before we can route calls to a port group, create the port group first as below. On the **Call & Routing → Port** Group, click **Add** to create a new port group, and then click **Click to Select Ports to This Group**.



Port 0, port 1, port 5, port 6 and port7 are assigned to port group 7.

- Route any calls to the port group

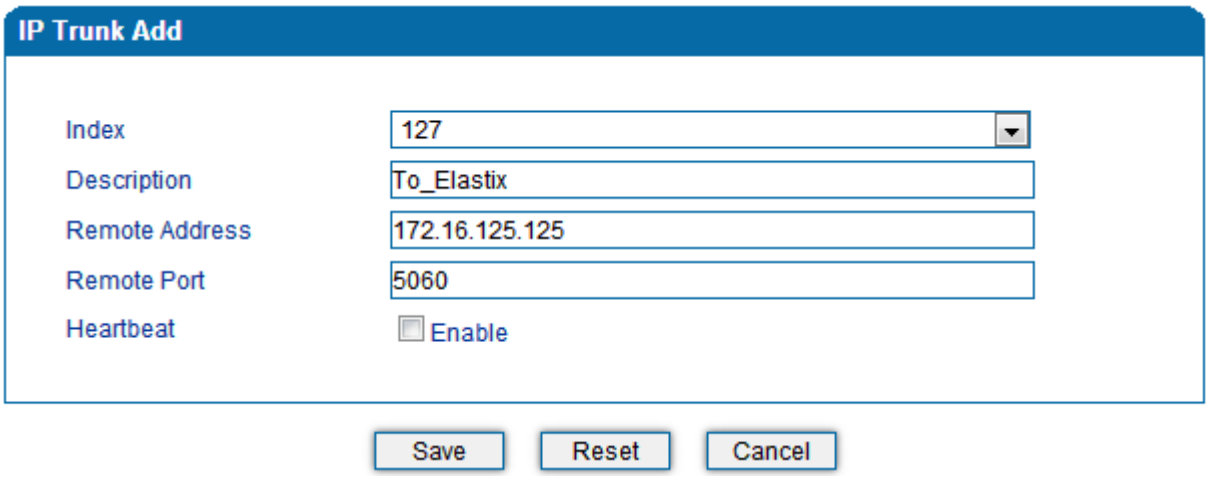
On the **Call & Routing → IP-Tel Routing** interface, click **Add** to create a new routing rule.



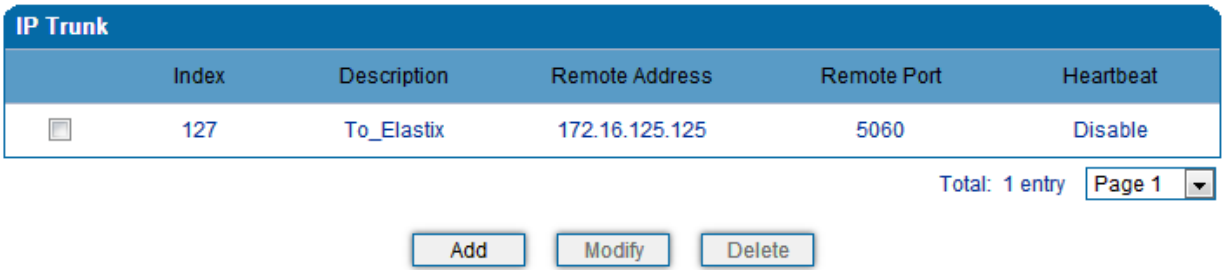
As shown above, if the routing rule is matched, calls will be routed to port group 7.

3.12.3 Route any calls from any port to specific SIP IP trunk

Create IP Trunk on the **Call & Routing → IP Trunk** interface:



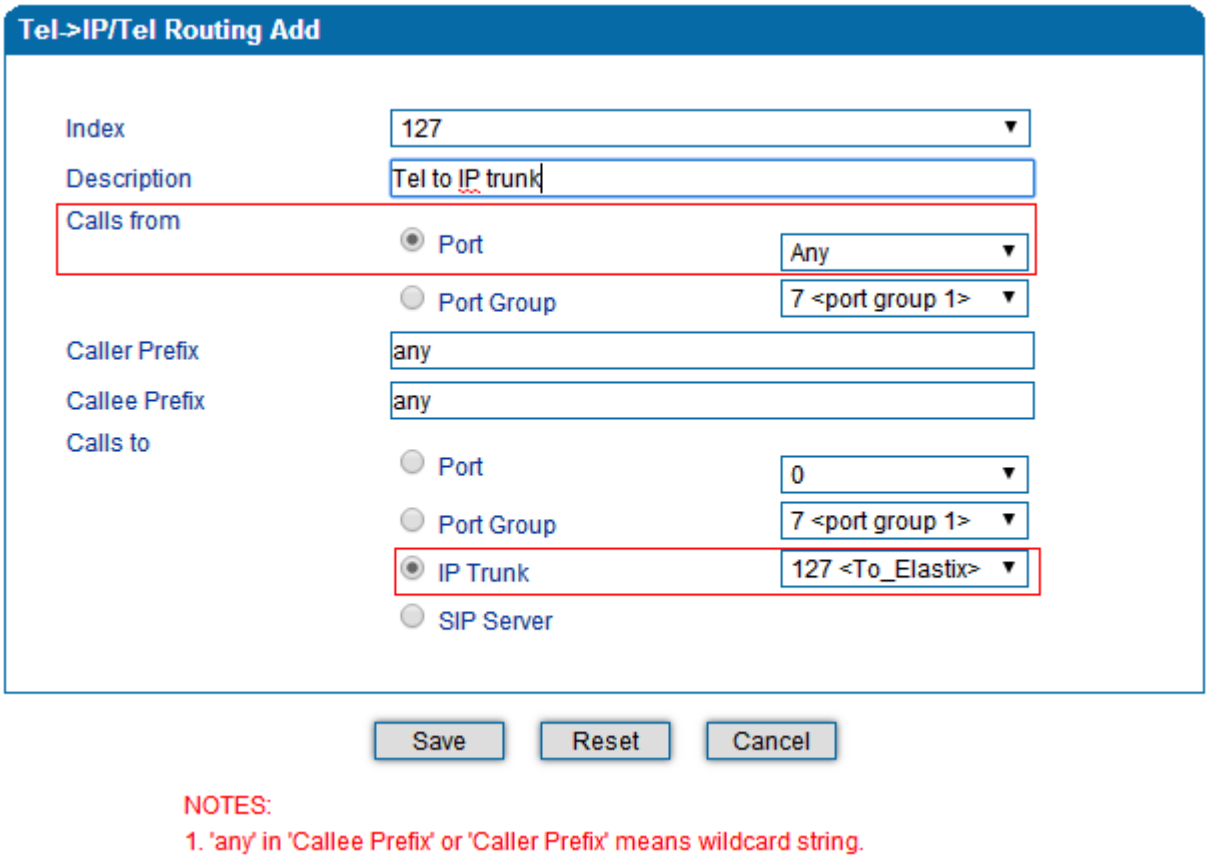
After IP Trunk is created, check the following configuration:



As shown above, the IP trunk is created, and the remote end IP address is 172.16.125.125, the SIP port is 5060.

Create Tel -> IP routing rule

On the **Call & Routing →Tel-IP Routing** interface, click “Add” to create a new Tel → IP routing rule.



All Tel calls from any caller number to any called number will be routed to IP trunk 127.

3.13 Management

3.13.1 TR069

ACS URL (auto-configuration server URL address) is provided by service provider. The ACS URL generally starts with http:// or https://, while username and password are used for ACS authentication.

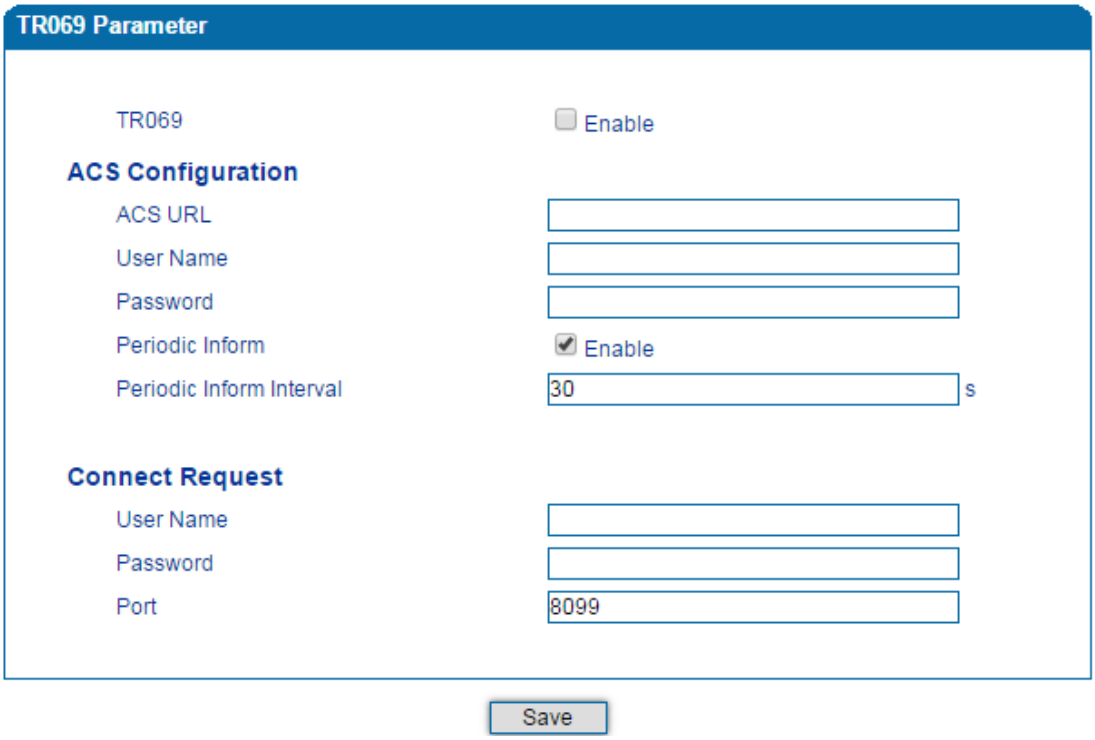


Figure 3.14-1 TR069 Parameters

3.13.2 SNMP (Simple Network Management Protocol)

SNMP Parameters:

