DINSTAR

DAG2000-16S VoIP Gateway

User Manual V3.0

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Preface

Welcome

Thanks for choosing **DAG2000-16S VoIP Gateway**! We hope you will make optimum use of this flexible, rich-feature VoIP-to-FXS 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.

For interoperability with different IPPBX/Softswitch platform, you can refer to relevant configuration guide of different systems.

This manual is written with reference to the default configurations of the **DAG2000-16S** VoIP Gateway.

Intended audience

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

System engineers are persons who customize the configurations to meet the requirements of users.

Parts of the document containing description of telephony features are aimed at users who are the persons who will actually use the gateway.

Contents

1 Introduction of DAG2000-16S

1.1 Overview

DAG2000-16S VoIP gateway provides voice services based on IP network. It's a cost-effective and flexible solution for SOHO (Small Office-Home office), remote office, medium-sized enterprise and enterprise with multiple branches.

The gateway connects to analog telephone, fax and traditional analog PBX with standard voice interfaces and provides high quality voice service.

The gateway, based on standard SIP protocol is compatible with leading IP PBX, soft-switch and SIP-based platform.

The FXS analog gateway available in the following configurations:

Model	Voice Channels	FXS Ports	Physical Port Labels
DAG2000-16S	16	16	0-15

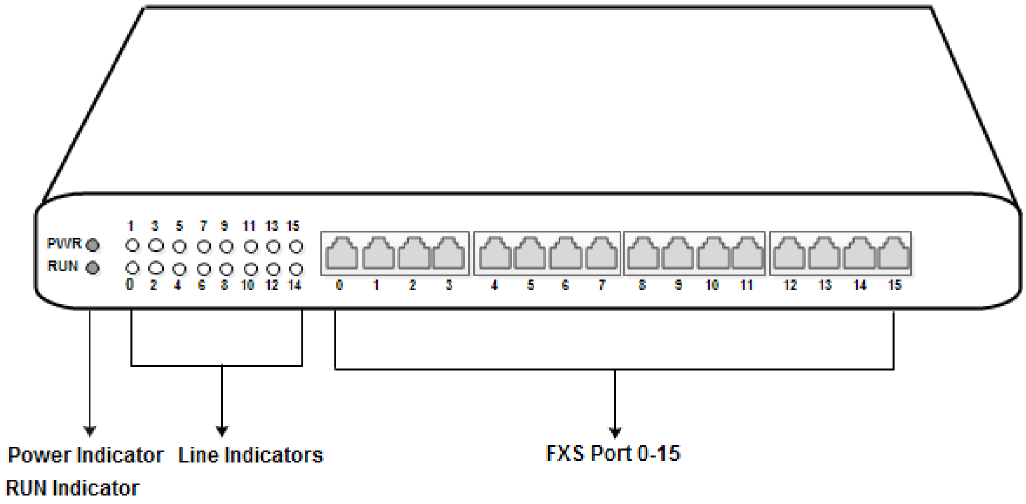
For detailed hardware and software features, please refer to "product specifications".

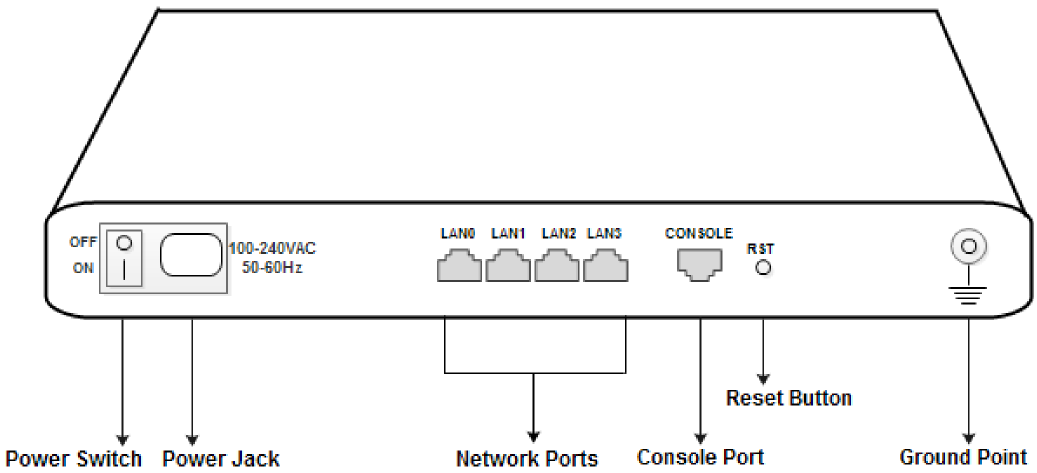
1.2 Equipment Appearance

Front View

Back View

1.3 Ports and Connector





Port Name	Connector	Description
Power Jack	Power Jack	To connect 100-240V AC 50-60HZ power supply
LAN Port	RJ45	to connect to the IP network over a DSL modem or Router or a LAN switch
FXS Ports 0-15	RJ11	FXS ports to connect standard analog phone or FAX machine or a PBX
Console Port	RJ45	Console port is used to carry out maintenance-related configurations

1.4 Functions and Features

1.4.1 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)
- TFTP/HTTP/HTTPS
- DNS/DNS SRV (RFC 1706/RFC 2782)
- VLAN 802.1P/802.1Q

1.4.2 Voice and Fax parameters

- G.711A/U law, G.723.1, G.729AB,iLBC,AMR
- 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

1.4.3 Supplementary service

- Call waiting

- Call transfer（Blind transfer, Attend transfer,）
- Quick pickup
- Call Forwarding Unconditional
- Call Forwarding on No Reply
- Hotline
- Call hold
- DND
- Three-way calling
- Voice mail
- Direct IP Call

2 Basic Operations

2.1 Methods to Number Dialing

Dial mobile phone or extension number

- Dial the number directly and wait for 3 seconds (Default "*No dial timeout*");
- Dial the number directly and press #.

2.2 Direct IP Calls

The DAG2000-16S gateway allows users to directly call through IP address. Under this circumstance, the user only needs an analog phone which is connected to a FXS port of the gateway, and calls can be established without register.

Calls can be established through IP address as long as one of the following conditions is met.

- Both the DAG2000-16S and other VoIP device have public IP addresses;
- The DAG2000-16S and other VoIP device use private IP addresses of a same LAN;
- The DAG2000-16S and other VoIP device can be connected through a router and use public or private IP addresses (with necessary port forwarding or DMZ).

Operation Process：

Step1: Pick up the analog phone and then dial "*47";
Step2: Enter the target IP address.

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

Example:

Assume that the target IP address is 192.168.0.160, user need to dial ***47** and then **192*168*0*160**. After that**,** press the "#" key or wait 3 seconds. Then signaling interaction is completed and ringing can be heard.

【Note】:You cannot make direct IP calls between two FXS ports of a same DAG2000-16S since they are using same IP addresses. Call through IP address is only routed to the default destination port 5060.

2.3 Call Holding

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has the button).Press the "flash" button again to release the previously held caller and resume conversation. If no "flash" button is available, use "hook flash" instead.

2.4 Call Waiting

If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear a IVR voice 'Please hold on, the subscriber you dialed is busy' and the called party will hear three beeps.

By pressing the flash button or the flash hook, the called party is able to switch between the new incoming call and the current call.

2.5 Call Transfer

2.5.1 Blind Transfer

Blind transfer is used to transfer call to a third party without informing the caller. Assume that A and B are in a conversation. A wants to blind Transfer B to C:

- A presses **FLASH** on the analog phone to hear the dial tone;
- Then A dials ***87** and C's number and # (or wait for 4 seconds);
- A will hear the confirm tone. Then, A hangs up, and B and C enter into a conversation.

Note:

"*Call features enable*" must be set to "Yes" on WEB configuration page. Caller A can place a call on hold and wait for one of the three situations:

- A quick confirmation tone (similar to call waiting tone) which follows the dial tone. This indicates the transfer is successful. At this point, Caller A can either hand up or make another call.
- A quick busy tone which follows a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone indicates the transfer has failed.
- Continuous busy tone. This means the call has timed out.

2.5.2 Attended Transfer

Attended transfer allows the transferring party either connects the call to a ringing phone (ringback heard) or speaks with the third party before transferring the call to the third party.

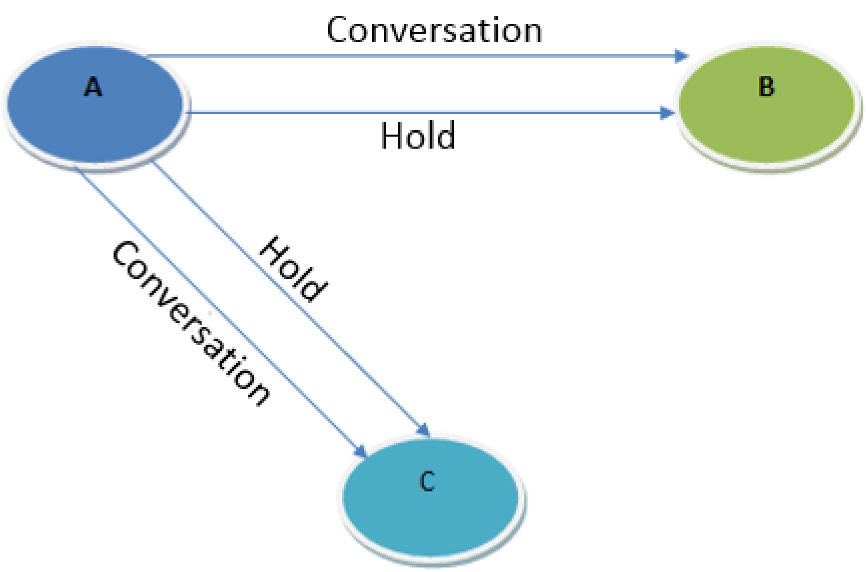
Assume that A and B are in conversation. Caller A wants to *attended transfer* B to C:

- A presses **FLASH** on the analog phone and wait for dial tone;
- Then dial C's number followed by # (or wait for 3 seconds);
- If C answers the call, A and C are in conversation. Then A can hang up to complete the transfer;
- If C does not answer the call, A can press "flash" to resume call with B.

2.6 Three-way Calling

Three-way calling：

- A calls B,B picks up the phone, then A and B enters into conversation;
- A presses the hook flash, and the call between A and B is placed on hold. Then C calls A and A answers the call.
- A presses hook flash again, then the calls between A and B and between A and C are placed on hold. At this time, if A presses 1 , conversation between A and B is resumed; if A presses 2, conversation between A and C is resumed; if A presses 3, A,B and C enter into conversation.



2.7 Description of Feature Codes

The DAG2000-16S gateway supports all traditional and senior phone function. It provides feature codes for easy maintenance and easy entry to phone functions.

Feature Codes	Corresponding Function
***158#**	Dial *158# to inquiry the IP address of LAN port
***114#**	Dial *114# to inquire port account
150	Dial *150* to set the way of obtaining IP address
157	Dial *157*0 to set route mode; dial *157*1 to set bride mode
152	Dial *152* to set IPv4 address
153	Dial *153* to set subnet mask

156	Dial *156* to set default gateway's IP address
***193#**	Dial *193# to renew the IP address
***166*000000#**	Dial *166*000000# to reset to factory defaults
***111#**	Dial *111# to restart the gateway
***#**	Dial *# to place a call on hold
47	Dial *47* to establish a call through IP address
***51#**	Dial *51# to enable 'call waiting' feature
***50#**	Dial *50# to disable 'call waiting' feature
87	Dial *87* to blind transfer a call
72	Dial *72* to enable 'unconditional call forwarding' feature
***73#**	Dial *73# to disable 'unconditional call forward' feature
90	Dial *90* to enable 'busy call forwarding' feature
***91#**	Dial *91# to disable 'busy call forwarding' feature
92	Dial *92* to enable 'no answer call forwarding' feature
***93#**	Dial *93# to disable 'no answer call forwarding' feature
***78#**	Dial *78# to enable DND
***79#**	Dial *79# to disable DND
***200#**	Dial *200# to access voice mail
Flash/Hook	Used to switch between incoming calls. If the phone is not in session, flash/hook will switch a new channel for a new call.

2.8 Sending and Receiving Fax

The DAG2000-16S gateway supports four fax modes:

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

2.8.1 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. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

2.9 Local IVR Operation

2.9.1 Inquire IP address

Connect analog phone to FXS ports of the DAG2000-16S gateway, then pick up the phone. After dialing tone, dial *158# to inquire the IP address of LAN port.

2.9.2 Factory Reset

Pick up the phone, and then dial *166*000000#. After hearing a voice prompt of 'setting successfully', hang up the phone and the gateway is reset to factory defaults.

2.9.3 Configure LAN Port's IP Address

Before configuration, please ensure:

- The gateway is power on;
- Device has been connected to network;
- Telephone is connected to FXS port of the DAG2000-16S 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 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 DAG2000-16S gateway:
  Pick up the phone, dial *158#.

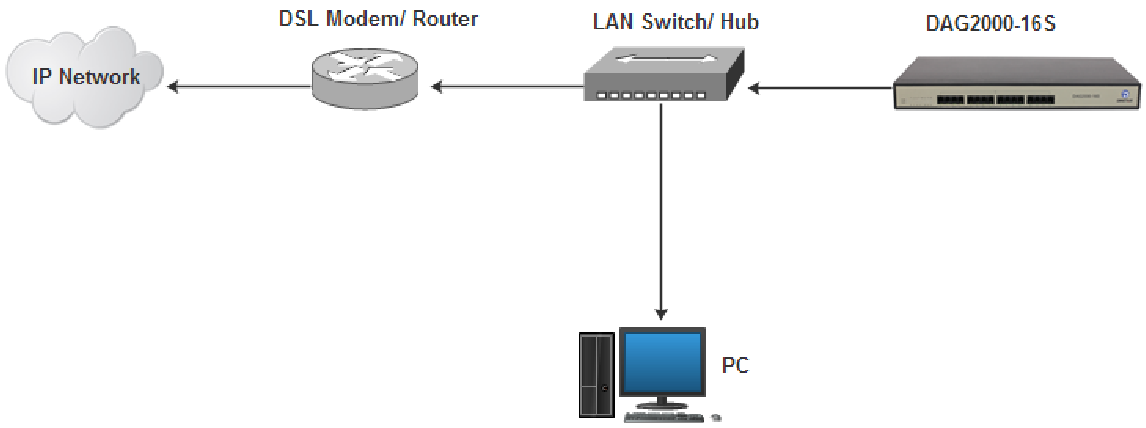
If the gateway uses PPPoE method to get IP address， the IP address needs to be configures through web browser.

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

3 Configurations on Web Interface

3.1 Network Connection

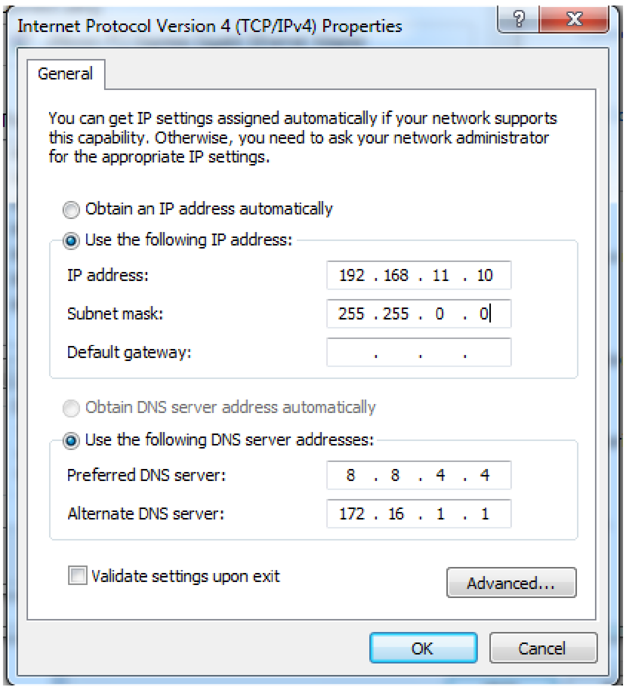
Connect the DAG2000-16S gateway to the network according to the following network topology, and dial *158 to query the IP address of the gateway.



3.2 Preparations for Login

Modify the IP address of the PC to make it at the same network segment with the DAG2000-16S 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：

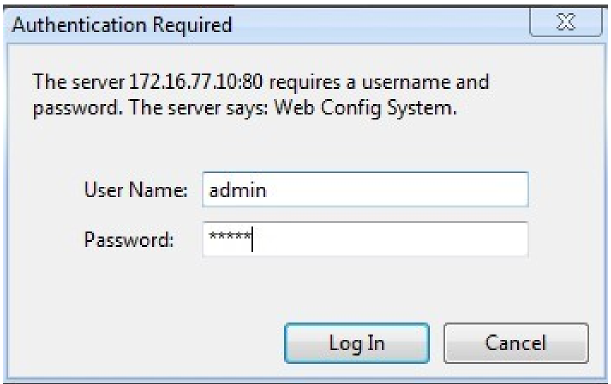


Check the connectivity between the PC and the gateway. Click **Start → Run** of PC and enter cmd to execute 'ping 192.168.11.1' to check whether the IP address of the DAG2000-16S gateway runs normally.

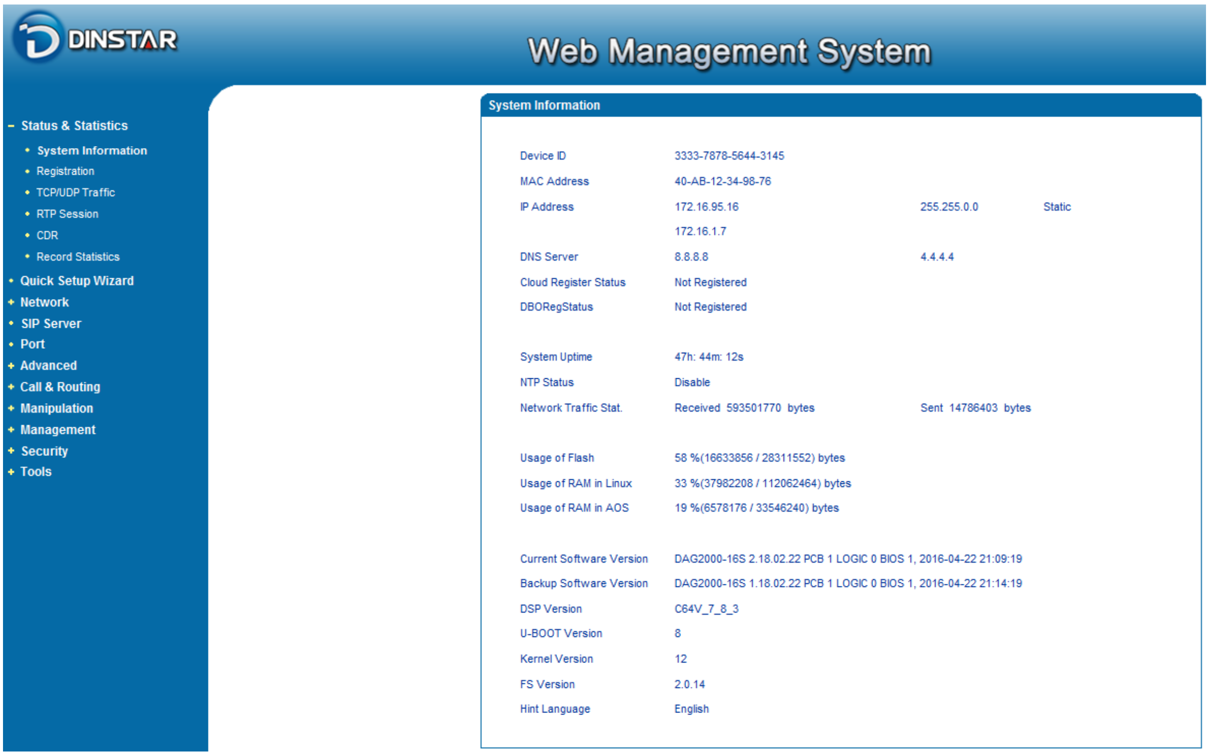
3.3 Log in Web Interface

Open a web browser and enter the IP address of the LAN port of the DAG2000-16S (the default IP of LAN port is 192.168.11.1). 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.