- SNMP enable: to disable or enable the SNMP feature
- SNMP version: the DAG FXO gateway supports SNMP v1 and v2
- Community: the community name used to read through SNMP protocol
- Source: the IP address of SNMP server

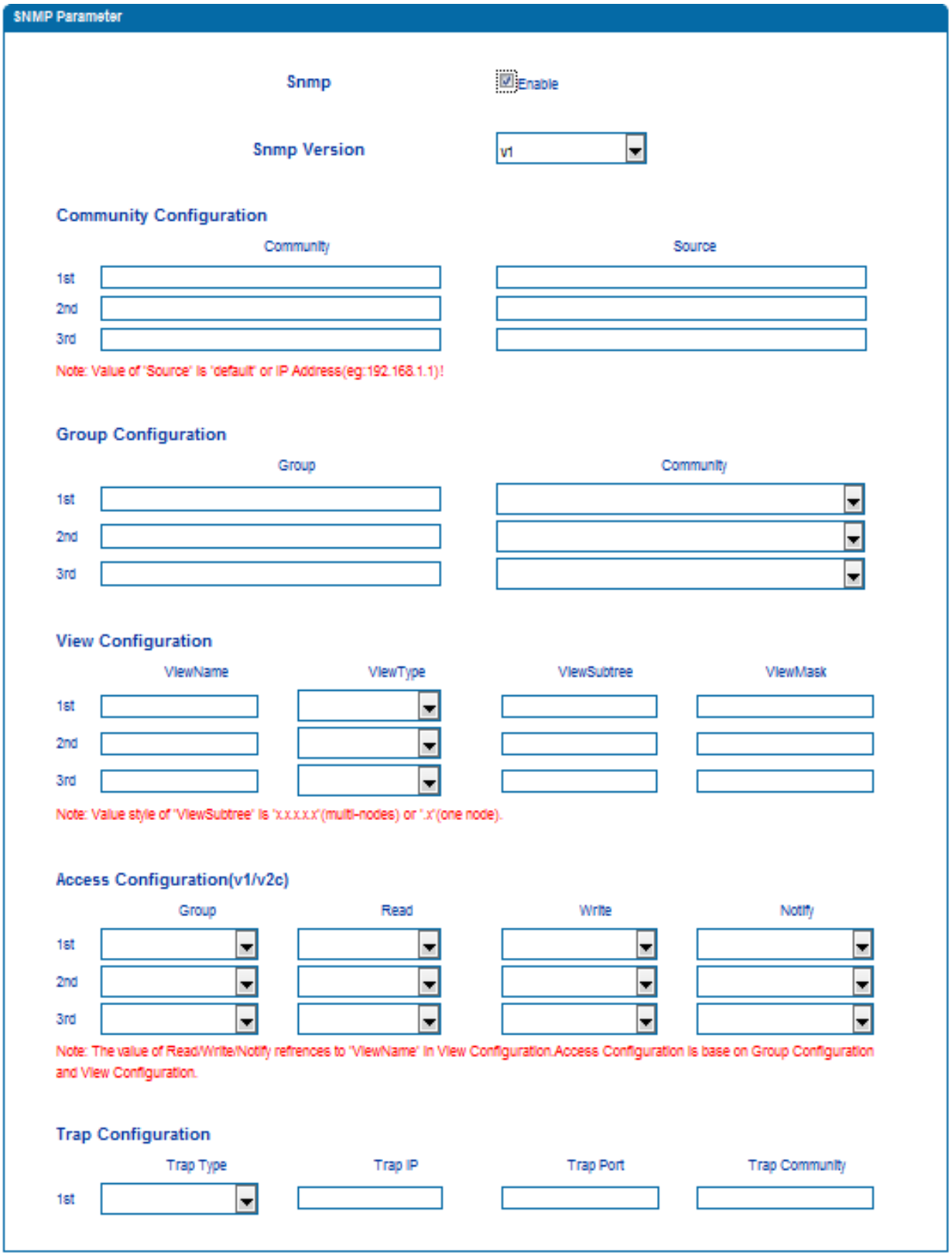
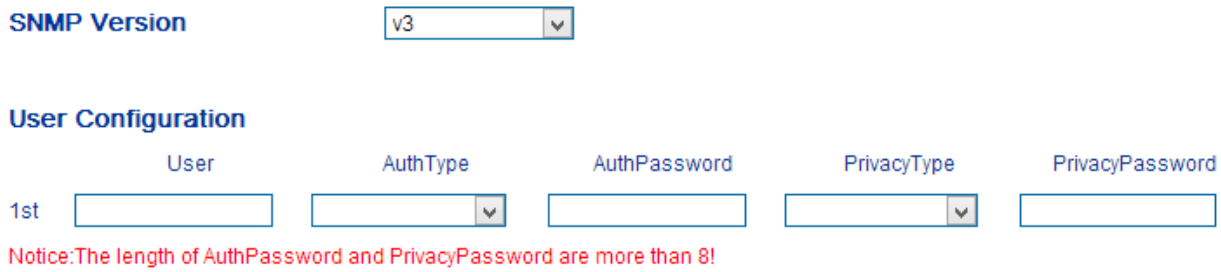


Figure 3.14-2 SNMP Parameters

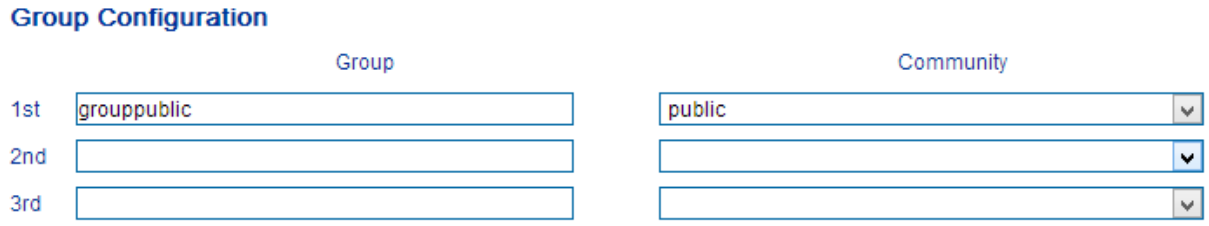
User configuration is only available on SNMP v3.



Group configuration

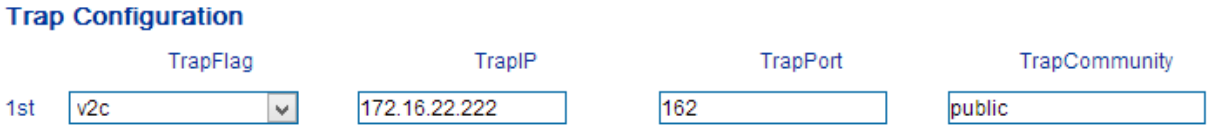
Group: community group name which consist of character string.

Community: let community join the community group which configured above



Trap configuration

Trap configuration enable to configure Trap server IP and port. This setting available for SNMP v2c and v1.



3.13.3 Syslog

Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators of problems or performance. There are 5 levels of syslog, Including NONE, DEBUG, NOTICE, WARNING and ERROR.

The Signal Log is include following traces which defined in system by default

- *SD, hardware debug*
- *SIP, SIP signaling trace*
- *STUN, STUN logs*
- *ECC, detail information of call control module*
- *RE, the common communication module for SCP and SIM*
- *SCP, the communication protocol between gateway and cloud server*

The media log is include following traces which defined in system by default

- *RTP, RTP stream info collection*
- *SIM, to output traces between gateway and remote SIM cards*

The System Log is include following traces which mainly used by developer

- *SYS, system log*
- *TIMER, system process*
- *TASK, system task process*
- *CFM, system process*
- *NTP*

The Management Log is include following traces which defined in system by default

- *CLI, command line*
- *TEL,*
- *LOAD, firmware upload*
- *SNMP*
- *WEBS, embedded web server*
- *PROV, provisioning*

Server Syslog:

When the gateway register to SIM Cloud server, the option will be changed to un-configurable and all logs to be storage on server.

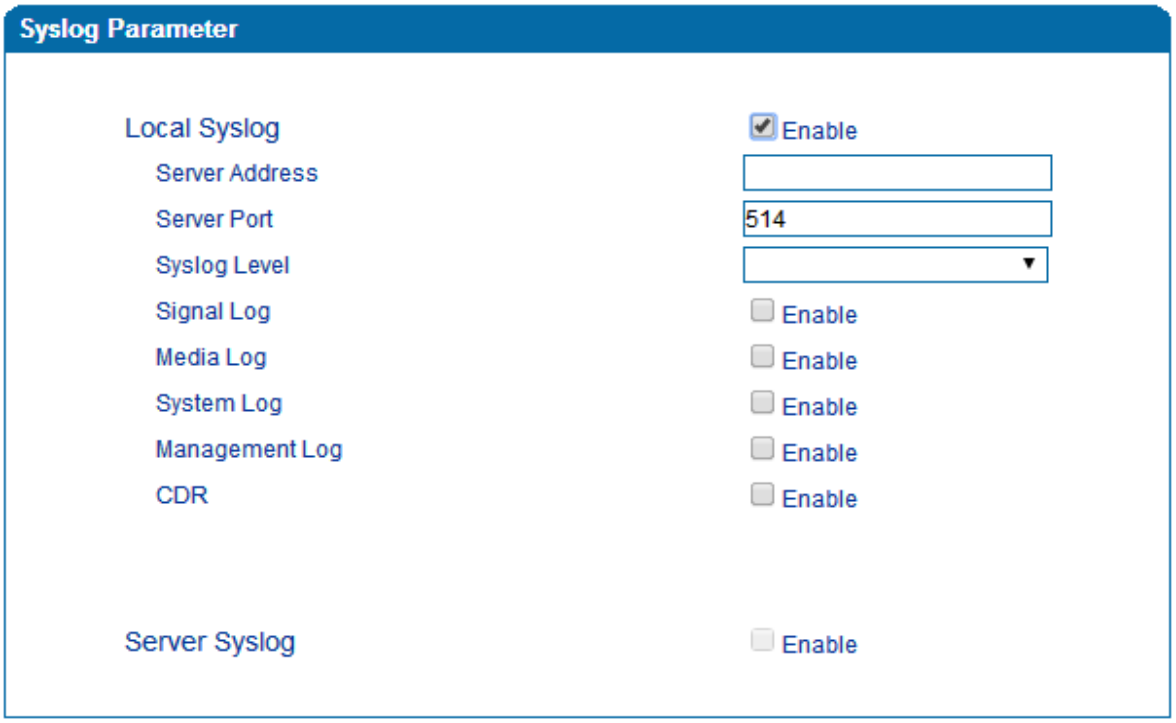


Figure 3.14-3 Syslog Parameter

Enable send CDR, and then send communication information to syslog server.

3.13.4 Provision

Provision is used to make the DAG FXO automatically upgrade with the latest firmware stored on an http server an ftp server or a tftp server.

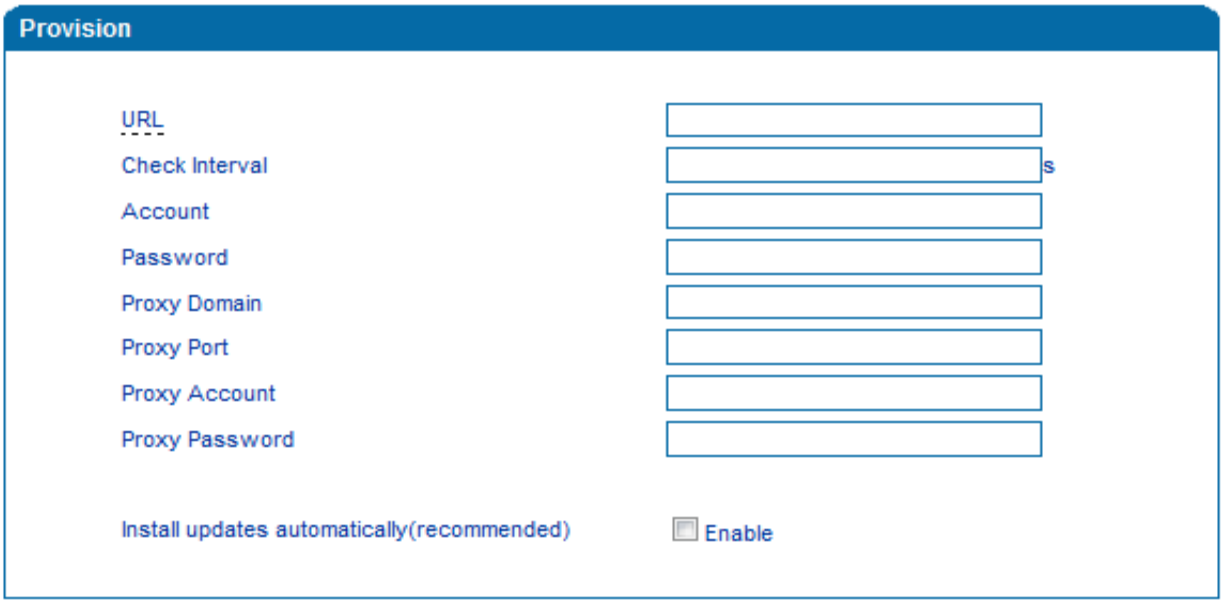


Figure 3.14-4 Provision

URL	Provisioning server URL, support HTTP, TFTP, FTP
Check Interval	The interval to check the changes on the provisioning server
Account	Account for login provisioning server
Password	Account for login provisioning server

3.13.5 Cloud server

Your can register the gateway to cloud server, and then the gateway will be managed by cloud server.

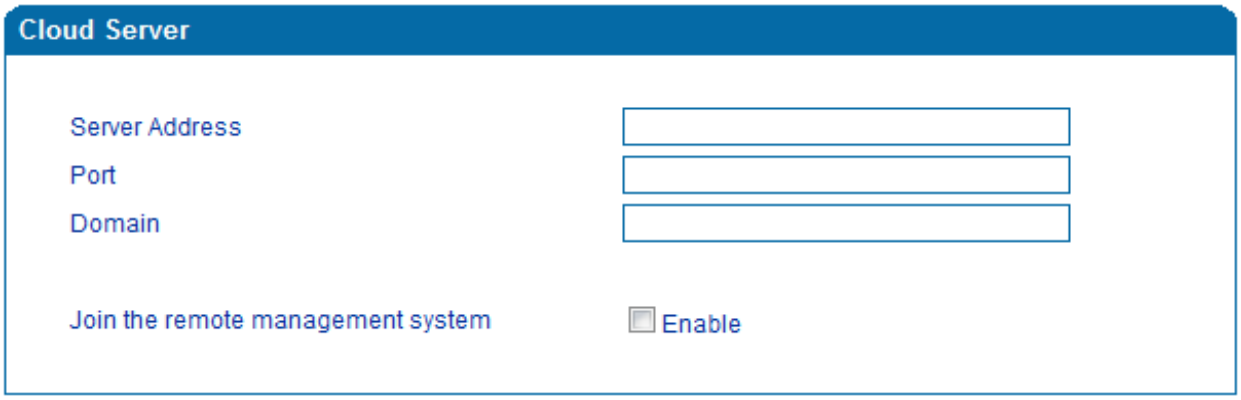


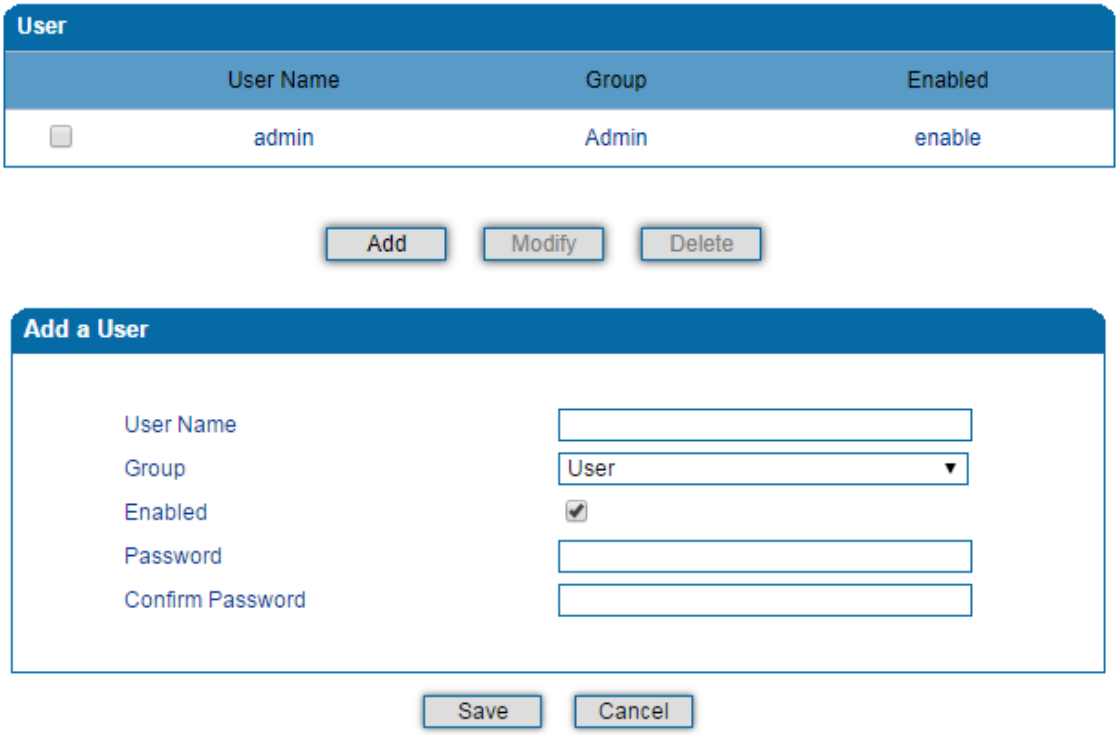
Figure 3.14-5 Cloud Server

Explanation of related parameters

Server Address	The IP address or domain of the cloud server
port	The listening port of the cloud server
Password	Password for register with cloud server