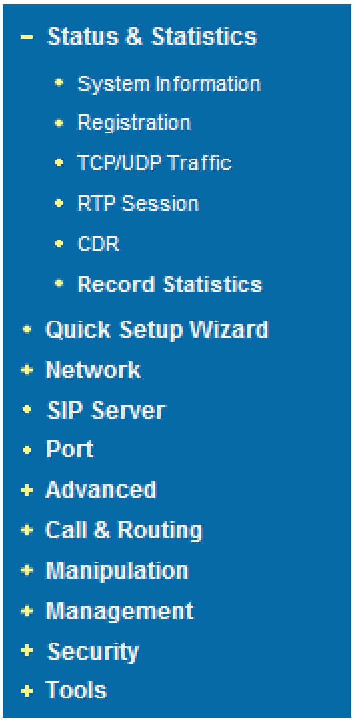
Enter default username and password: admin/admin, then click "Log in" to enter into the Web interface. And then you can see the following web interface.



3.4 Navigation Tree

The web management system of the DAG2000-16S VoIP gateway consists of the navigation tree and detailed configuration interfaces.

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



3.5 State and Statistics

3.5.1 System Information

On the System Information interface, you can view the information of device ID, MAC address, network mode, IP addresses, and version information, sever register status and so on.

Figure 3.5-1 System Information

Explanation of items on System Information interface

Device ID	A unique ID of each device. This ID is used for warranty and cloud server authentication.
MAC address	Hardware address of the LAN port
IP Address	The IP address of the gateway is shown. **DHCP: Obtain IP address automatically**. DAG2000-16S is regarded as a DHCP client, which sends a broadcast request and looks for a DHCP server from the LAN to answer. Then the first discovered DHCP server automatically assigns an IP address to the DAG2000-16S from a defined range of numbers. **Static IP Address:** Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a dynamic IP address,

	which is assigned *ad hoc* at the start of each session, normally changing from one session to the next. If you choose static IP address, you need to fill in the following information: ● IP Address: the IP address of the LAN port of the DAG2000-16S; ● Subnet Mask: the netmask of the router connected the DAG2000-16S; ● Default Gateway: the IP address of the router connected the DAG2000-16S; **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.
DBORegStatus	Whether the DAG2000-16S gateway is registered or not.
System Uptime	The running time of the DAG2000-16S since it is powered on.
NTP Status	Succeed: the DAG2000-16S gateway is sync to NTP server successfully; Failed: the DAG2000-16S 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 DAG2000-16S gateway

3.5.2 Registration Information

Port Registration Information

Port No.	Type	Primary User ID	Primary User Status	Secondary User ID	Secondary User Status
0	FXS	1101	Registered	---	---
1	FXS	1102	Registered	---	---
2	FXS	1103	Registered	---	---
3	FXS	1104	Registered	---	---

Port Group Registration Information

Port Group	Port	Primary User ID	Primary User Status	Secondary User ID	Secondary User Status
---	---	---	---	---	---

Figure 3.5-2 Port and Port Group Registration Information

Primary/Secondary User status:

- Registered: the port is registered to SIP server successfully;
- Unregistered: the port fails to be registered to SIP server.

3.5.3 TCP/UDP Statistics

TCP/UDP Traffic

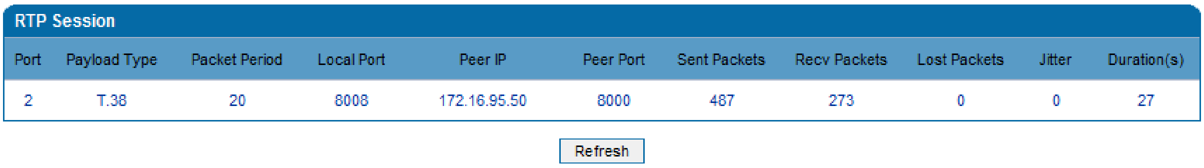
TCP Sent Packets	TCP Recv Packets	UDP Sent Packets	UDP Recv Packets
1092	820	567	311

Refresh

Figure 3.5-3 TCP/UDP Statistics Information

The above interface shows the statistical number of sending or receiving packets over TCP, and the number of sending or receiving packets over UDP since the DAG2000-16S is booted up.

3.5.4 RTP Session Statistics



RTP Session

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

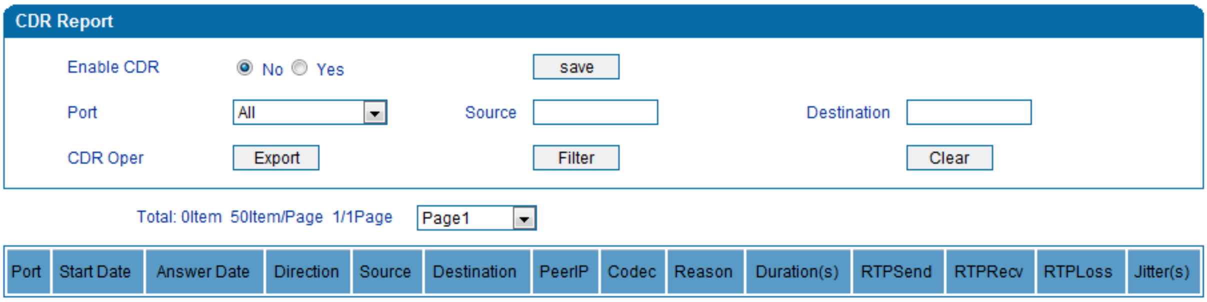
Refresh

Figure 3.5-4 RTP Session Statistics

The above interface shows real-time RTP session information, including: port, payload type, packet period, local port, peer IP, peer port, sent packets, receive packets, lost packets, jitter and duration.

3.5.5 CDR Statistics

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.



CDR Report

Enable CDR	No Yes		save		
Port	All	Source		Destination	
CDR Oper	Export		Filter		Clear

Total: 0Item 50Item/Page 1/1Page Page1

Port	Start Date	Answer Date	Direction	Source	Destination	PeerIP	Codec	Reason	Duration(s)	RTPSend	RTPRecv	RTPLoss	Jitter(s)

On the **Status & Statistic → CDR** interface, details of all calls through the ports of the DAG2000-16S are displayed. The CDR function can be enabled on this interface.

3.6 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 DAG2000-16S device.

3.7 Network Configuration

3.7.1 Local Network

The DAG2000-16S only works in the bridge network mode. It serves as a 16-port Ethernet switch. Under this network mode, user only needs to configure the IP address of LAN port and DNS.

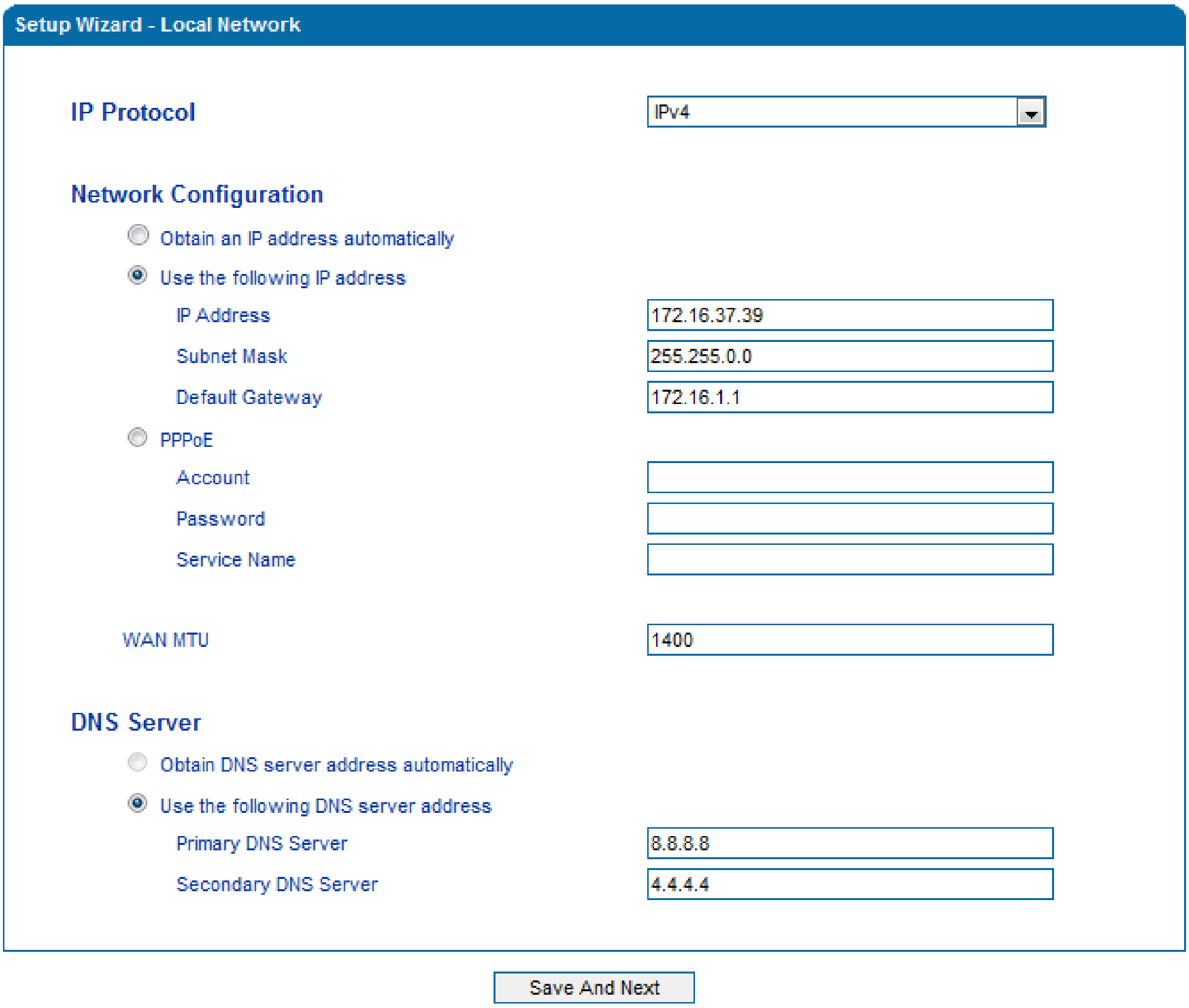


Figure 3.7-1 Local Network

- 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】:

- If DHCP is selected to obtain IP address, please ensure DHCP server in the network works normally.
- After the configurations are finished, please restart the gateway for the configurations to take effect.

3.7.2 VLAN（Virtual Local Area Network）

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.

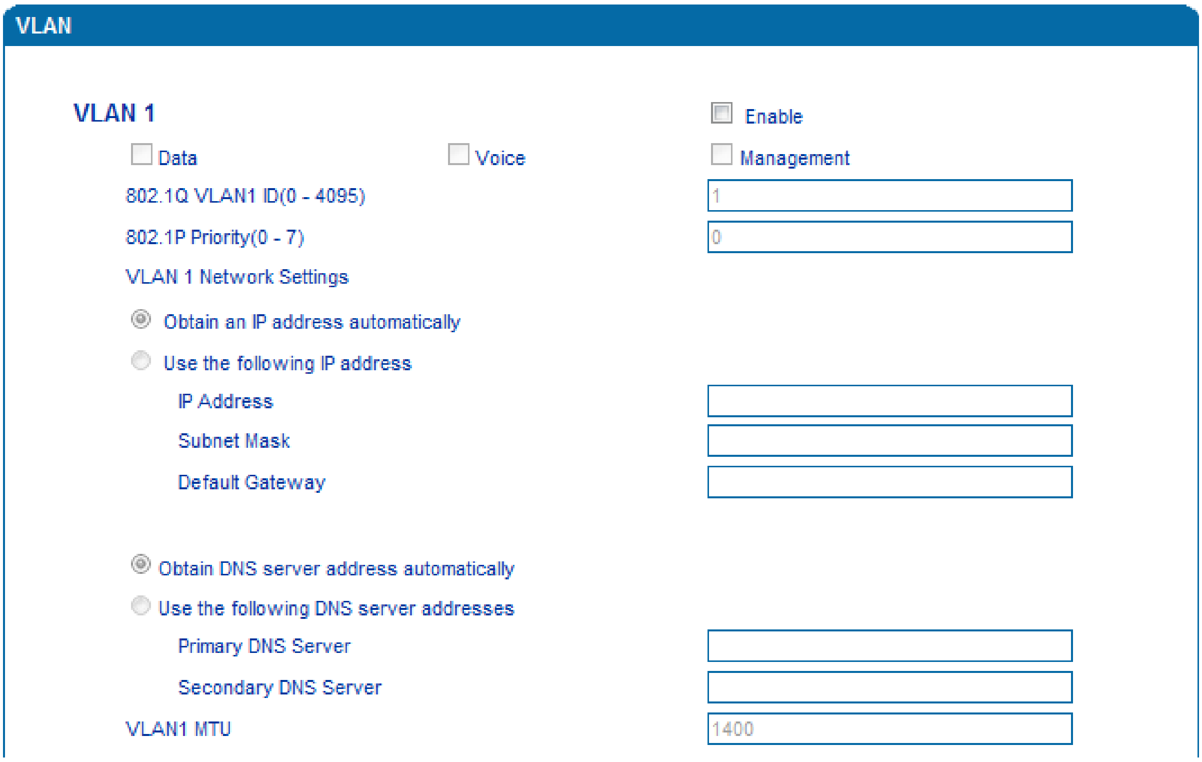


Figure 3.7-3 VLAN parameter configuration

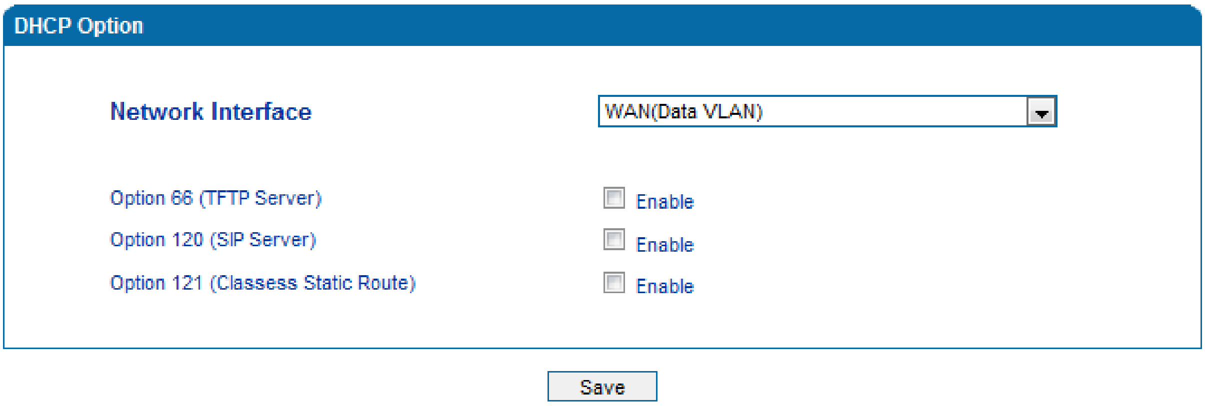
Explanations of the parameters in VLAN interface:

VLAN1/VLAN2/VLAN3	The gateway supports three VLANs at most. Please enable VLAN according to actual needs.

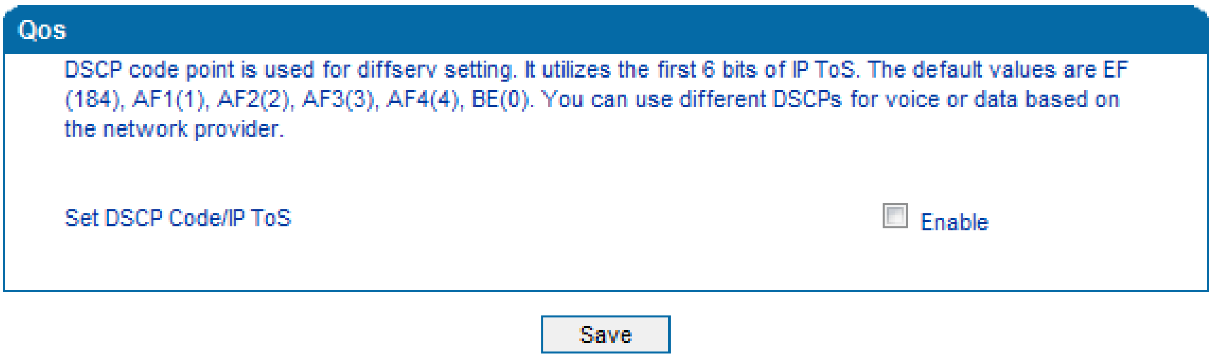
Data/Voice/Management,	If the checkboxes on the right of data, voice and management of VLAN1 are selected, it means data messages, voice messages and management messages are subject to the network setting, 802.1Q VLAN1 ID and 802.1P Priority of VLAN1.
802.1Q VLAN ID(0-4095)	Set an ID to identify a VLAN 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.

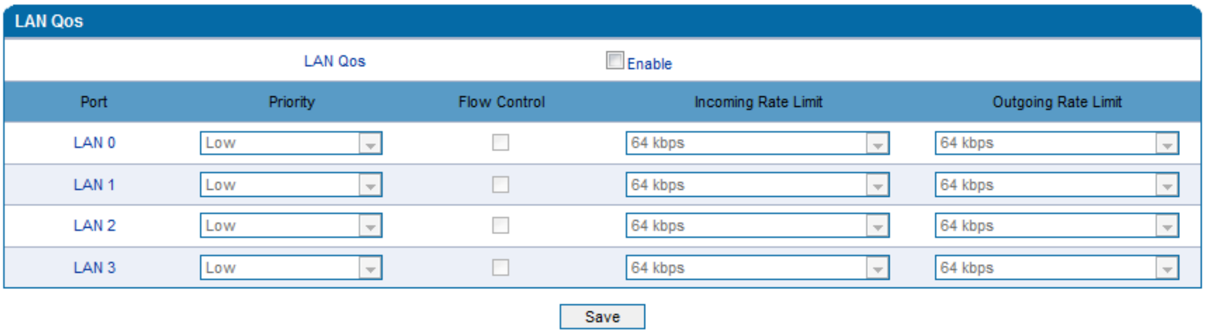
3.7.3 DHCP Option



3.7.4 Qos



3.7.5 LAN Qos



3.7.6 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.