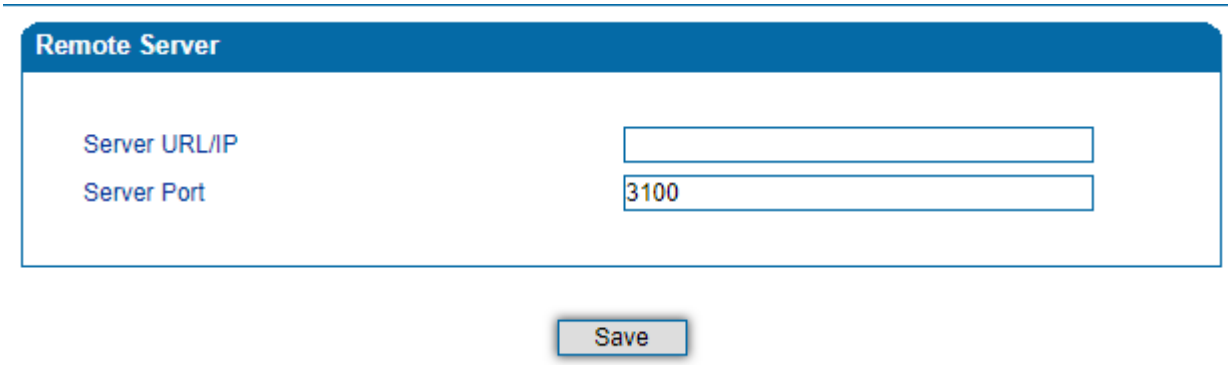
3.13.6 User manage

Click **Management → User Manager**, and you can modify the username name. Meanwhile, if you are the administrator of the device, you can assign a role (a user or a guest) for other users. That is to say, this function allows you to create multiple users with multiple levels of privileges. The device supports to create admin, user and guest privileges for different users.



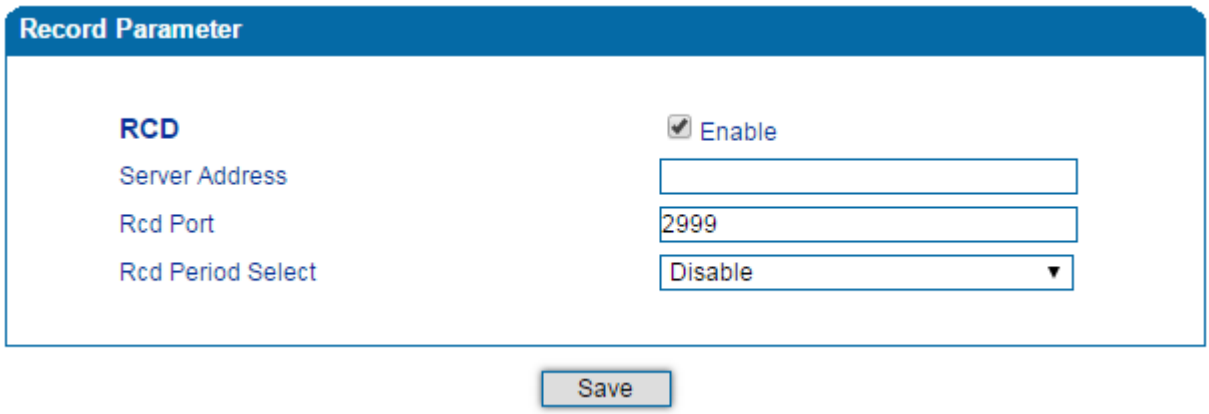
3.13.7 Remote server

Remote server is a type of cloud service which develop for Dinstar devices, aim to helps user manage web and telnet remotely. Please provide your device's SN and contact with support to get server link.

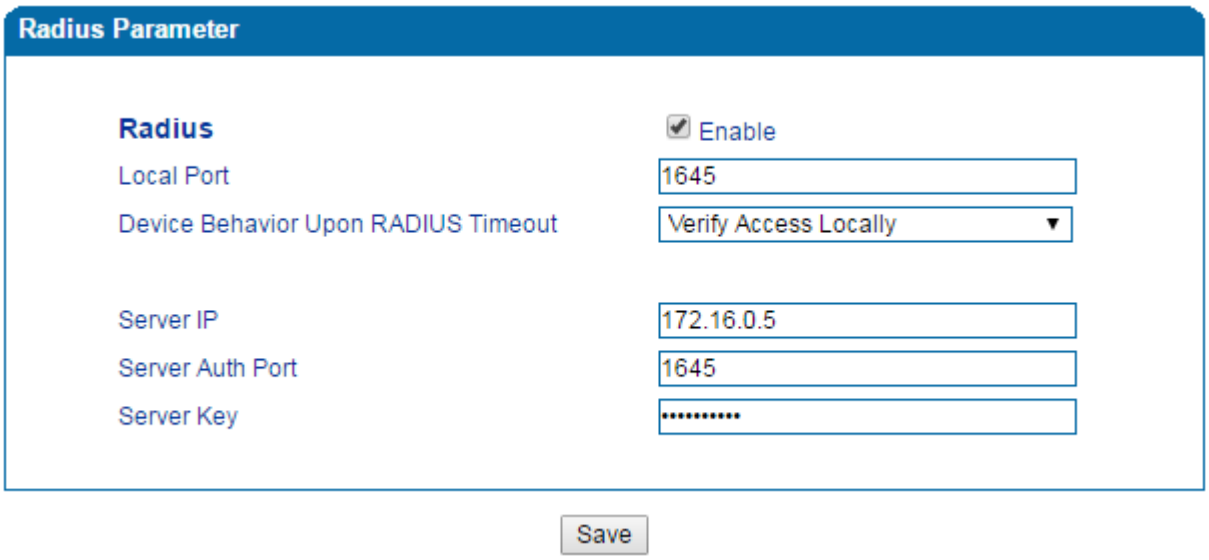


3.13.8 Record Parameter

On the following interface, you can fill in the IP address of the recording server and then you record the coversations of calls.



3.13.9 Radius Parameter



3.13.10 Action URL

Action URL allows VoIP platforms to obtain the status and other information of the DAG device and report the information of the DAG device to other devices.

Startup: To report the startup of the DAG device.

Offhook: To report the Offhook status of an FXO port.

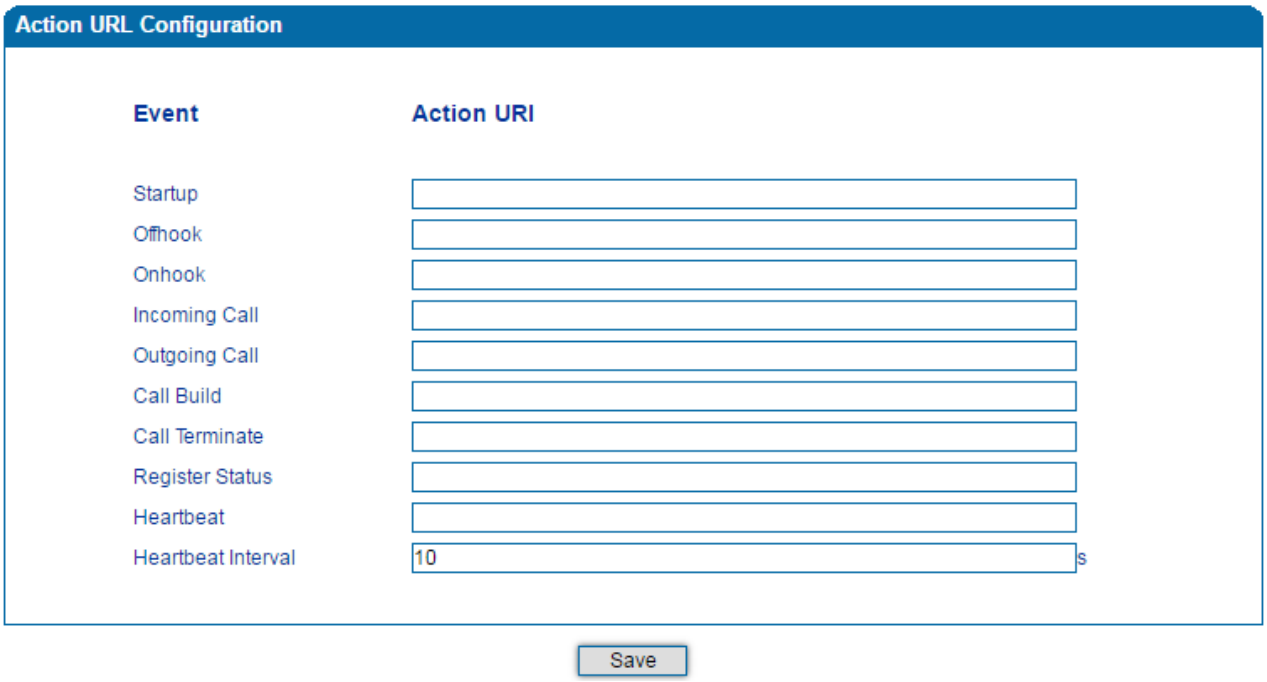
Onhook: To report the onhook status of an FXO port.

Incoming call: an incming call is received by an FXO port of the DAG device.

Outgoing call: a call is going out from an FXO port of the DAG device.

Call built: a call is being establishing in an FXO port of the DAG device.

Call Terminate: a call is terminated in an FXO port of the DAG device.



3.14 Security

3.14.1 WEB ACL

ACL (Access Control List) for WEB is used to configure IP addresses (users) that are allowed to access the WEB page of the gateway. The IP address list can't be null once ACL is enabled.