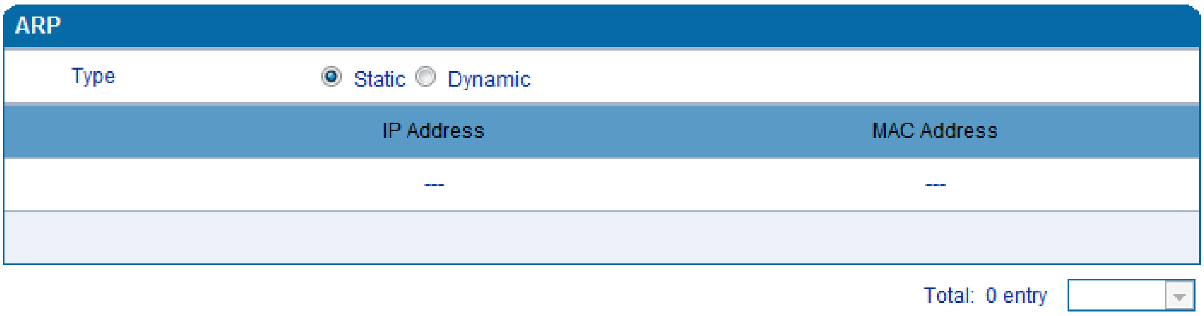


Figure 3.7-9 ARP Parameters

3.8 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.

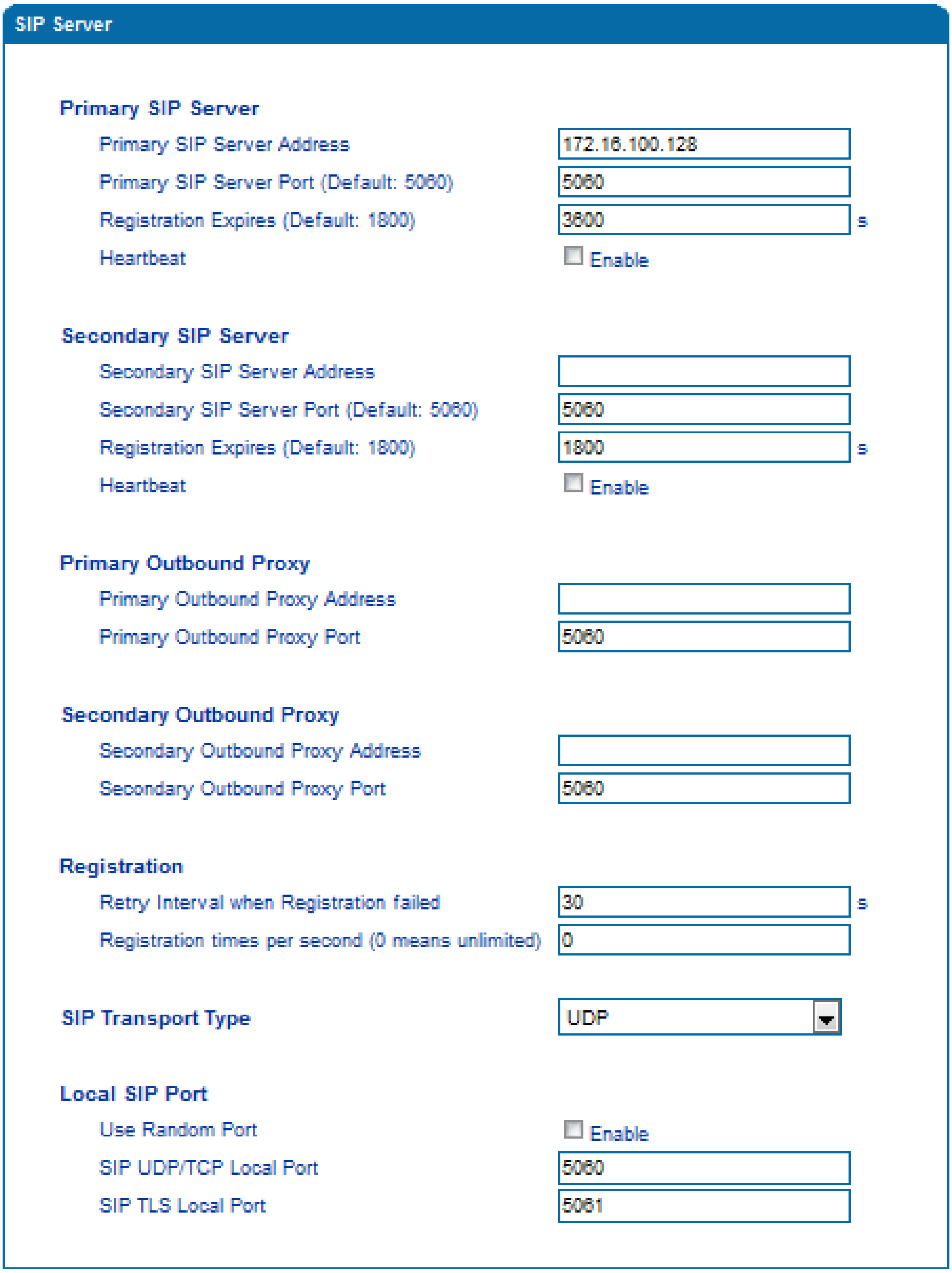


Figure 3.8-1 Configuration Interface for SIP Server

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.

3.9 Port

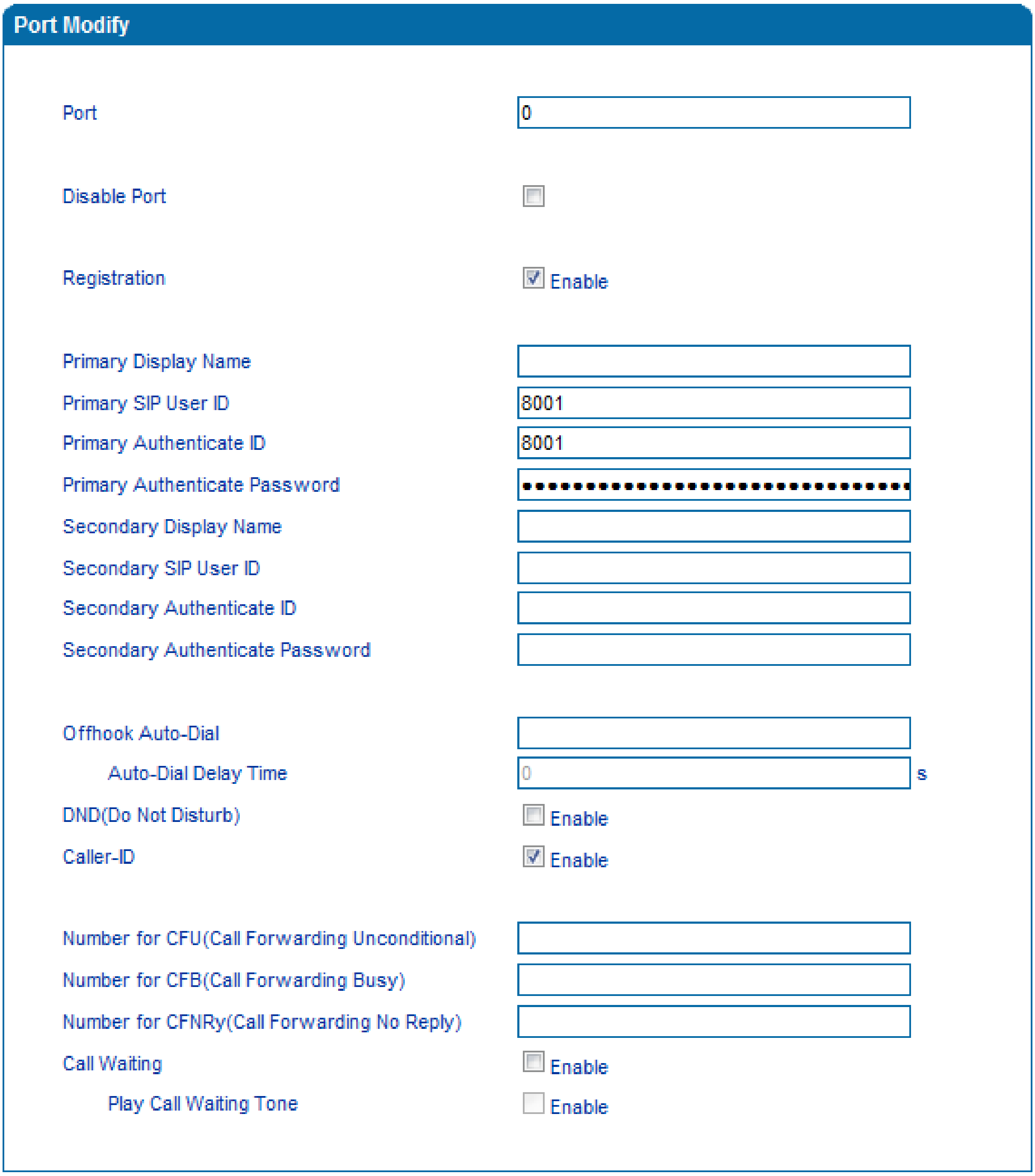


Figure 3.9-1 Port Configuration Interface

Explanations for port parameters:

Port	Port number
Disable port	Whether to disable port temporally
Registration	Whether to enable registration for the port

Primary /Secondary SIP Display Name	Primary /Secondary SIP account description. It is used to identify the SIP account
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 SIP Authenticate ID	SIP service subscriber's authenticate ID used for authentication. It can be identical to or different from SIP User ID.
Primary/Secondary 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.
DND	Do not disturb, the phone won't receive any calls in case it enabled
Caller ID	Enable or disable caller ID for corresponding port. If it is disabled, the caller ID for the calls through the port won't be displayed.
Number for CFU	Call forward unconditional. All incoming calls will be forwarded to pre-assigned number automatically
Number for CFB	Call forward on busy. If the line is busy, the call will be forwarded to pre-assigned number automatically
Number for CFNRy	Call forward no reply. If the call is not answered, the call will be forwarded to pre-assigned number automatically
Call Waiting	If call waiting is enabled, a special tone is sent if another caller tries to reach you
Play Call Waiting Tone	If call waiting tone is enabled, caller will hear special tone.

3.10 Advanced

3.10.1 FXS/FXO Parameters

FXS parameters include: timeout Call Progress Tone, Timeout for Dialing, Send Polarity Reversal etc.

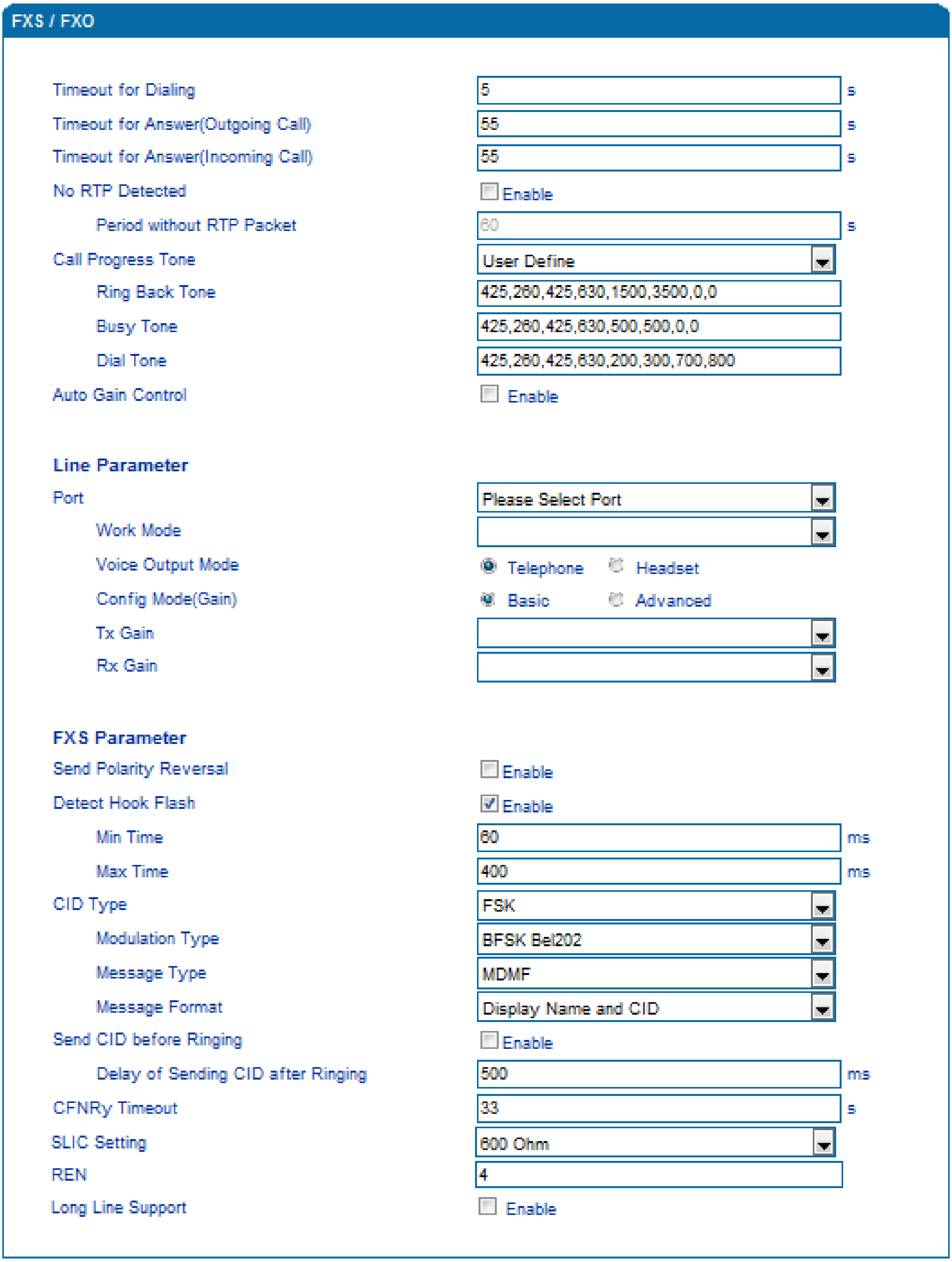


Figure 3.10-1 Configuration Interface for FXS Parameters

Explanation for FXS parameters:

Timeout for dialing	With the help of dialing timeout, you can limit the time between two digits while users are typing the digits of a number through an extension. If the timeout expires, the gateway will consider the dialing has finished and will try to send message to SIP server. Default value is 4 seconds.
Timeout for answer(Outgoing call)	This parameter determines how long the caller party will wait for answer when making outgoing calls through a phone.
Timeout for answer(Incoming call)	This parameter determines how long the phone rings when there are incoming calls
No RTP Detected	If this parameter is enabled, the situation will be detected when there is no RTP packets received during the set time period.
Period without RTP Packet	The time period when there is no RTP packets received.
Call Process Tone	The signal tone standard after a phone is picked up. Choose national standards from the drop-down box. Default value is the United States.
Auto Gain Control	Whether to enable automatic gain control
Send Polarity Reversal	If polarity reversal is enabled, call tolls will be calculated based on the changes in voltage. If polarity reverse is disabled, you need to set the time for offhook detection and call tolls will be calculated starting from the set time.
Detect Hook flash	If 'Detect Hook Flash' is enabled, you need to set a minimum time and a maximum time. If a phone's hook flash is pressed for a time period greater than the set minimum time but less than the maximum time, the action is considered as a 'hook flash' operation. If a phone's hook flash is pressed for more the set maximum time, the action is considered as 'hang up the phone'.
CID Type	There are two CID types, namely DTMF and FSK.
Message Type	There are two call display types including SDMF and MDMF
Message Format	The call display format in analog phone. It can be "Display Name and CID", "CID only", or "Display Name only"; default value is "Display Name and CID"

Send CID before Ringing	If this parameter is enabled, the gateway send Caller ID to phone before ringing, otherwise the caller ID will be displayed after ringing.
Delay of sending CID after Ringing	The time how long the caller ID will be delayed when the caller ID is set to be displayed after ringing. Default value is 500ms.
CFNRy Timeout	Timeout for 'call forwarding on no answer' service
SLIC Setting	Impedance matched with analog phone.
Long Line Support	Whether to enable 'Long Analog Extension Line'.

3.10.2 Media Parameter

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

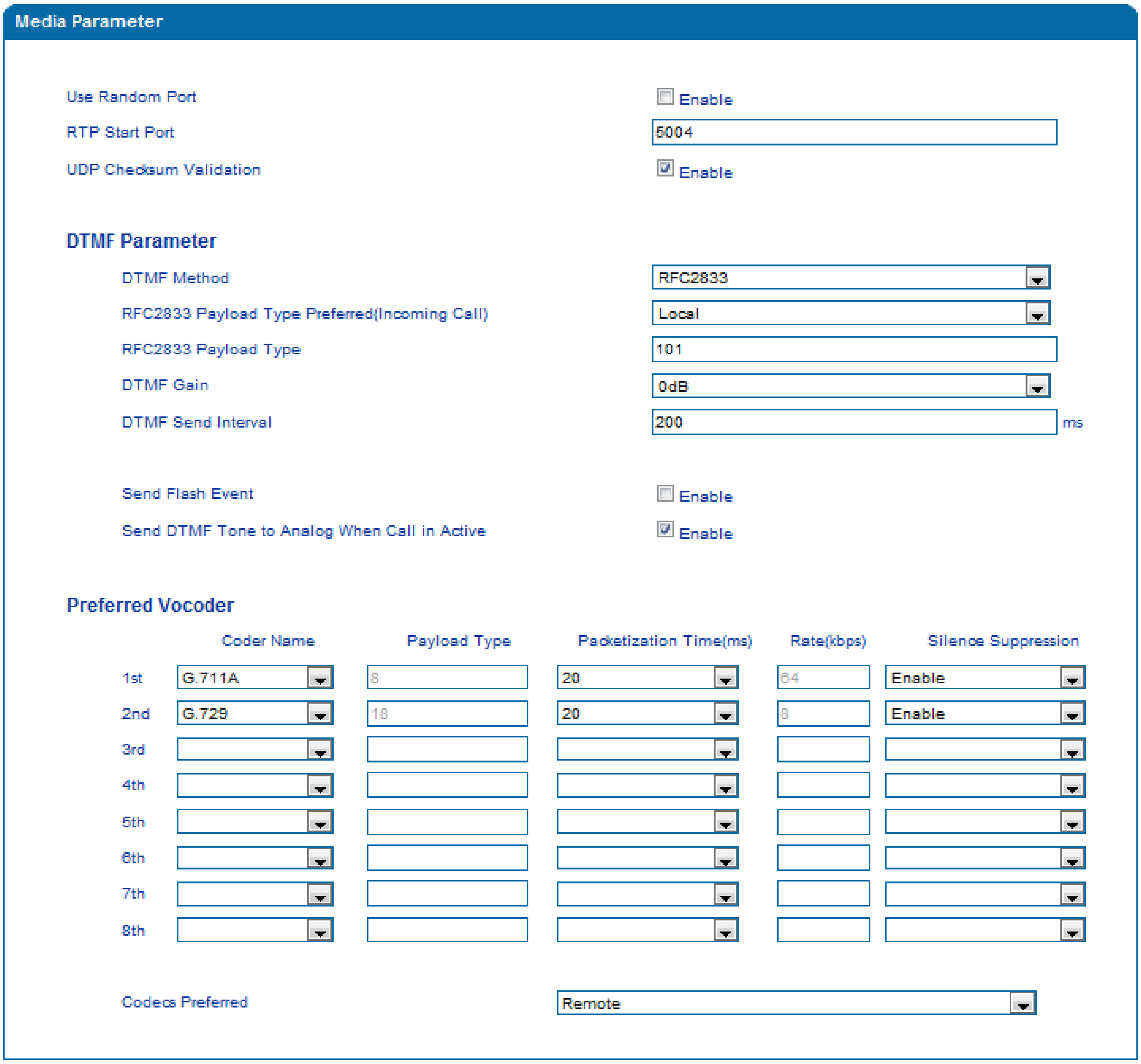


Figure 3.10-2 Configuration Interface for Media Parameters

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.