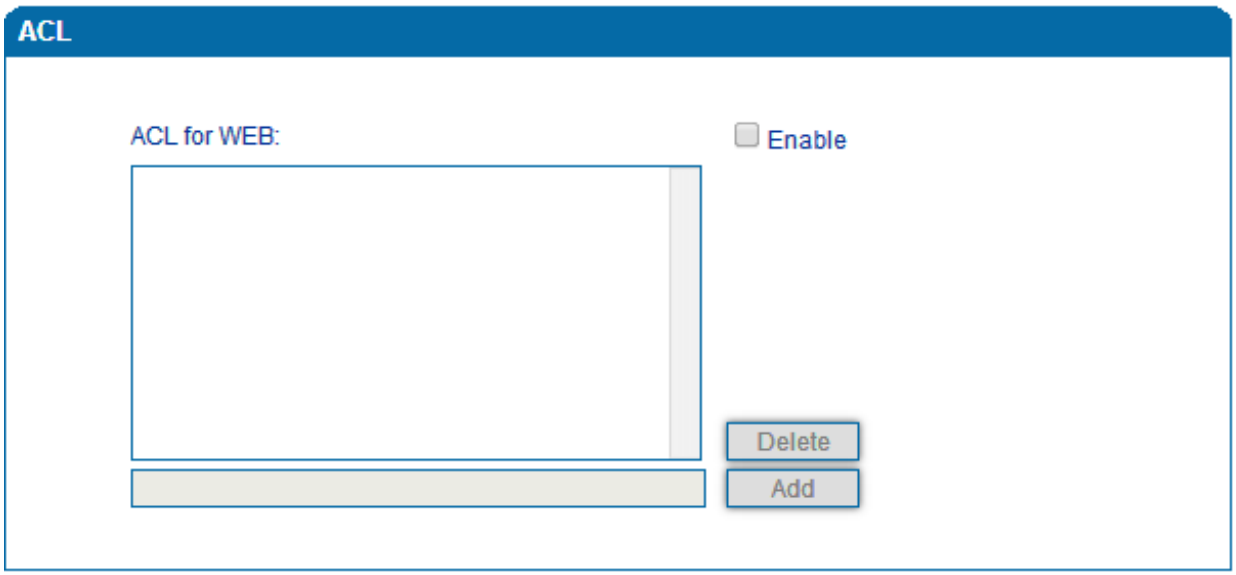


Figure 3.15-1 ACL for WEB

3.14.2 Telnet ACL

ACL (Access Control List) for WEB is used to configure IP addresses (users) that are allowed to access the Telnet page of the gateway. The IP address list can't be null once ACL is enabled.

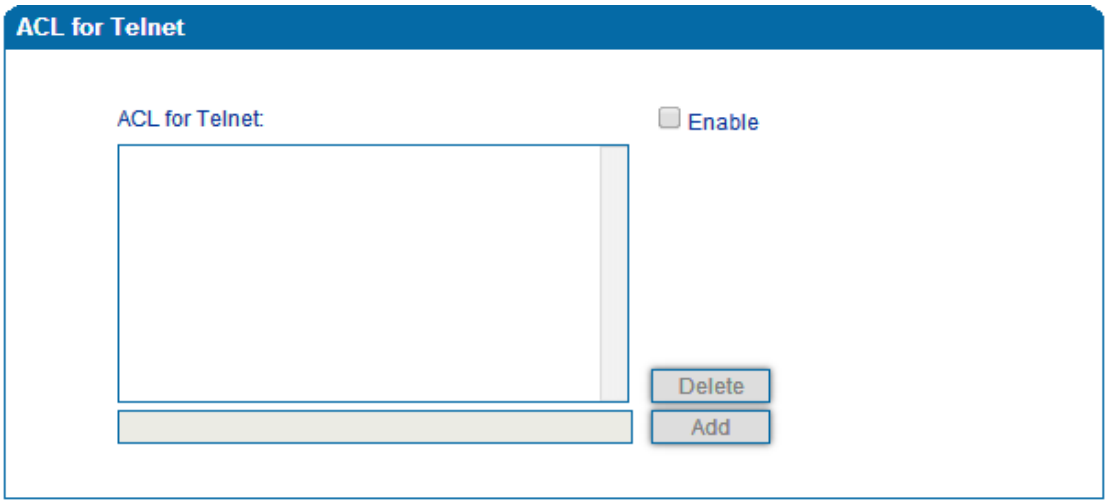


Figure 3.15-2 ACL for Telnet

3.14.3 Passwords

On the following interface, user can configure or modify the username and password for access the WEB interface and the Telnet interface.

Note：Both the username and password of Web and Telnet are 'admin' and 'admin'.

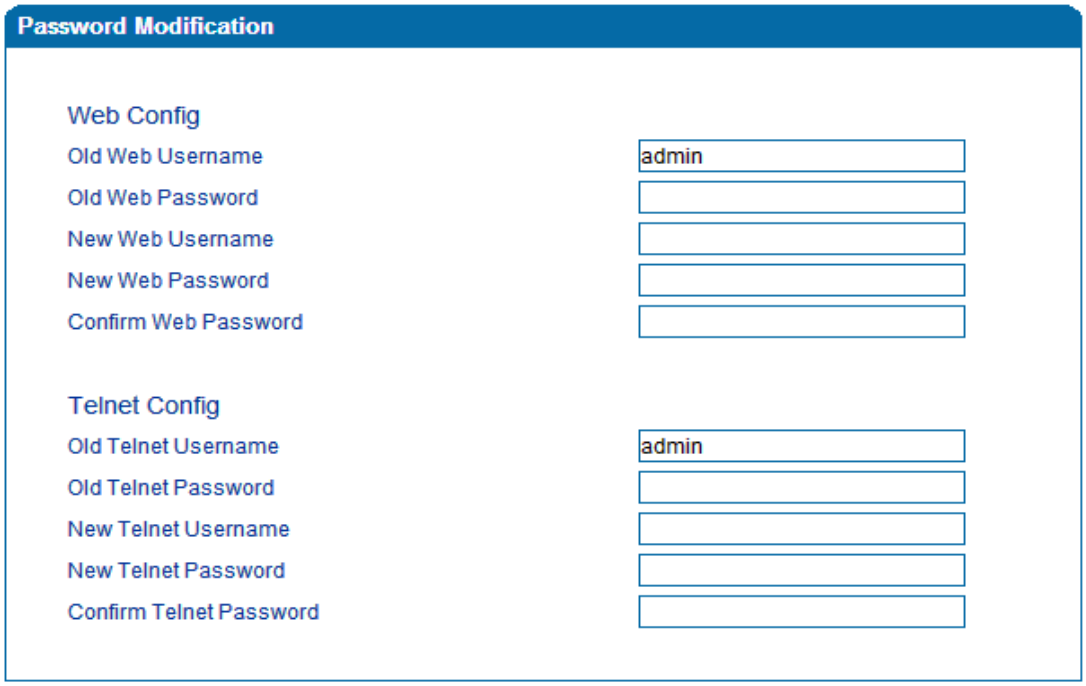
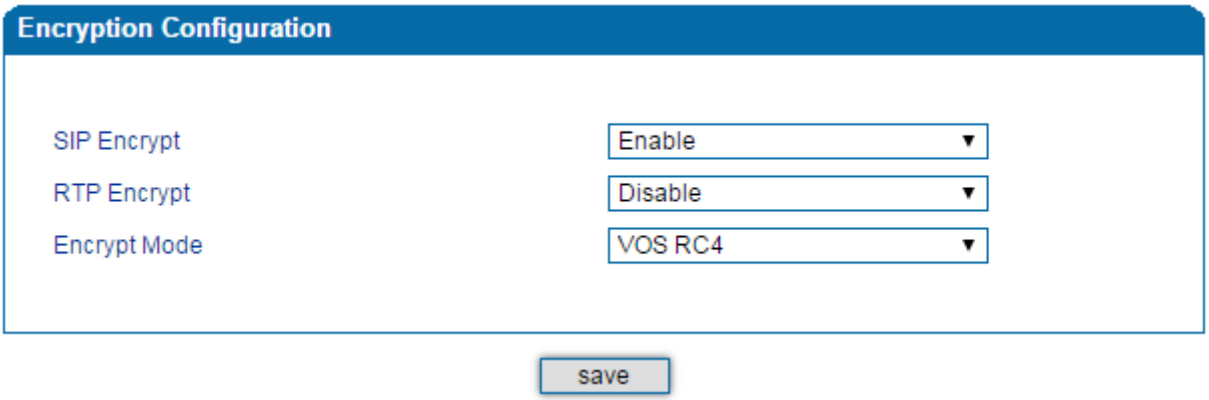


Figure 3.15-3 Password Modification

3.14.4 Encrypt

On the following interface, you can enable or disable SIP encryption and RTP encryption. The envryption mode is defaulted as VOS RC4.



3.15 Tools

3.15.1 Firmware upload

Firmware upload steps:

Step 1.

Check the current firmware version on the ***System Information page***

Figure 3.16-1 Firmware Version

Step 2.

Prepare firmware package. The most important is that the package must match with the existing version. Package version consists of the following parts:

2.81.xx.xx

01/02 is vendor name

81 is hardware version, xx.xx is version number

Step 3.

Upload firmware, select the package from specific folder on the computer and click ***Upload*** button.

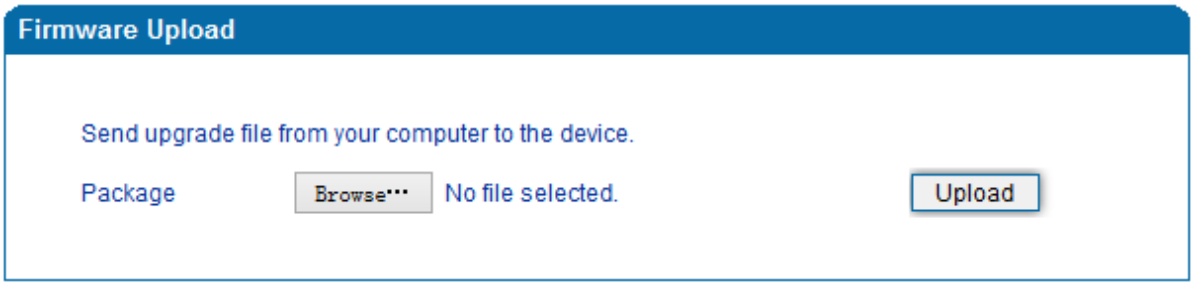


Figure 3.16-2 Firmware Upload

Step 4.

Keep waiting until it prompts 'Software loaded successfully!'

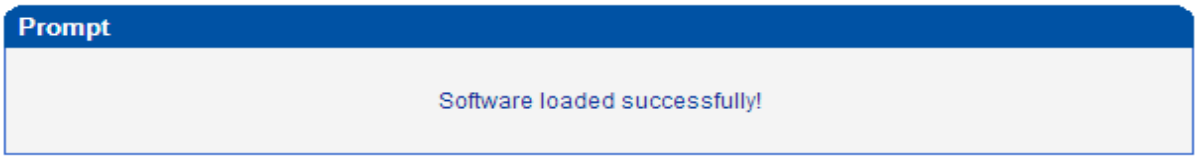


Figure 3.16-3 Successful Firmware Upload

Step 5.

Reboot gateway. Refer to web page ***Maintenance-> Device Restart***

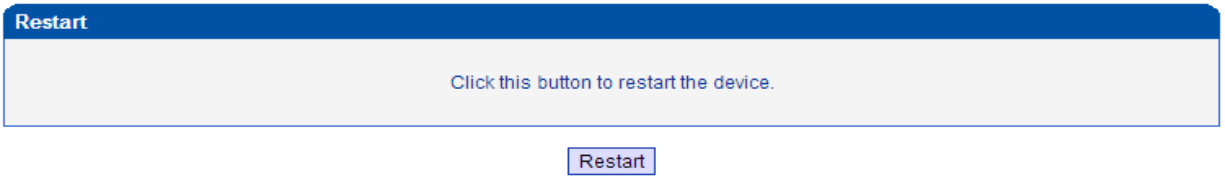


Figure 3.16-4 Restart Gateway

3.15.2 Data Backup

The process data backup:

1) Click "Data Backup"
2) Click "Backup" to backup data to PC.

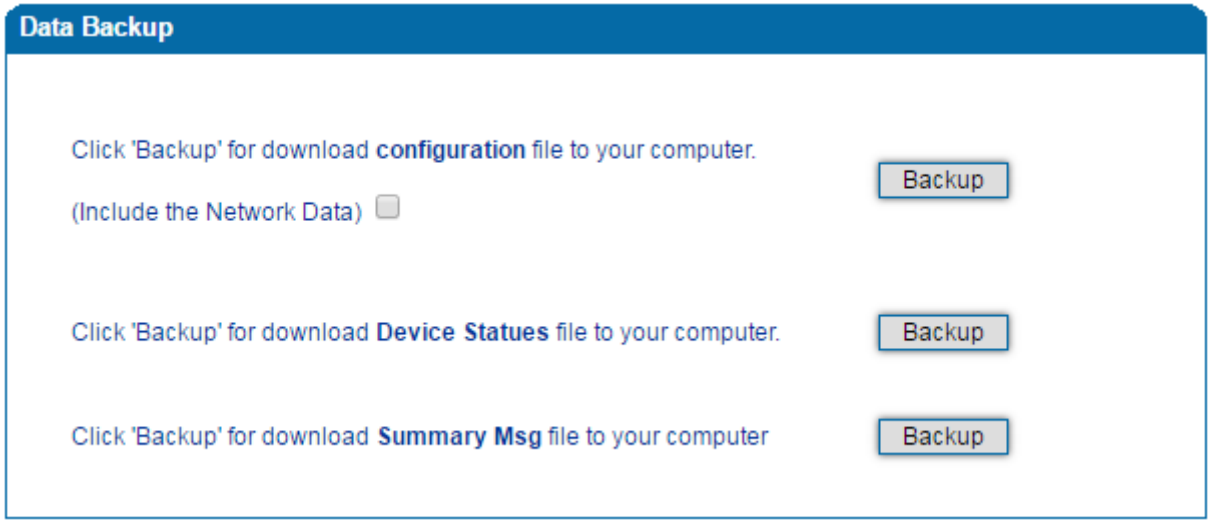


Figure 3.16-5 Data Backup

3.15.3 Data Restore

The processes of data restore:

- Click ‘Data Restore’;
- Browse file, select data file.
- Click ‘Restore”’and then import successfully, the device will restart automatically.

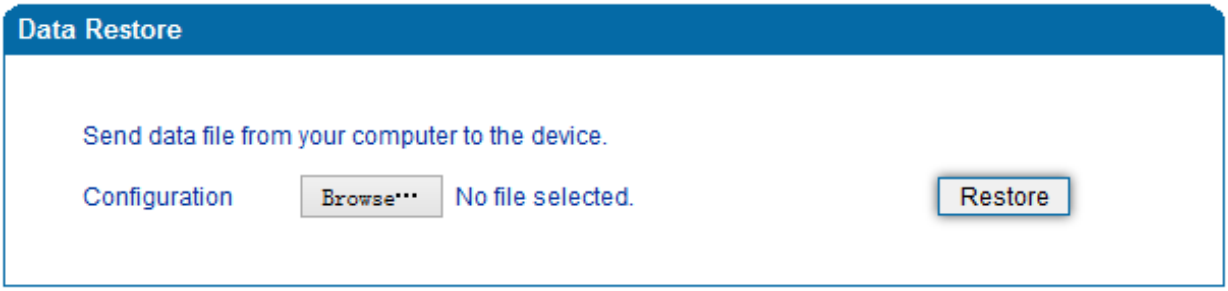
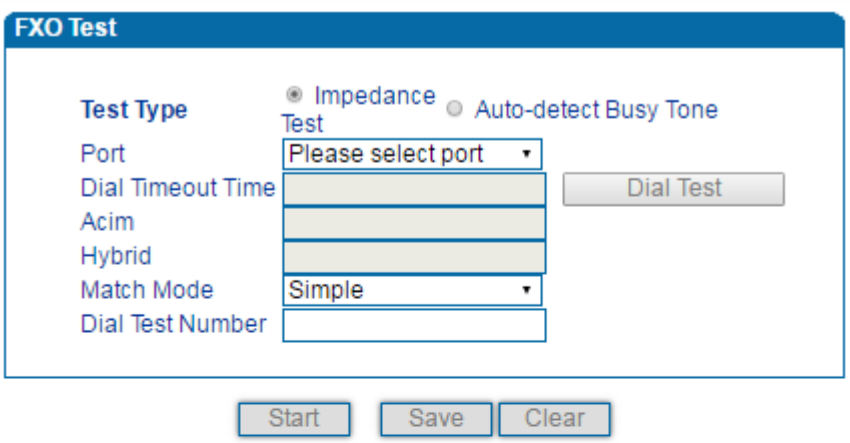


Figure 3.16-6 Data Restore

3.15.4 FXO Test



1. FXO ports impedance test is mainly to the technical personnel of O for impedance matching
2. Dial test number can configure by yourself,but avoid using * service the same number, and must all digit
3. You can just select online status port to test
4. If you don't know dial timeout time, can to test dial timeout time first, lasts about 10 seconds, finished the test will show timeout.Error is up to 30 seconds.
5. 'Match Mode': 'Simple' takes about 15 minutes, 'Standard' takes about 30 minutes, 'Exact' takes about 45 minutes.
6. If you want to give up the test result, click 'Clear'.
7. If you don't click 'Save',after the restart,test number,dial timeout and impedance will be invalid.
8. Before testing is completed, please don't leave the page, in order to avoid mistakes.

This function is used to automatically check the impedance of the PSTN line. The steps of how to use this function is as follows

1. Connect the PSTN line with the FXO port.

2. Select the the FXO port that is to be checked.

3. Select a mode. There are three modes: Simple, Standard and Exact. It is advised to use to the simple mode so as to save time.

4. Fill in the test digits instead of filling in a real number.

5. Click Start to test the port and wait for more than 15 minutes. During the time, please do not refresh the web.

6. After the test is completed, confirm the value in the “Acim” field.

7. Configure the value of “Acim” on the Advance ->FXO Parameter interface.

8. Restart the device to make the configurations to take effect.

3.15.5 Ping Test

On the **Tools → Ping Test** interface, user can use Ping to check whether the network is working or not.

Ping instructions:

1) Click 'Tools → Ping Test' on the navigation tree on the left;

2) Fill in IP address or domain whose connection needs to be checked, click **start**.

If a message is received, it indicates that network connection is normal. Otherwise the network connection is faulty.