3.10.3 SIP Parameters

SIP Parameter

SUBSCRIBE for MWI(Message Waiting Indicator)	☐ Enable
MWI Subscription Expires(Default: 3600)	3600 s
Voicemail User ID	
Visual MWI Type	NEON
RFC3407 Support	☐ Enable
IP-to-IP Call	☑ Enable
URI includes "user=phone"	☐ Enable
INVITE with "P-Preferred-Identity" Header (RFC3325)	☐ Enable
Only Accept Calls from ACL(SIP Server or IP Trunk)	☐ Enable
Anonymous Call	☐ Enable
Reject Anonymous Call	☐ Enable
'#' as Ending Dial Key	☐ Enable
'#' Escape	☐ Enable
Send '#' when First Dial Number is '*'	☑ Enable
Value of "Refer To" refers to "Contact"	☐ Enable
Third Party Do Not Send 18x Response	☐ Enable
REFER Delay	☐ Enable
Send BYE when Recv REFER Response(Unattended)	☐ Enable
Send New REGISTER when Recv 423 Response	☑ Enable
Cseq Start with 1	☐ Enable
Forbid Invalid m=line in reINVITE	☐ Enable
Call Confirm Tone	☐ Enable
RTP Mode in SDP when Call Holding	sendonly
Support Call Waiting of Huawei IPPBX	☐ Enable
Accept Orphan 200 Ok	☐ Enable
Called Number Preferred	Request-Line
Caller-ID Preferred	From Header
Report SDP Whatever	☐ Enable
18x Response Preferred	18x Response with SDP
FlashHook Operation Mode	Mode three
Wait Dial Time	5 s
Attended Transfer Trigger	Flashhook+4
Domain Query Type	A Query
Domain Re-resolution Inteval(0 means disable)	0 min
DNS Cache	☑ Enable

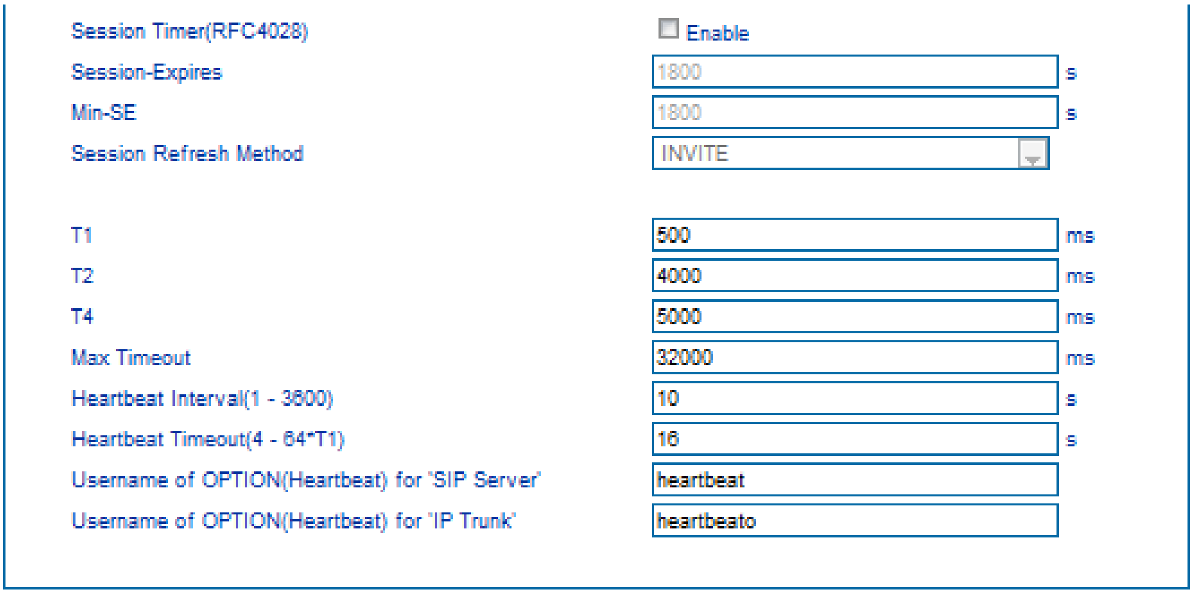


Figure 3.10-3 SIP Parameter Configuration Interface

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.

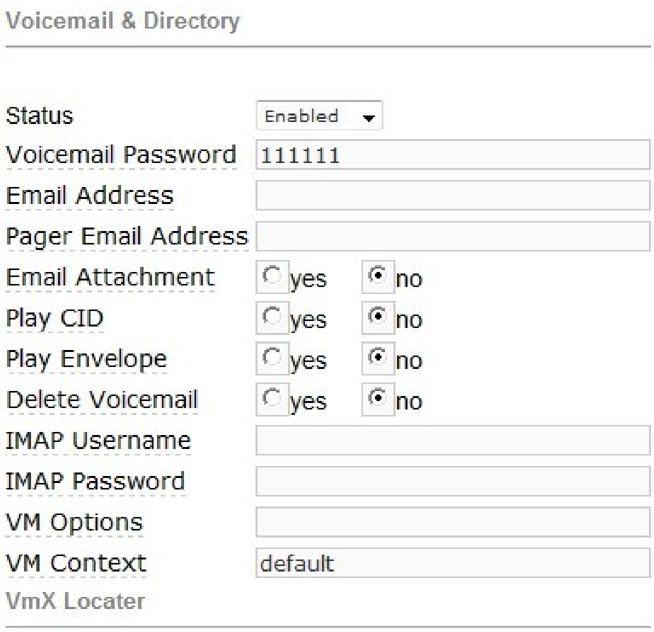
Reject Anonymous Call	If this parameter is enabled, all anonymous calls will be rejected. Default value is 'not disable'.
# as ending Dial Key	'# ' is used as the end mark for dialing.
# Escape	If this parameter is enabled, '#' is considered as a digit of the number that is dialed.
Value of "Refer To" 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 Response	If this parameter is enabled, the third party will not send 18x 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 'invilad m=line' from being carried in the SDP of re-INVITE.
RTP Mode in SDP when Call Holding	Use 'sendonly ' 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
Username of OPTION(Heartbeat) for "IP TRUNK"	The user ID part of OPTION SIP message in the heartbeat request for IP trunk

Voicemail instructions:

Here takes the DAG2000-16S gateway together with Elastix as the example to introduce how voicemail works in the gateway.

1） After the gateway registers to Elastix server, enable the voicemail function in Elastix for the corresponding extension number and then set password. As below：

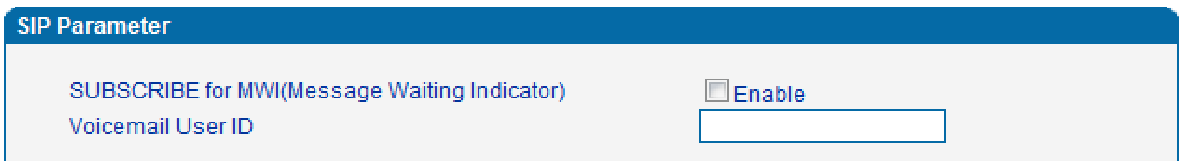


Elastix Voicemail Configuration Interface

2）Check feature code in Elastix and change it if necessary. Its default feature code setting is as follows:

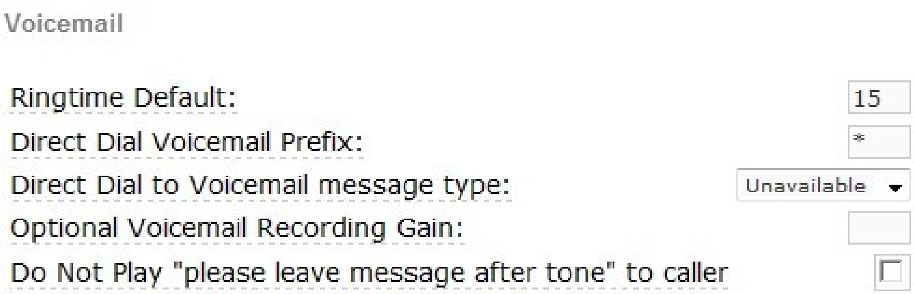
Elastix Voicemail Setting

On the Web interface of DAG2000-16S, click **Advanced → SIP Parameter** in the navigation tree and then enter voicemail User ID.



VoiceMail Setting in SIP Parameter

3）Set ringing time in Elastix. Elastix will prompt user to leave a message after the corresponding extension rings 15 seconds (by default). Then the Elastix sever will record the message. Related setting is shown as follows:



Voicemail Setting

4）Dial *200# on the extension which is connected to DAG2000-16S, then dial voicemail user ID and enter password for authentication. After that user will hear voice message.

3.10.4 Fax Parameter

Figure 3.10-4 Configuration Interface for Fax Parameter

Explanation of fax parameters:

Fax Mode	There are four fax modes: T.38, T.30(Pass-through),Modem and Adaptive.
Include “a=X-fax” Attribute	If this parameter is enabled, “a=X-fax” attribute will be carried in SDP
Include “a=fax” Attribute	If this parameter is enabled, “a=fax” attribute will be carried in SDP

Include "a=X-modem" Attribute	If this parameter is enabled, "a=X-modem" attribute will be carried in SDP
Include "a=modem" Attribute	If this parameter is enabled, "a=modem" attribute will be carried in SDP
ECM	Whether to enable 'Error Correction Mode'.
Rate	The rate of sending or receiving fax
Tone Detection by	Fax sound is detected by caller, callee or automatically

3.10.5 Digit Map

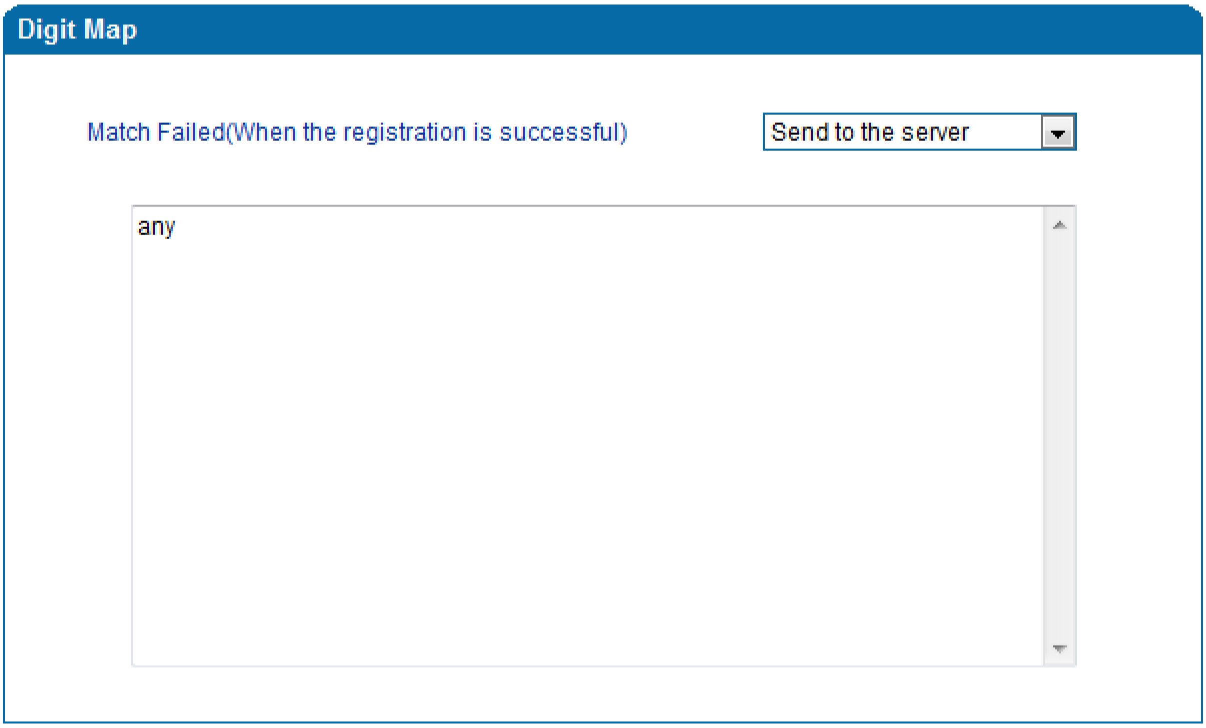


Figure 3.10-5 Digit Map

Digit Map Syntax

Supported objects	Digit	0-9
	T	Timer
	DTMF	A digit, a timer, or one of the symbols of A, B, C, D, #, or *.
Range	[]	One or more DTMF symbols enclosed in the [], but only one DTMF symbol can be selected.

Range	()	One or more expressions enclosed the (), but only one can be selected.
Separator	|	Separated expressions or DTMF symbols.
Subrange	-	Two digits separated by hyphen (-) which matches any digit between a nd including the two.
Wildcard	x	Matches any digit of 0 to 9
Modifiers	.	Matches 0 or more times of the preceding element
Modifiers	?	Matches 0 or 1 times of the preceding element

Examples:

(13 | 15 | 18)xxxxxxxxx	Matches the phone numbers with stating digits as 13, 15 or 18 and the left nine digits as any of 0 to 9.

3.10.6 Feature Codes

Please make reference to 2.7 Description of Feature Codes and the following table.

Inquiry LAN port IP address	Dial*158# to obtain device's LAN port IP address
Inquiry Phone Number	Dial*114# to obtain port account
Inquiry PortGroup Number	Dial *115# to obtain port group number
Setting IP Mode	*150*0#, means pppmodem, *150*1#, means static IP, *150*2#, means obtain IP address by DHCP, *150*3#, means pppoe.
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

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

3.10.7 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.

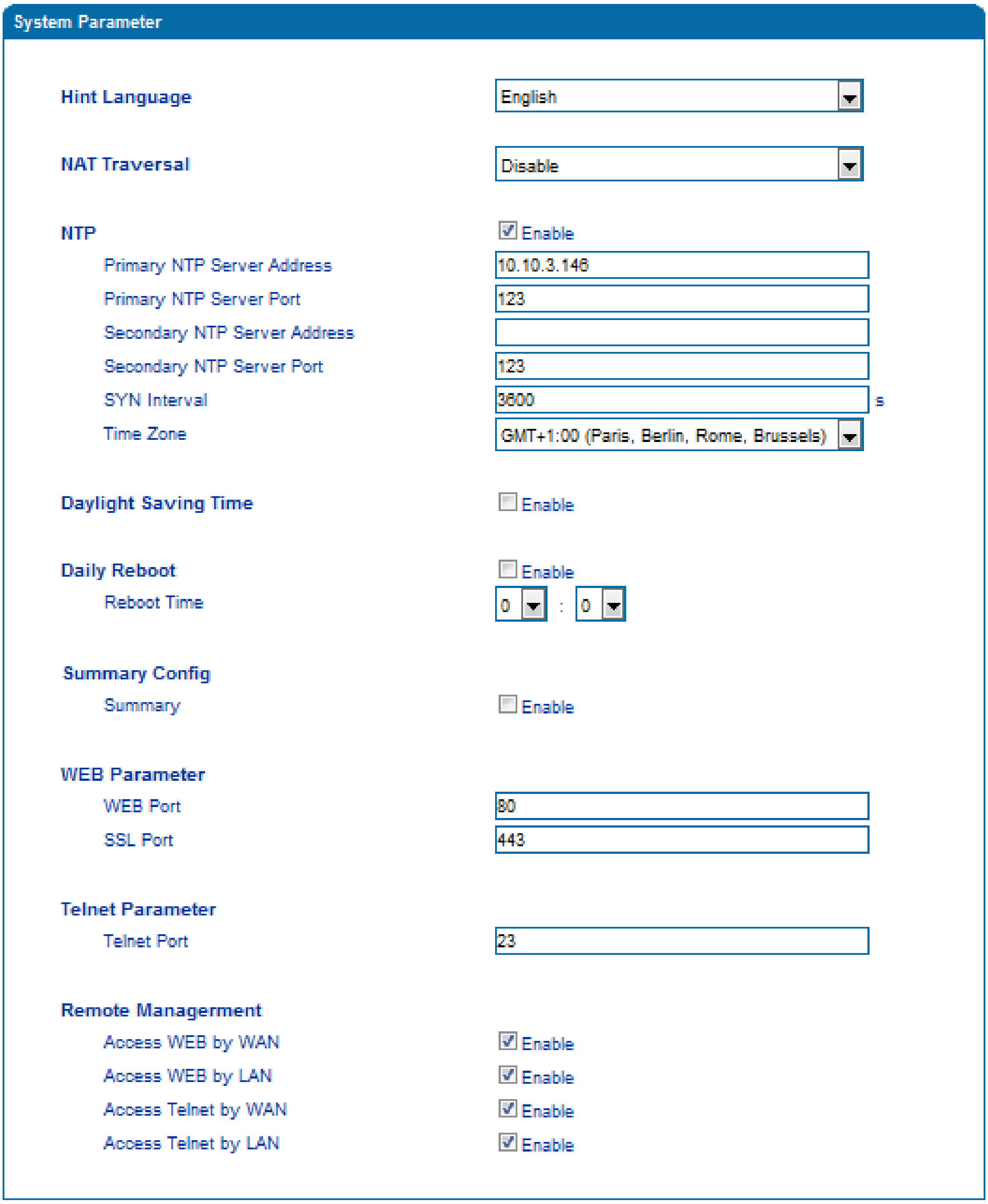


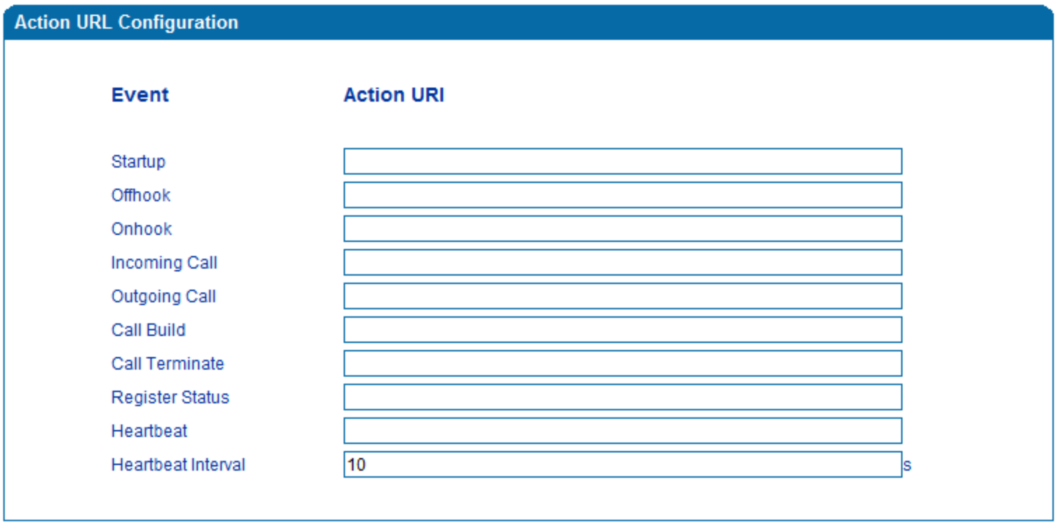
Figure 3.10-7 Configuration Interface for System Parameters

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 DAG2000-16S. 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

3.10.8 Action URL

Action URL can be used as a means to allow the VoIP platform to learn about the DAG gateway's status. It transmits data via GET request over the HTTP protocol. The DAG gateway is an HTTP client. At HTTP server side, GET request must be processed by the VoIP platform. Thus, the purpose is achieved.