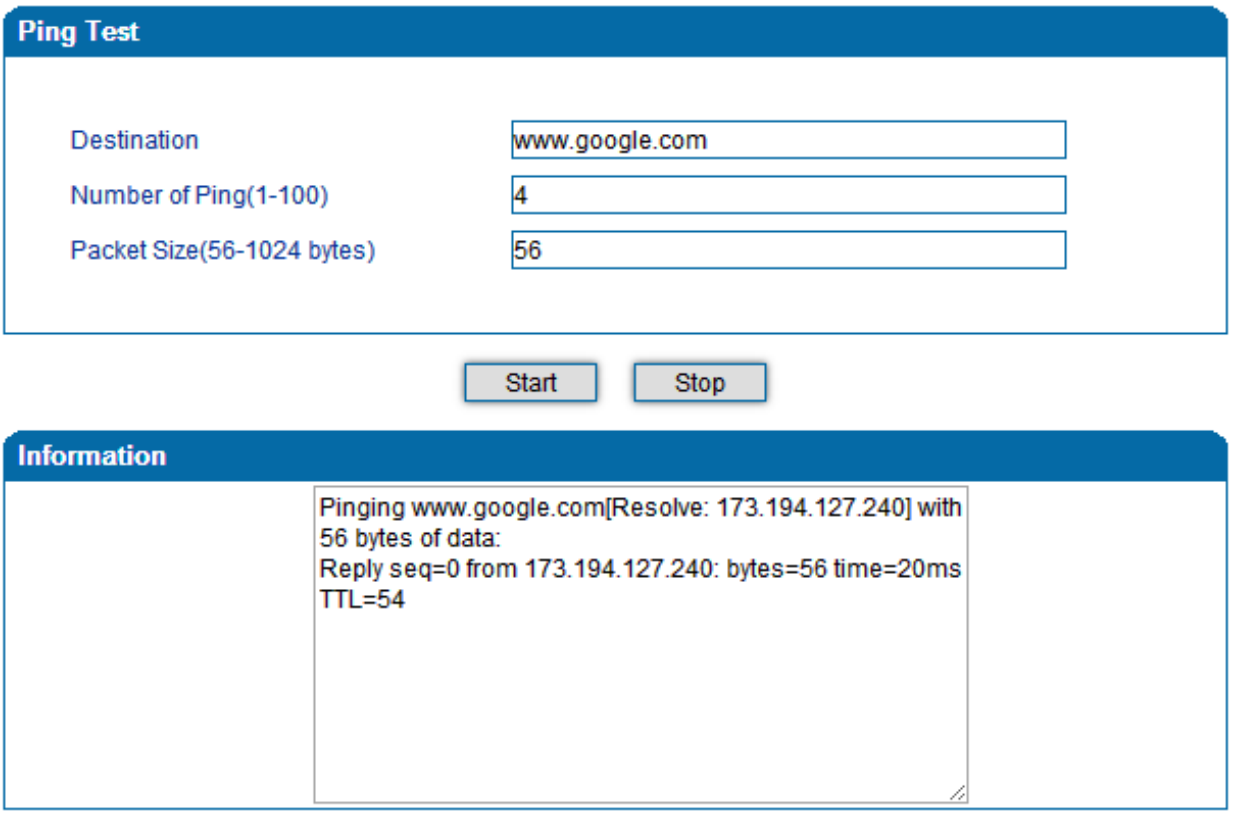


Figure 3.16-7 Ping Test

3.15.6 Tracert Test

Tracert is a trace router used to track routing.

Tracert sends a sequence of Internet Control Message Protocol (ICMP) echo request packets addressed to a destination host. Determining the intermediate routers traversed involves adjusting the time-to-live (TTL), aka hop limit, Internet Protocol parameter. Frequently starting with a value like 128 (Windows) or 64 (Linux), routers decrement this and discard a packet when the TTL value has reached zero, returning the ICMP error message ICMP Time Exceeded.

Tracert works by increasing the TTL value of each successive set of packets sent. The first set of packets sent have a hop limit value of 1, expecting that they are not forwarded by the first router. The next set have a hop limit value of 2, so that the second router will send the error reply. This continues until the destination host receives the packets and returns an ICMP Echo Reply message.

Trace route uses the returned ICMP messages to produce a list of hops (which usually consists of routers and layer 3 switches) that the packets have traversed. The timestamp values returned for each router along the path are the delay (aka latency) values, typically measured in milliseconds for each packet.

Tracert introduce：

- Click ‘ Tracert Test’ in the navigation tree;
- Fill in IP address or domain whose route needs to be tracked, and then click **start**.

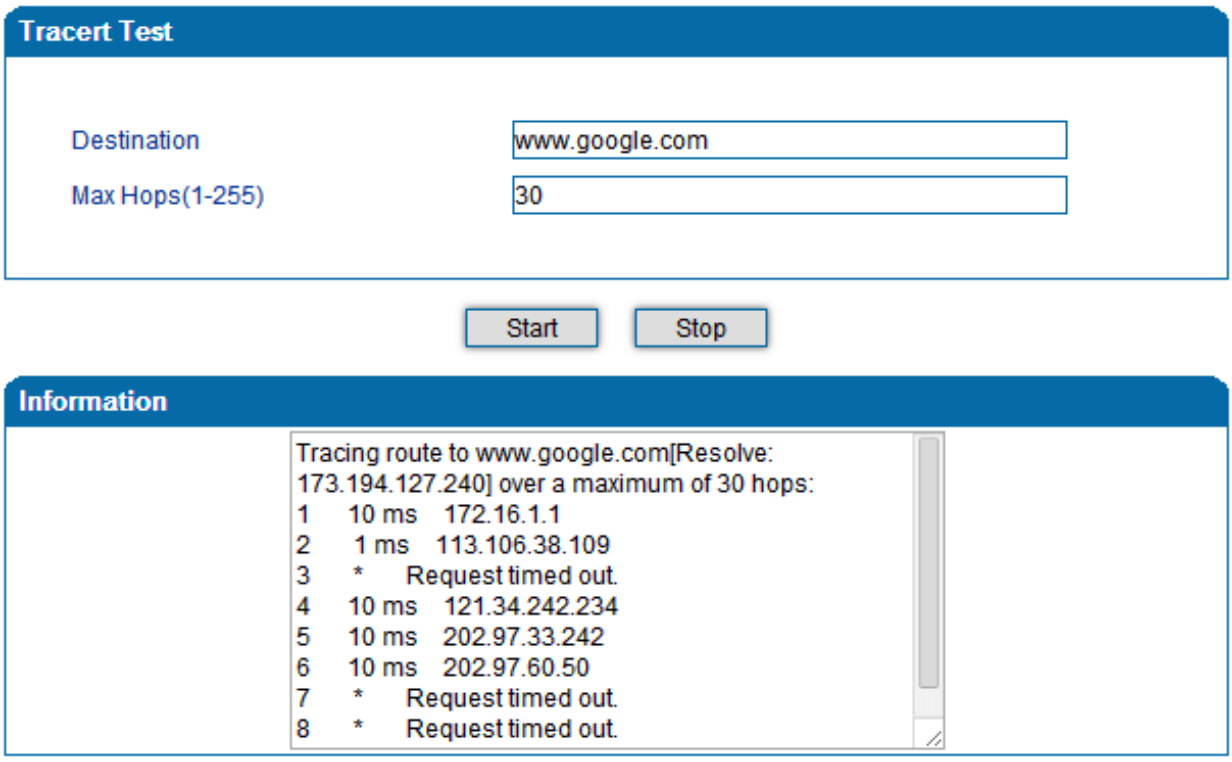


Figure 3.16-8 Tracert Test

3.15.7 Network Capture

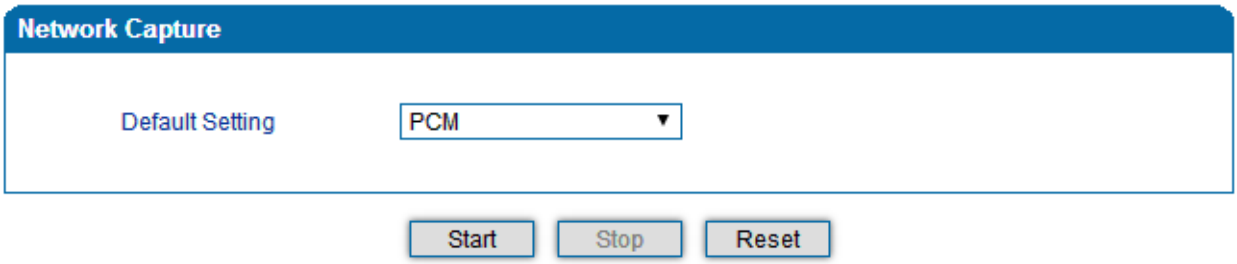
Network capture is a very important diagnostic tool for maintenance. It can be used to capture data packages of the available network ports.

Default Setting is PCM capture

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

- **To enable PCM capture**

- Select ‘PCM’ on Network Capture page



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

The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of PCM capture as below:

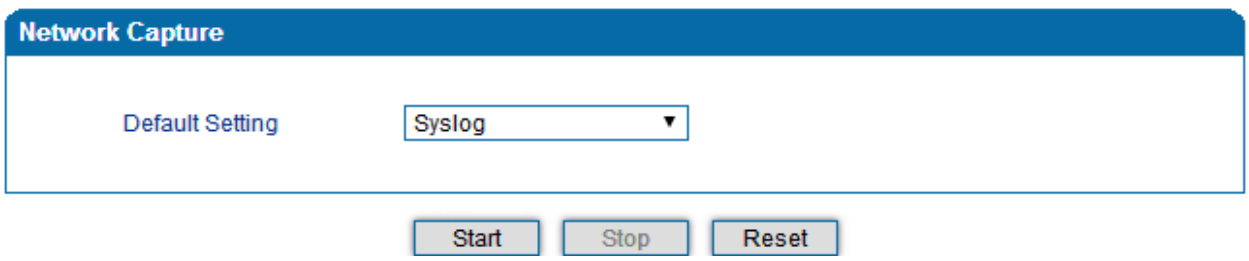
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 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)

Getting start to Syslog capture

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

To enable syslog capture

- Select Syslog special only on Network Capture page



- Click “Start’ to enable syslog capture
- Dialing out through gateway, start talking a short while then hangup the call.
- Click ‘Stop’ to disable syslog capture
- Save the capture to local computer

The capture is named to ‘capture(x).pcap’, x is serial number of capture and will be added 1 in next time. The sample of syslog capture as below:

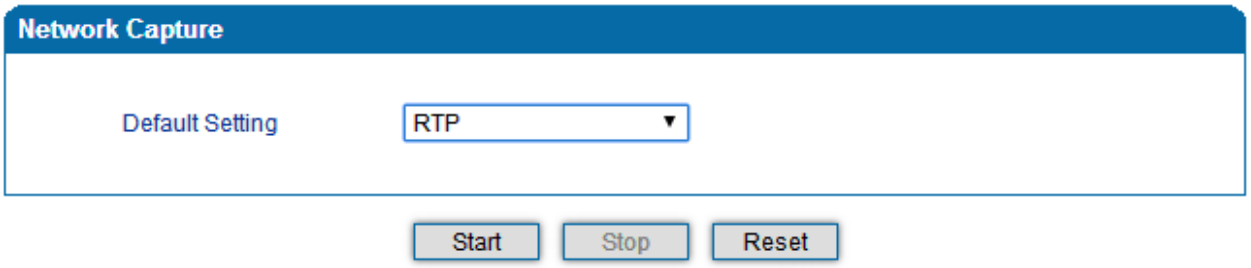
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1> [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,crypt
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, Phc
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,crypt
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, Phc
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 cal
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,cryp
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, Ph
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_INV
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 Phone
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,cryp
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, Ph
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_BYE

Getting start to RTP capture

PCM capture is help to analysis voice stream between gateway and remote IPPBX/SIP Server.

To enable RTP capture:

- Select RTP special on Network Capture page



- Click Start to enable RTP capture
- Dialing out through gateway, start talking a short while then hangup the call.
- Click Stop to disable RTP capture
- Save the capture to local computer

The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of RTP capture as below:

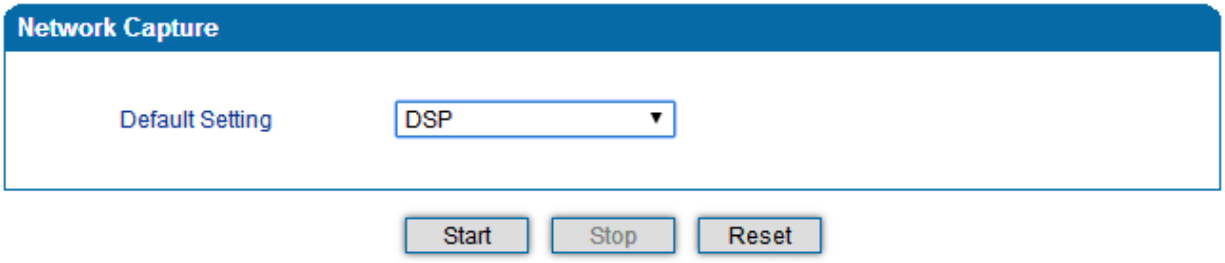
No.	Time	Source	Destination	Protocol	Length	Info
176	7.020000	172.16.221.228	116.204.105.50	SIP	565	Request: REGISTER sip:116.204.105.50 |
178	7.030000	116.204.105.50	172.16.221.228	SIP	411	Status: 200 OK (1 bindings) |
244	11.610000	172.16.221.228	58.56.64.101	SIP/SDP	814	Request: INVITE sip:201@58.56.64.101 |
248	11.710000	58.56.64.101	172.16.221.228	SIP	480	Status: 100 Trying |
249	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	733	Status: 183 Session Progress |
250	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	719	Status: 200 OK |
252	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
253	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
254	11.720000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1000, Time=160, Mark
255	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
256	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
257	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
258	11.740000	172.16.221.228	58.56.64.101	SIP	434	Request: ACK sip:201@58.56.64.101:5060 |
259	11.740000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1001, Time=320
261	11.770000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1002, Time=480
263	11.780000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1003, Time=640
264	11.810000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1004, Time=800
265	11.830000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1005, Time=960
266	11.840000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1006, Time=1120
267	11.870000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1007, Time=1280
268	11.890000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1008, Time=1440
270	11.900000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1009, Time=1600
271	11.930000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31521, Time=1806312883
273	11.930000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1010, Time=1760
274	11.940000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1011, Time=1920
275	11.950000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31522, Time=1806313043
277	11.970000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1012, Time=2080
278	11.970000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31523, Time=1806313203

Getting start to DSP capture

DSP capture is help to analysis voice stream inside DSP chipset. The DSP chipset will handle RTP from IP network as well as voice stream from analog phone.

To enable DSP capture:

- Select DSP only on Network Capture page



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

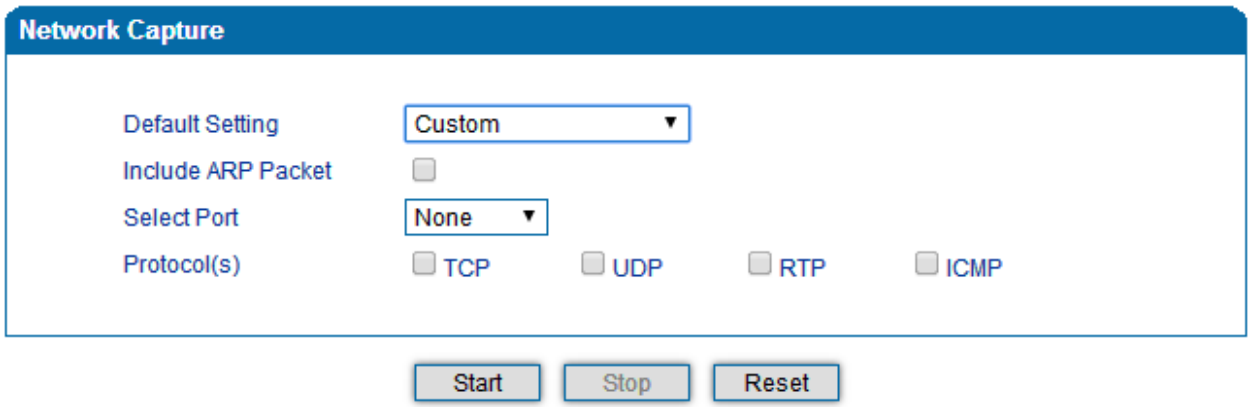
The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of RTP capture as below:

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)

- **Configurable capture options**

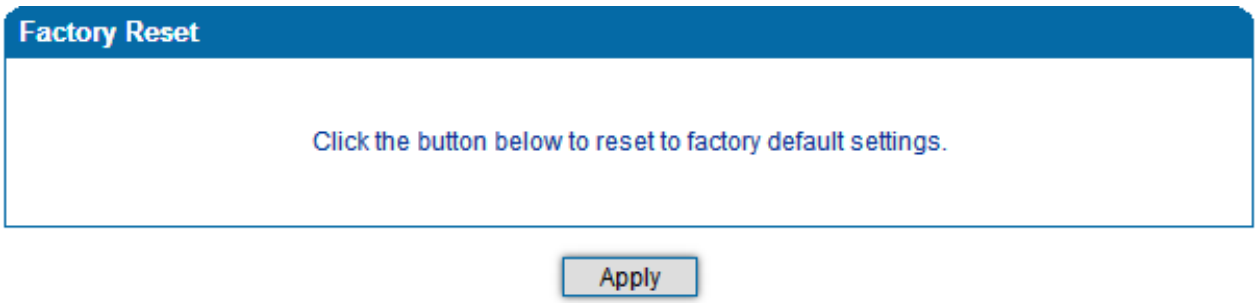
- **Getting start to custom capture**

This menu provides more options to capture specific packets according to actually needs.



3.15.8 Factory Reset

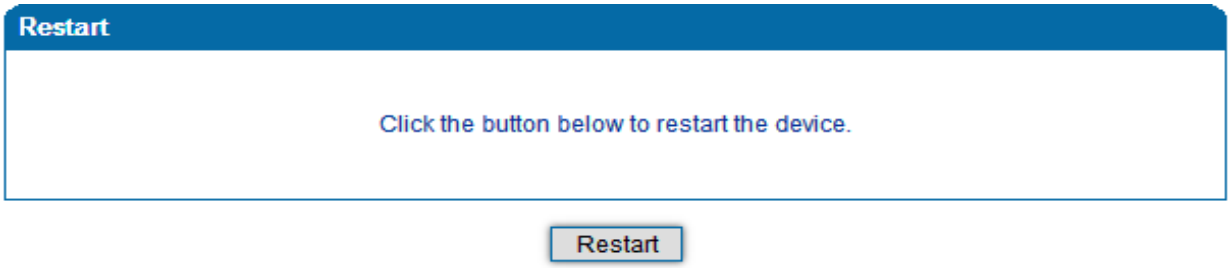
Click ‘Apply’ to restore the factory settings.



Factory Reset

3.15.9 Device Restart

After saving all the configurations or changes to the equipment, user can restart the DAG FXO gateway for the changes to take effect.



4 Glossary

- DNS：Domain Name System
- SIP：Session Initiation Protocol
- TCP： Transmission Control Protocol
- UDP： User Datagram Protocol
- RTP： Real Time Protocol
- PPPOE： 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
- IMS: IP Multimedia Subsystem
- ACL: access rule list
- SNMP: Simple Network Management Protocol
- FXO: Foreign Exchange Station
- FXO: Foreign Exchange Office