3.11 Call & Routing

3.11.1 Wildcard Group

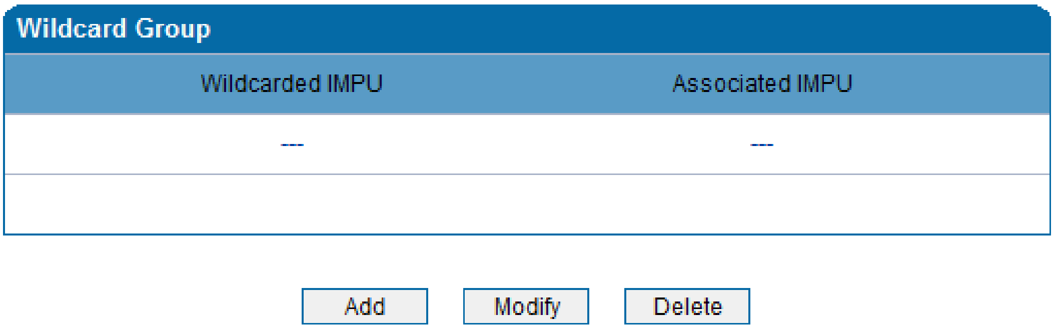


Figure 3.11-1 Wildcard Group

3.11.2 Port Group

On the **Port Group** interface, user 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.

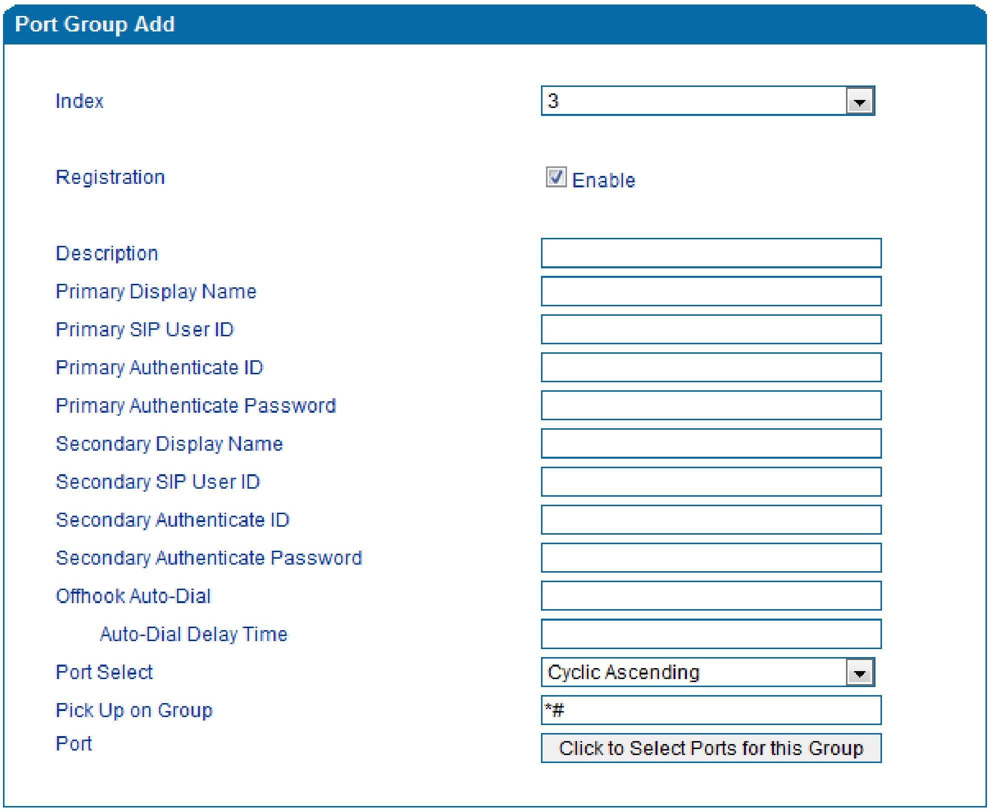


Figure 3.11-2 Configuration Interface for Port group

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

3.11.3 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.

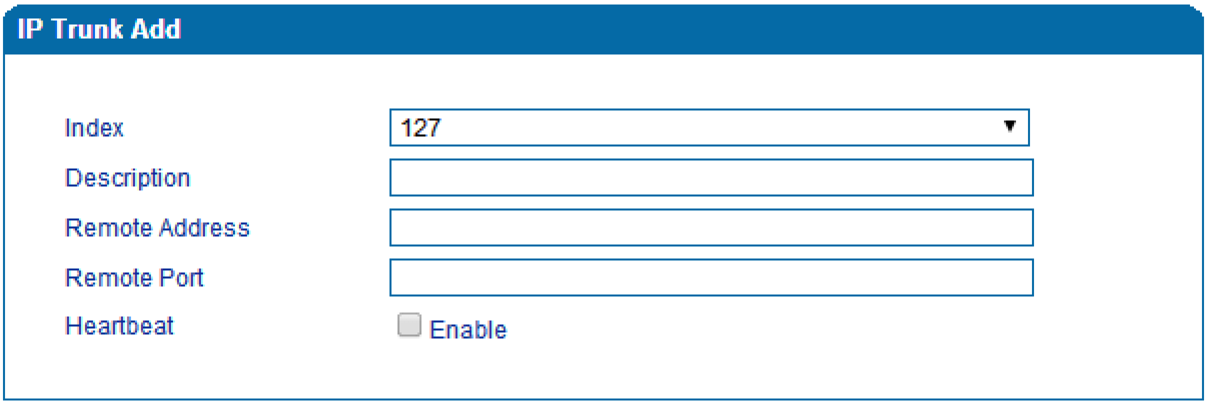


Figure 3.11-3 IP Trunk Configuration Interface

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.

3.11.4 Routing Parameter

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

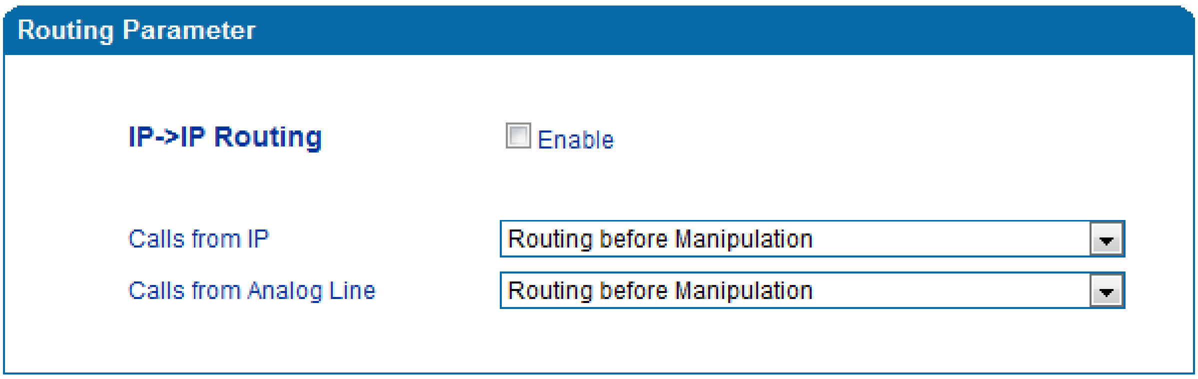


Figure 3.11-4 Configuration Interface for Routing Parameter

3.11.5 IP -> Tel Routing

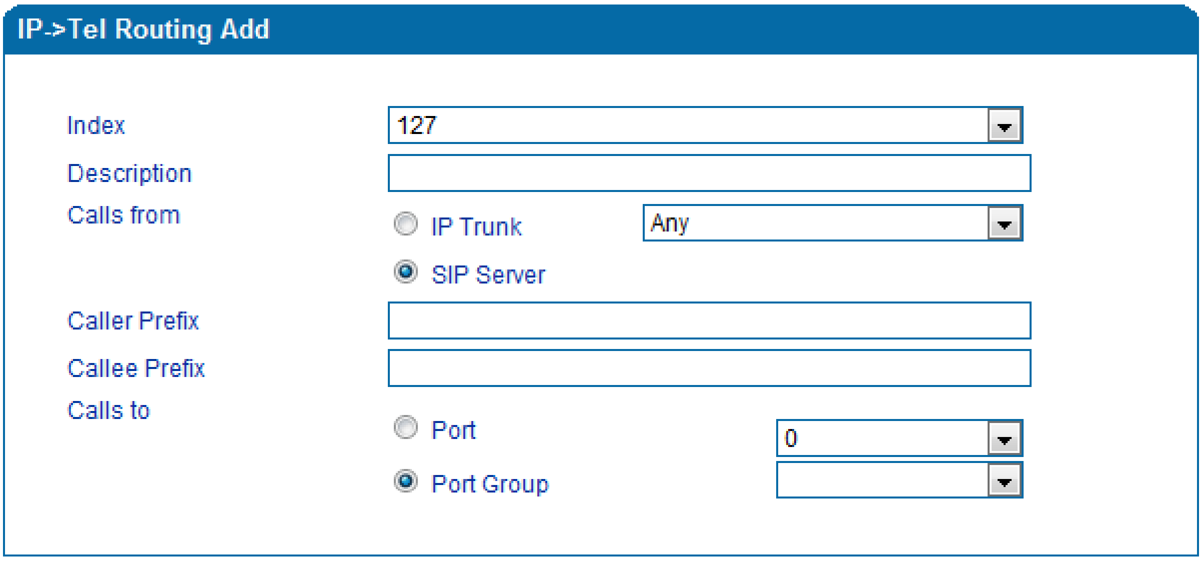


Figure 3.11-5 Configuration Interface for IP-Tel Routing

Explanation of related parameters:

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

3.11.6 Tel-IP/Tel Routing

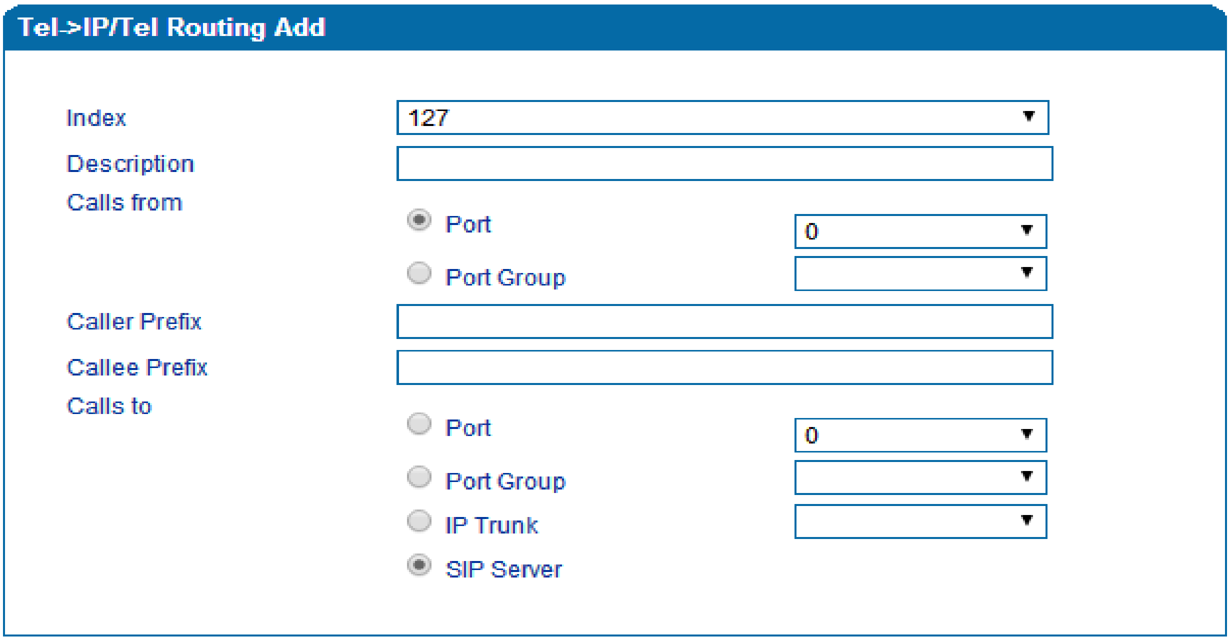


Figure 3.11-6 Configuration Interface for Tel-IP/Tel Routing

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

3.11.7 IP – IP Routing

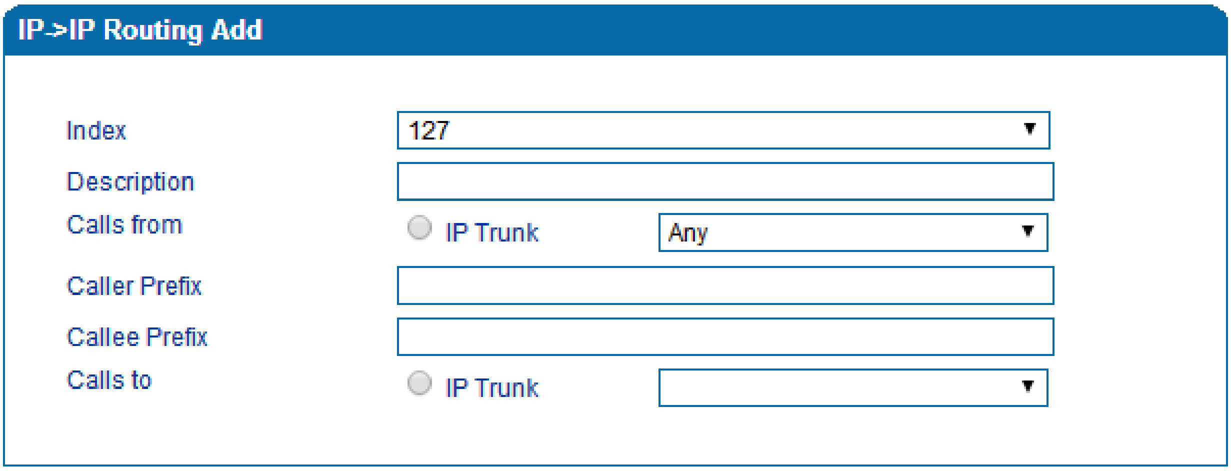


Figure 3.11-7 Configuration Interface for IP->IP Routing

Explanation of related parameters:

Index	The index of this IP →IP 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	Calls are from IP trunk.
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 IP trunk

3.12 Manipulation Configuration

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.

3.12.1 IP -> Tel Callee

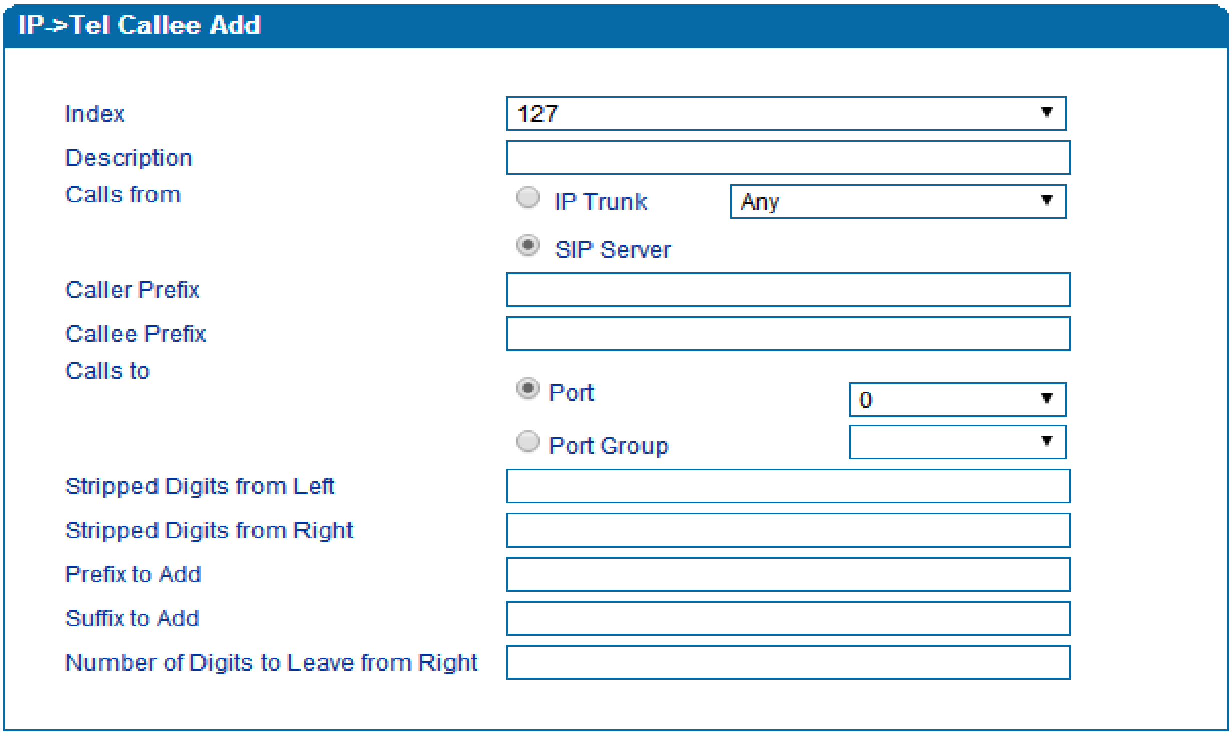


Figure 3.12-1 Add IP -> IP Callee

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

3.12.2 Tel -> IP/Tel Caller

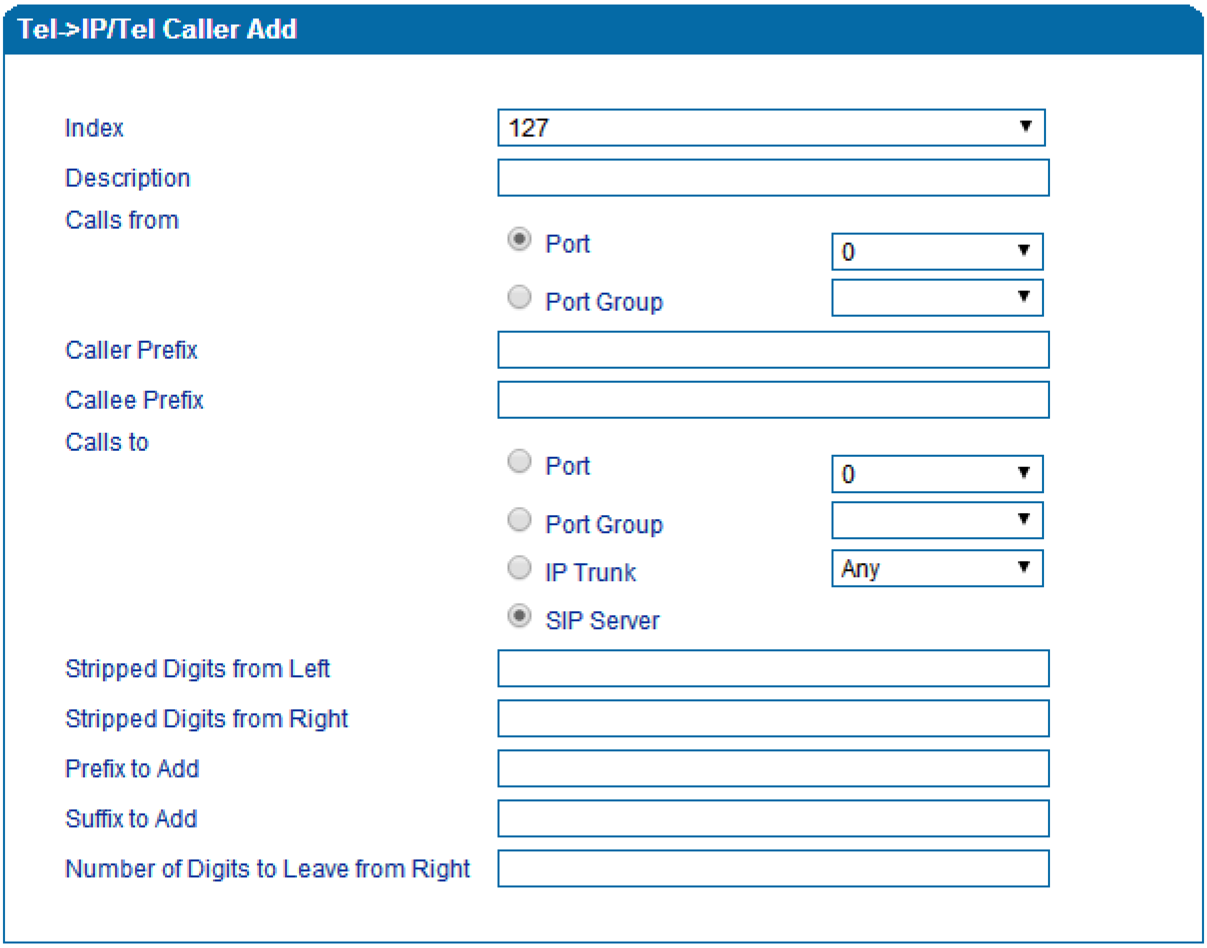


Figure 3.12-2 Add Tel -> IP Caller

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

3.12.3 Tel-IP/Tel Callee

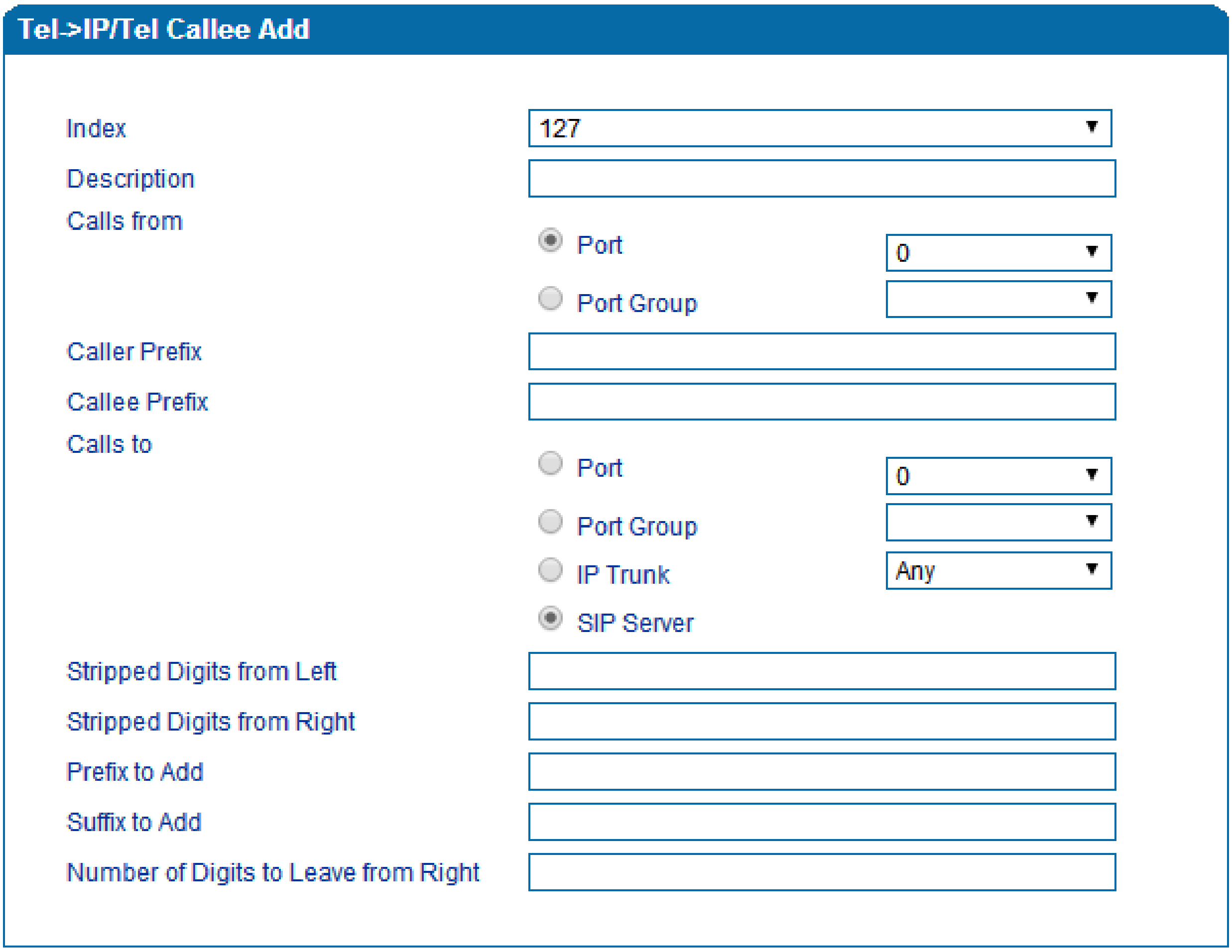


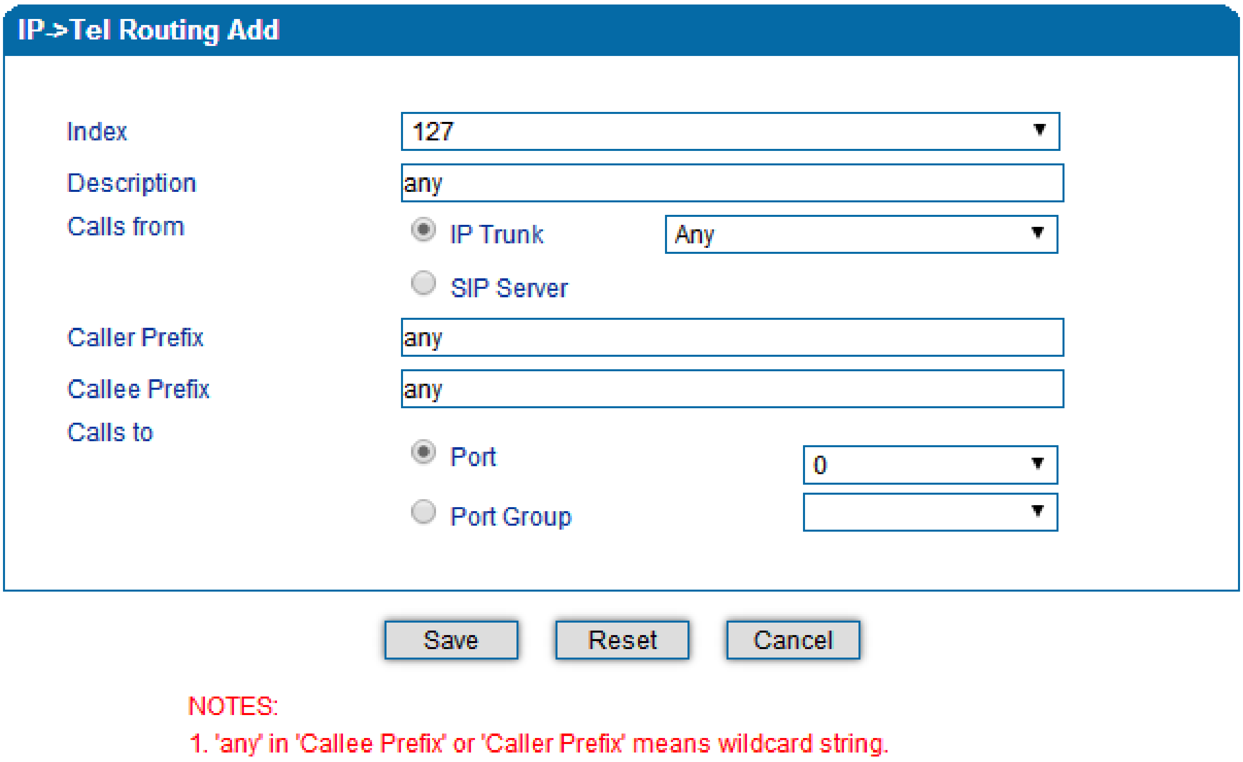
Figure 3.12-3 Add Tel-IP Callee

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

3.13 Routing rule examples

3.13.1 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.

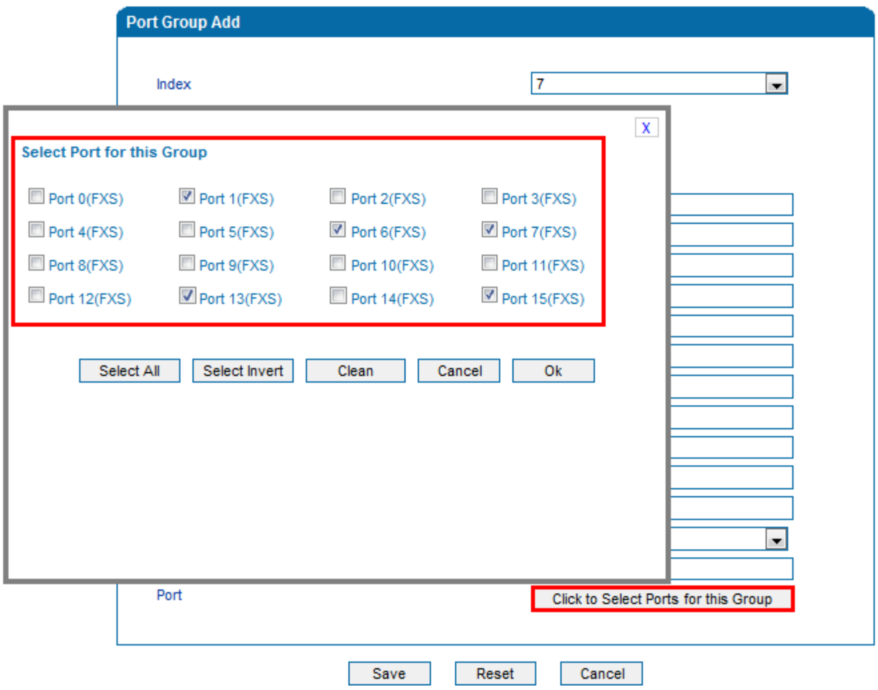


In the example above, all calls will be routed to port 0 when the routing rule is matched.

3.13.2 Route any calls from any IP to specified port group

▸Create port group

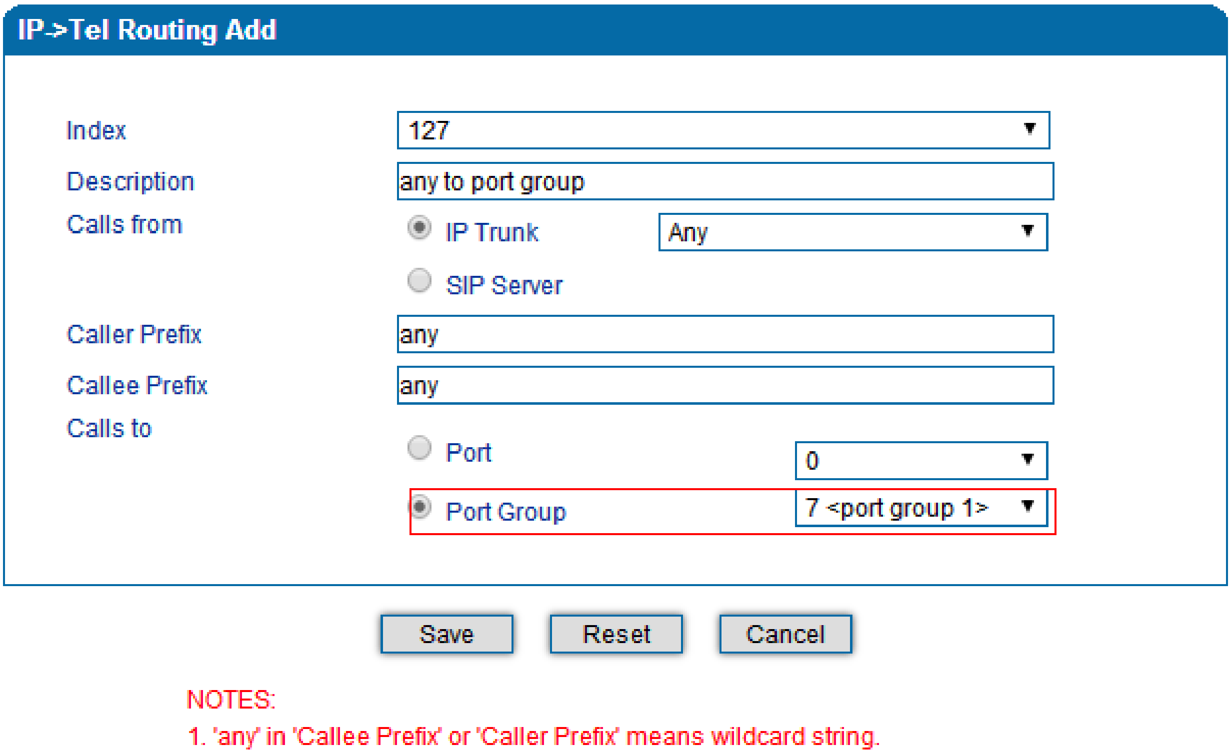
Before we can route calls to a port group, create the port group first as below. On the **Call & Routing → Port** Group, click **Add** to create a new port group.



Port 1, port 6, port 7, port 13 and port 15 are assigned to port group 7.

▸Route any calls to the port group

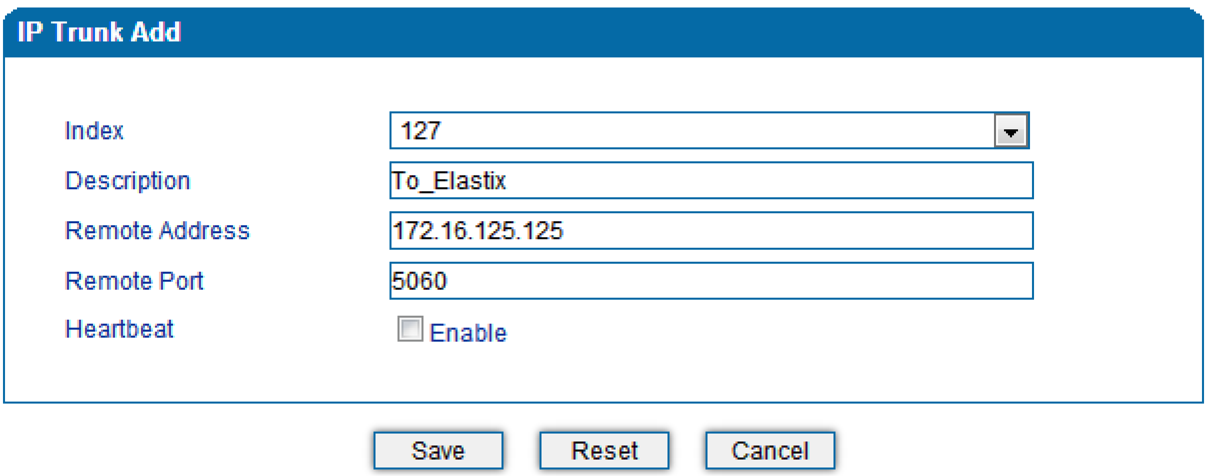
On the **Call & Routing → IP-Tel Routing** interface, click **Add** to create a new routing rule.



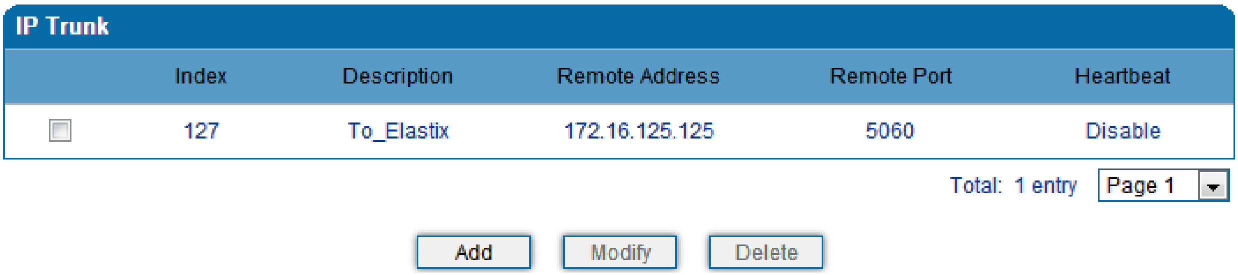
As shown above, if the routing rule is matched, calls will be routed to port group 7.

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

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



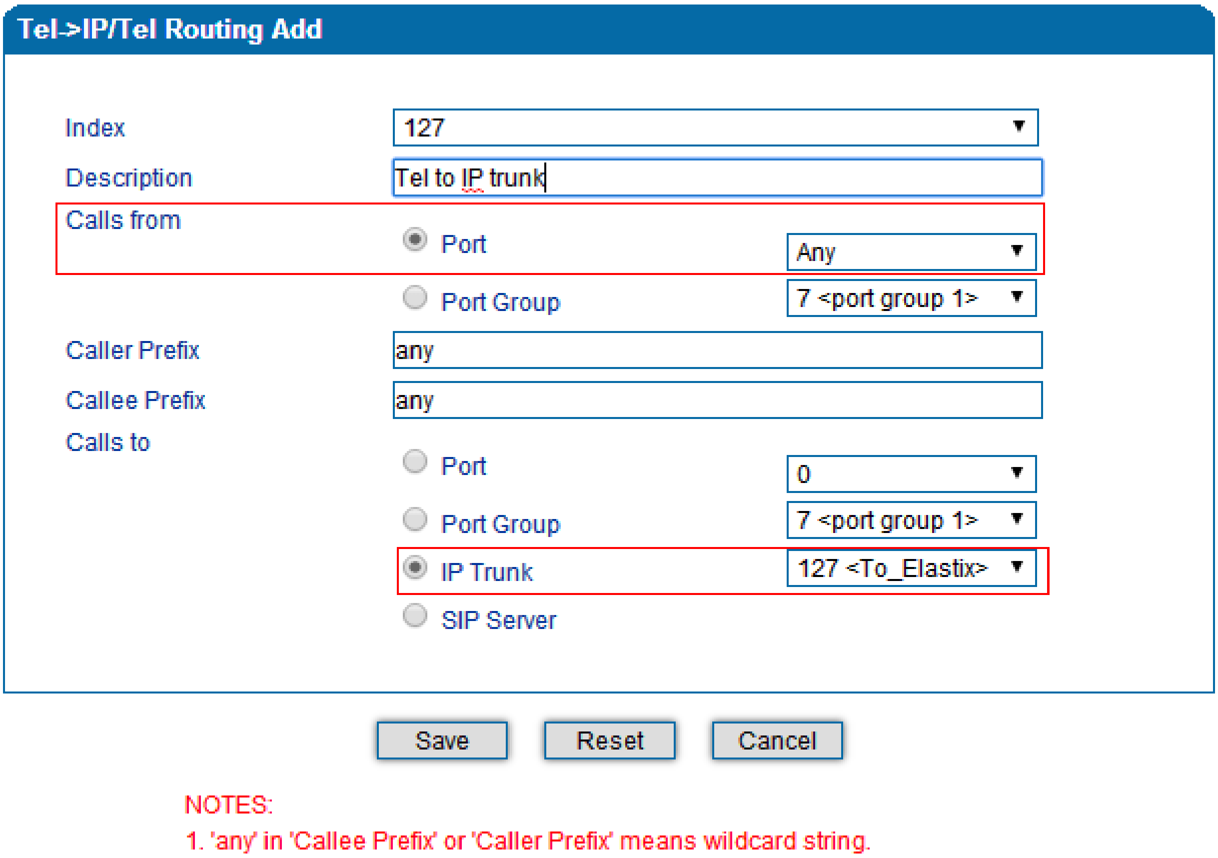
After IP Trunk is created, check the following configuration:



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.

3.14 Maintenance

3.14.1 TR069

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

Username and password are used for ACS authentication.

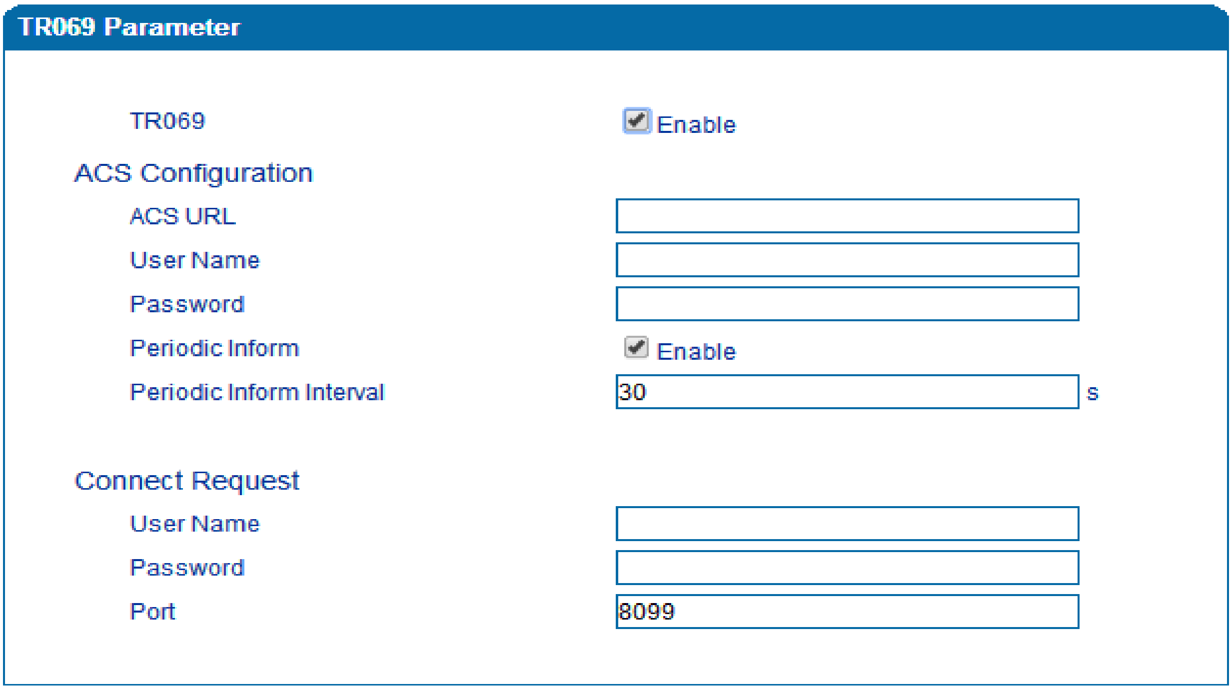


Figure 3.14-1 TR069 Parameters

3.14.2 SNMP (Simple Network Management Protocol)

SNMP Parameters:

- SNMP enable: to disable or enable the SNMP feature
- SNMP version: the DAG2000-16S gateway supports SNMP v1 and v2
- Community: the community name used to read through SNMP protocol
- Source: the IP address of SNMP server

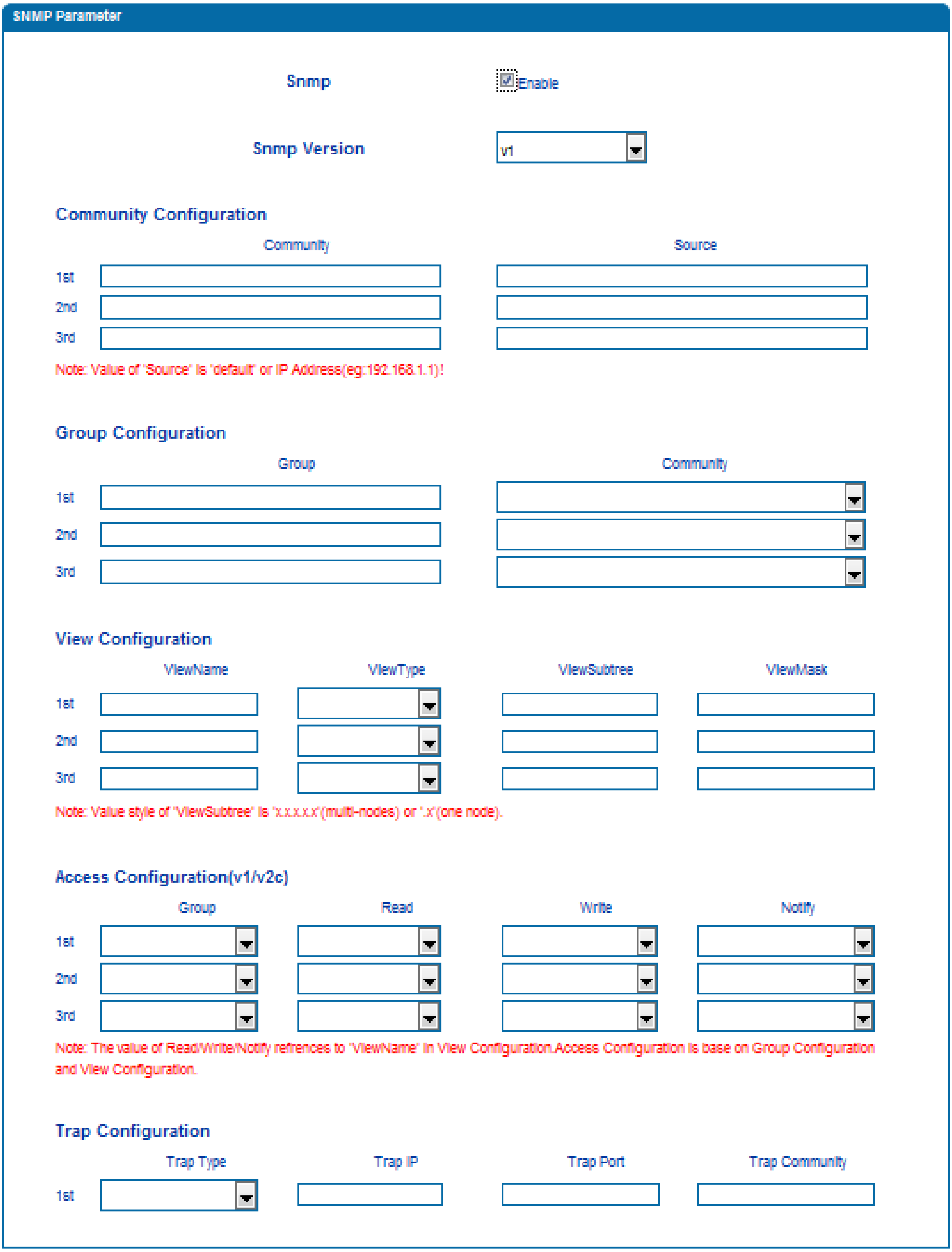
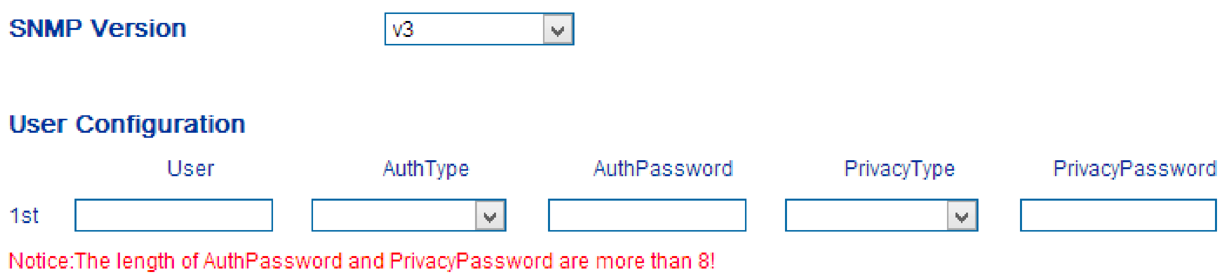


Figure 3.14-2 SNMP Parameters

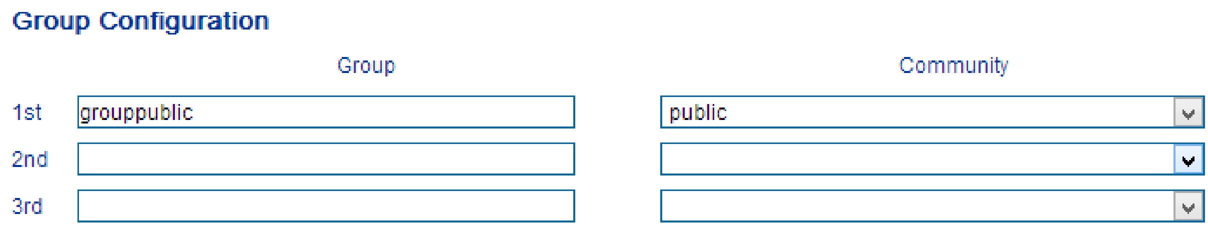
User configuration is only available on SNMP v3.



Group configuration

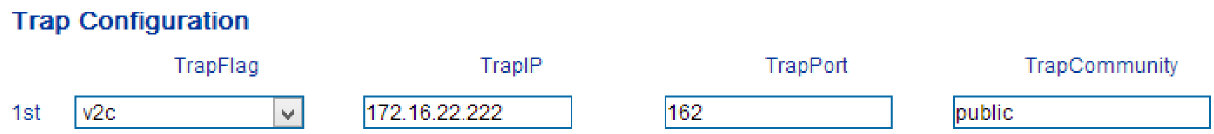
Group: community group name which consist of character string.

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.



3.14.3 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.

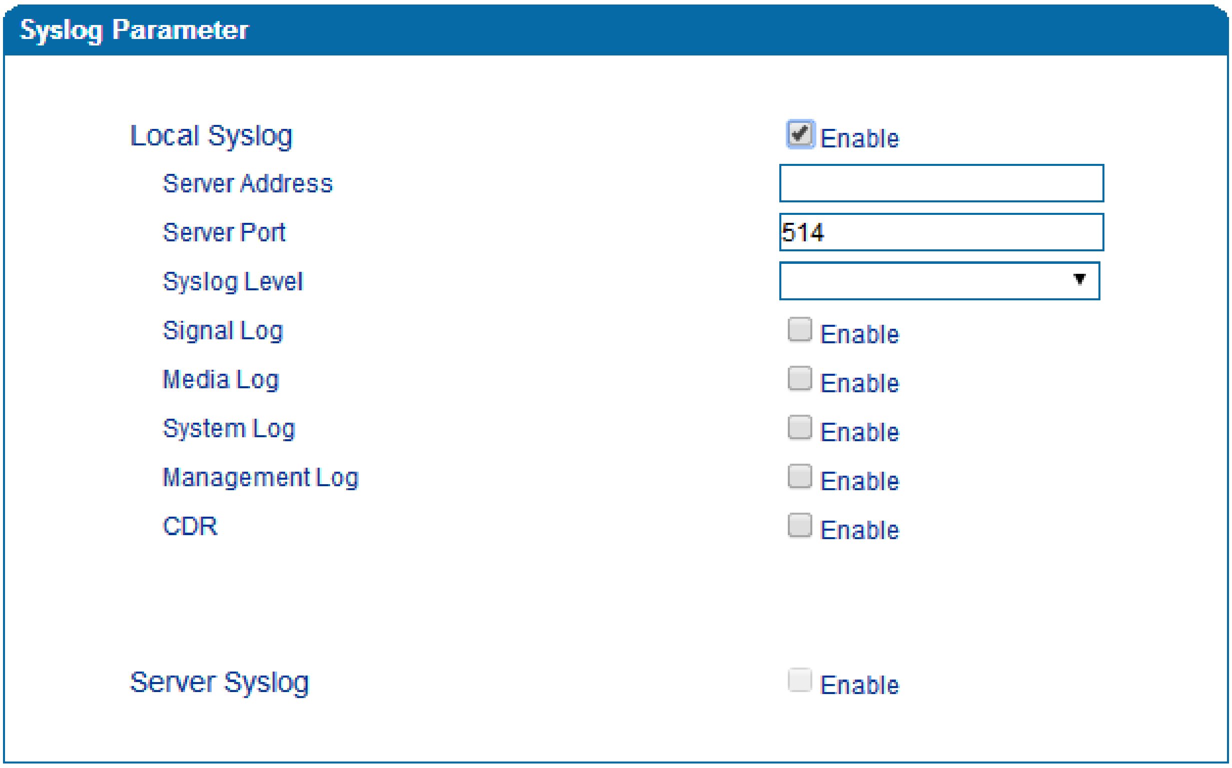


Figure 3.14-3 Syslog Parameter

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

3.14.4 Provision

Provision is used to make the DAG2000-16S automatically upgrade with the latest firmware stored on an http server an ftp server or a tftp server.

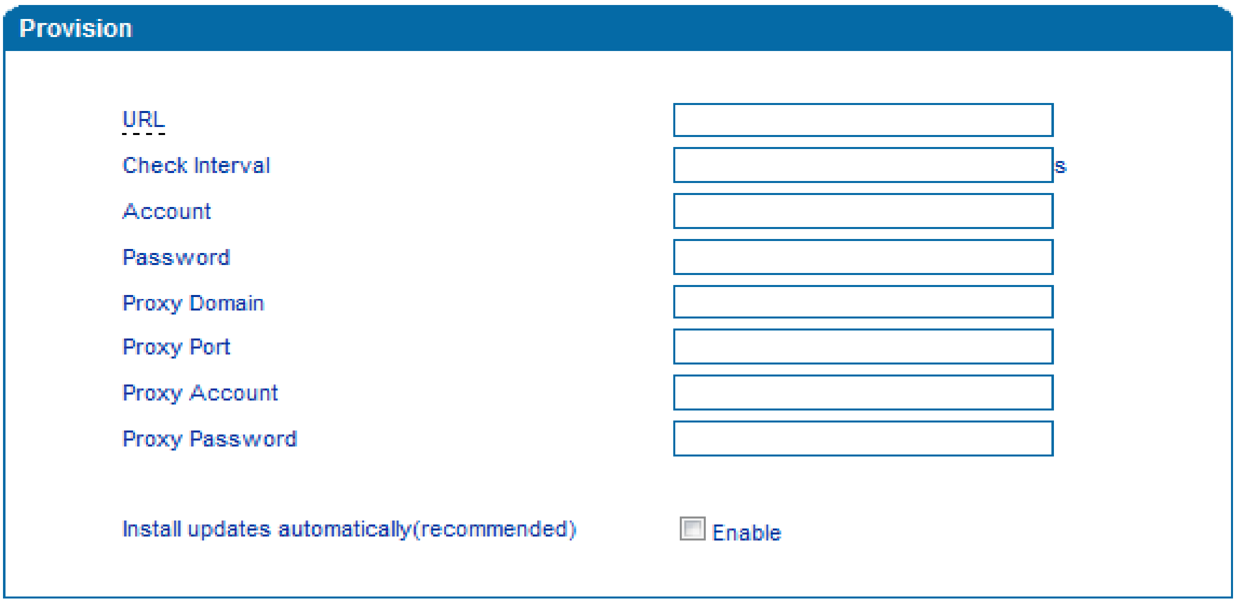


Figure 3.14-4 Provision

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

3.14.5 Cloud Server

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

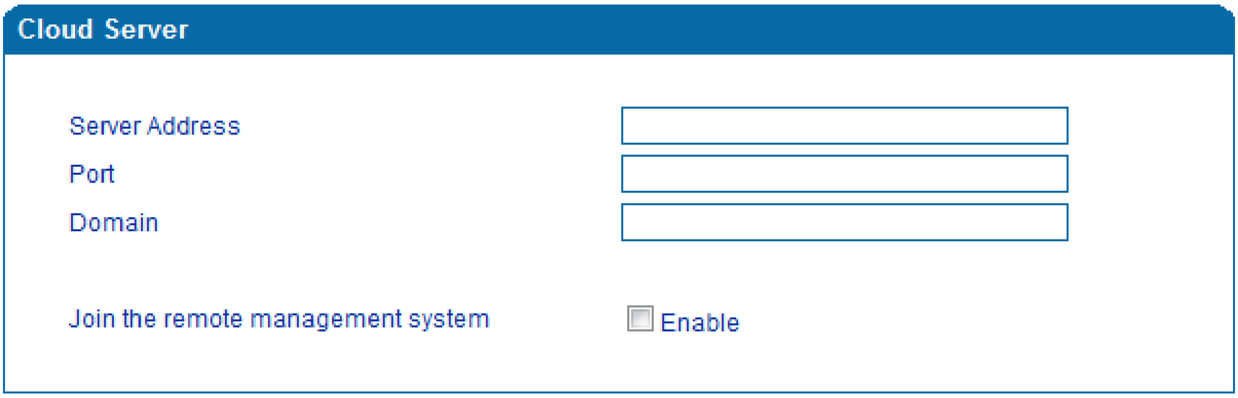


Figure 3.14-5 Cloud Server

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

3.15 User Manage

On the following interface, user can choose whether to enable the ‘User Manage’ function.

3.16 Remote Server

On the following interface, user can choose whether to enable remote server.

3.17 Record Parameter

On the following interface, user can choose whether to enable the record function.

3.18 Security

3.18.1 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.

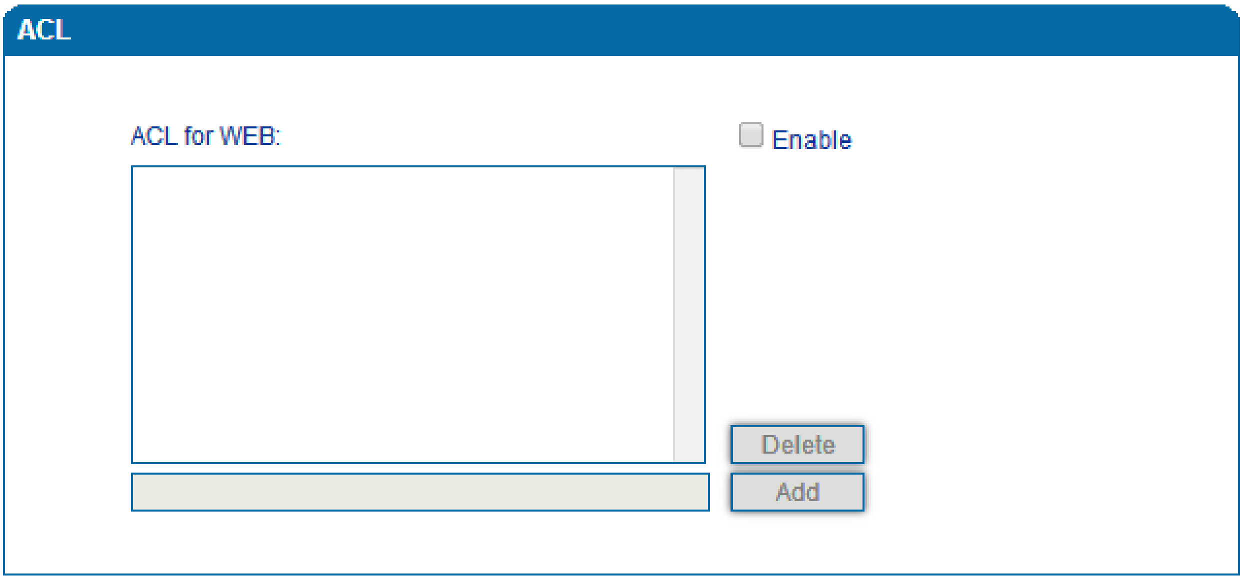


Figure 3.15-1 ACL for WEB

3.18.2 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.

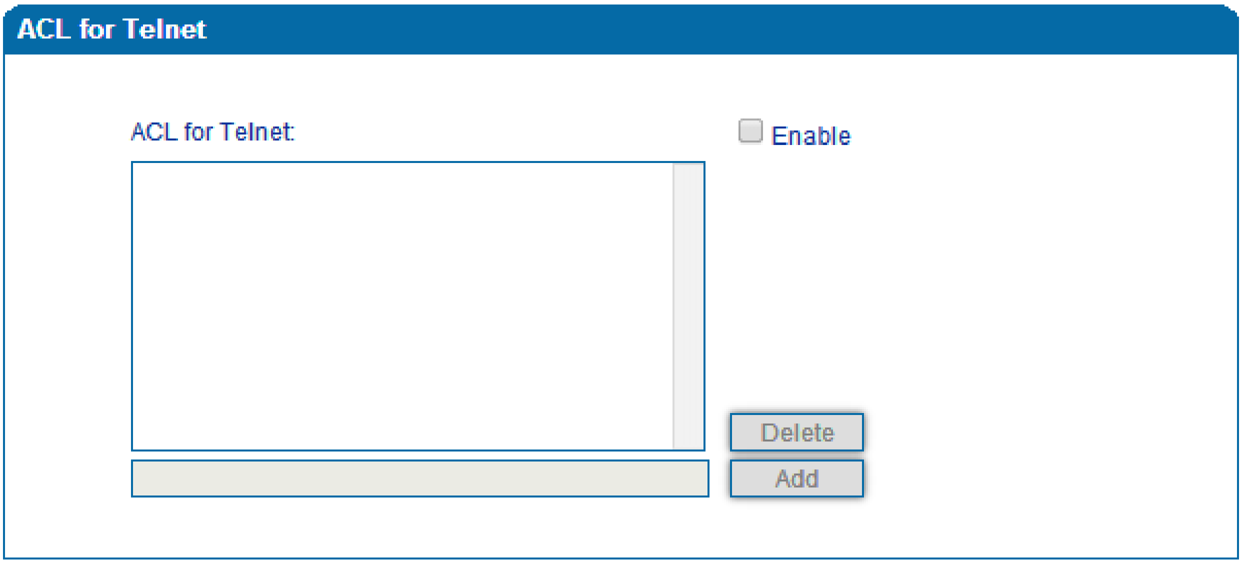


Figure 3.15-2 ACL for Telnet

3.18.3 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’.

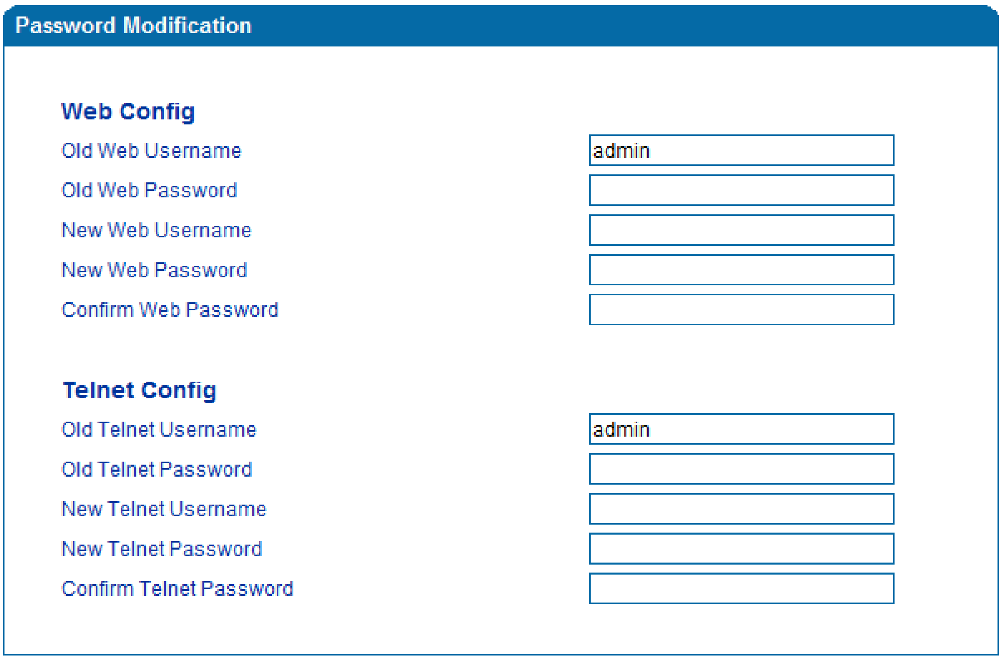
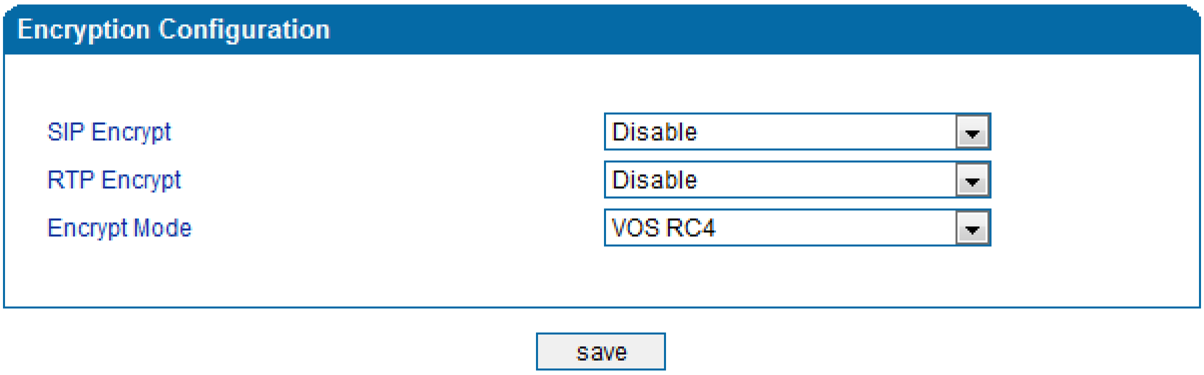


Figure 3.15-3 Password Modification

3.18.4 Encrypt



3.19 Tools

3.19.1 Firmware upload

Firmware upload steps:

Step 1.

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

Current Software Version	IAD-16S 1.18.02.20 PCB 1 LOGIC 0 BIOS 1, 2016-01-29 11:56:23
Backup Software Version	IAD-16S 1.18.02.11 PCB 1 LOGIC 0 BIOS 1, 2014-09-19 15:53:43
DSP Version	C64V_7_8_3
U-BOOT Version	8
Kernel Version	11
FS Version	1.0.13
Hint Language	English

Figure 3.16-1 Firmware Version

Step 2.

Prepare firmware package. The most important is that the package must match with the existing version.

Step 3.

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

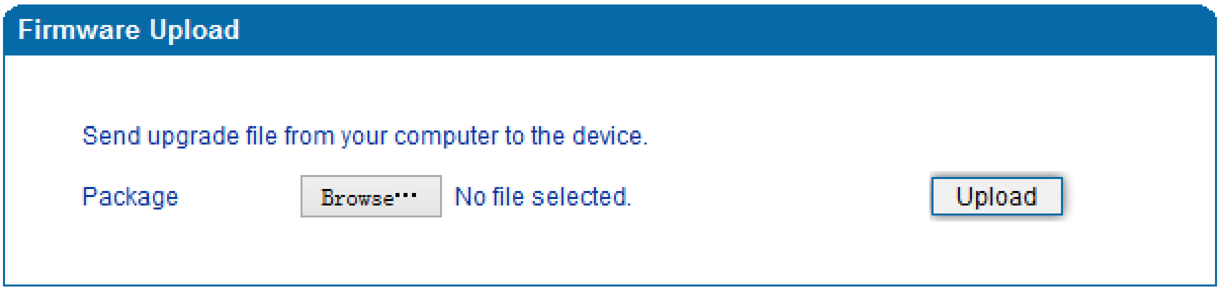


Figure 3.16-2 Firmware Upload

Step 4.

Keep waiting until it prompts 'Software loaded successfully!'

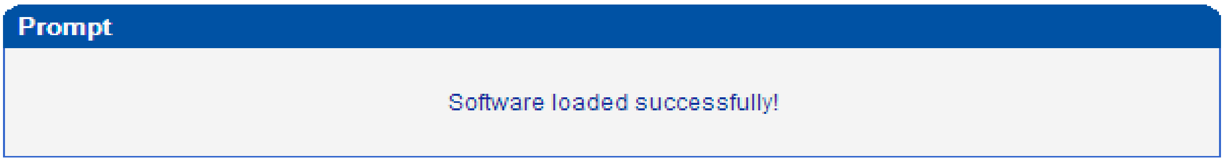


Figure 3.16-3 Successful Firmware Upload

Step 5.

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

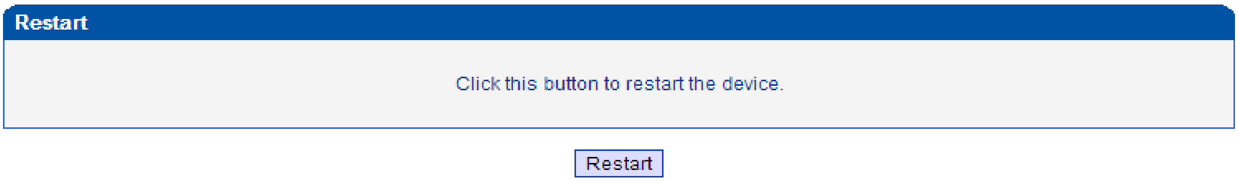


Figure 3.16-4 Restart Gateway

3.19.2 Data Backup

The process data backup:

1) Click "Data Backup"

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

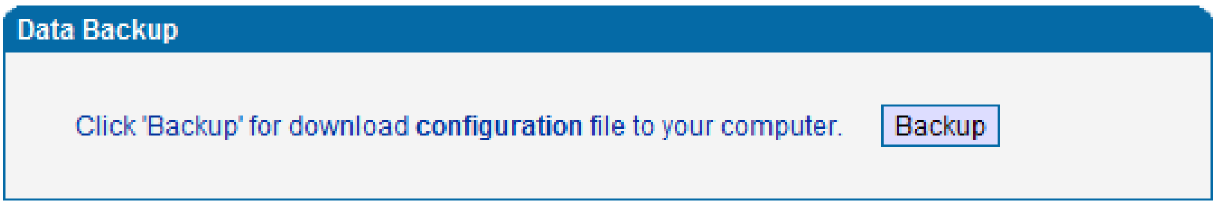


Figure 3.16-5 Data Backup

3.19.3 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.

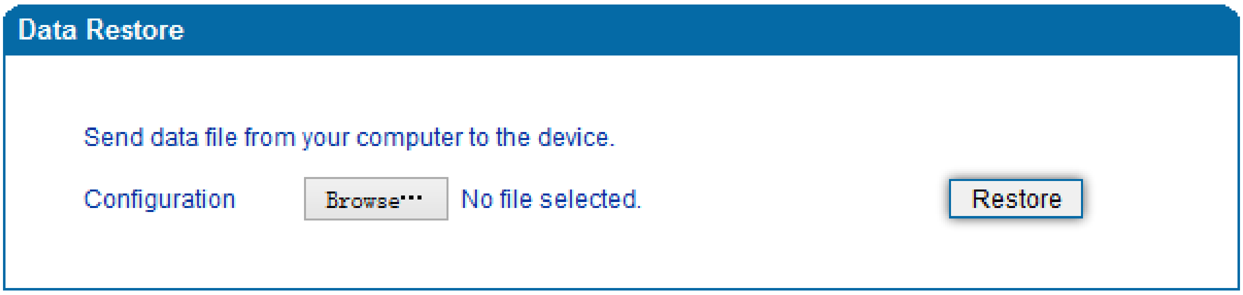


Figure 3.16-6 Data Restore

3.19.4 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.

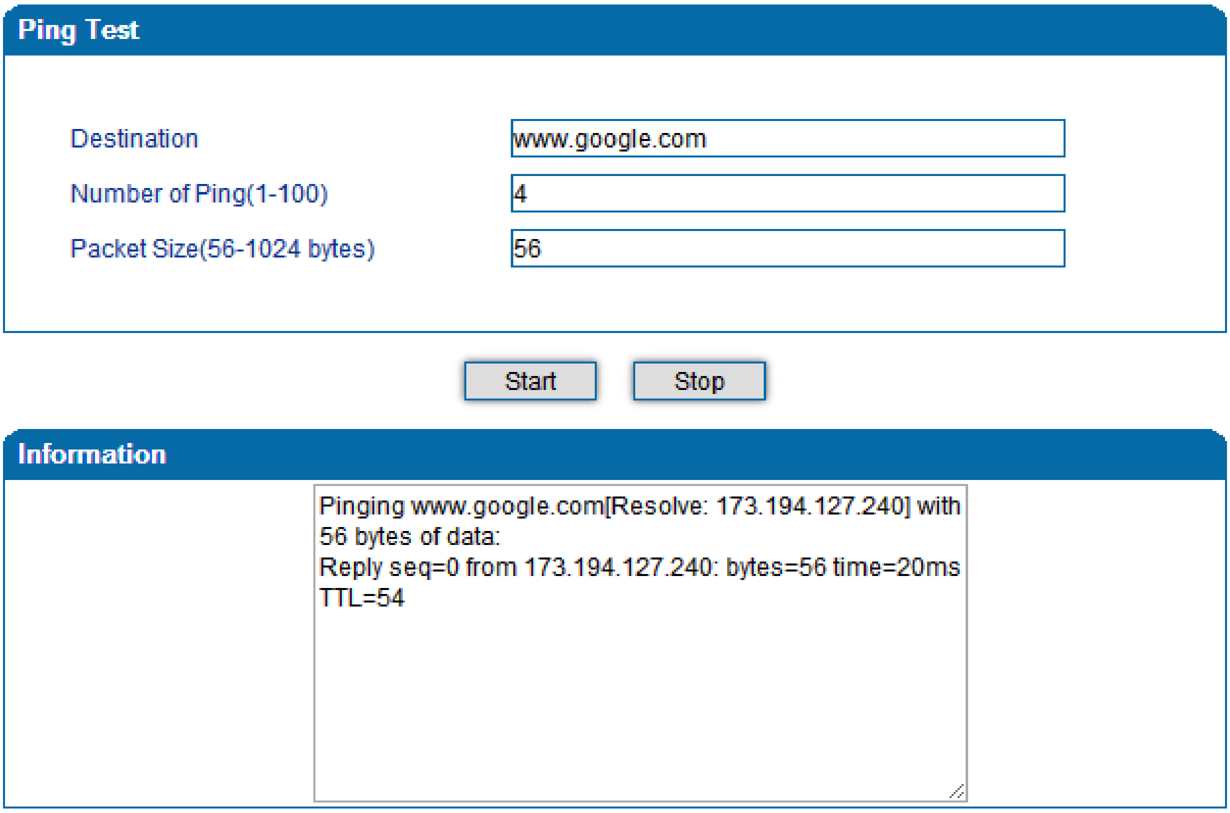


Figure 3.16-7 Ping Test

3.19.5 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**.

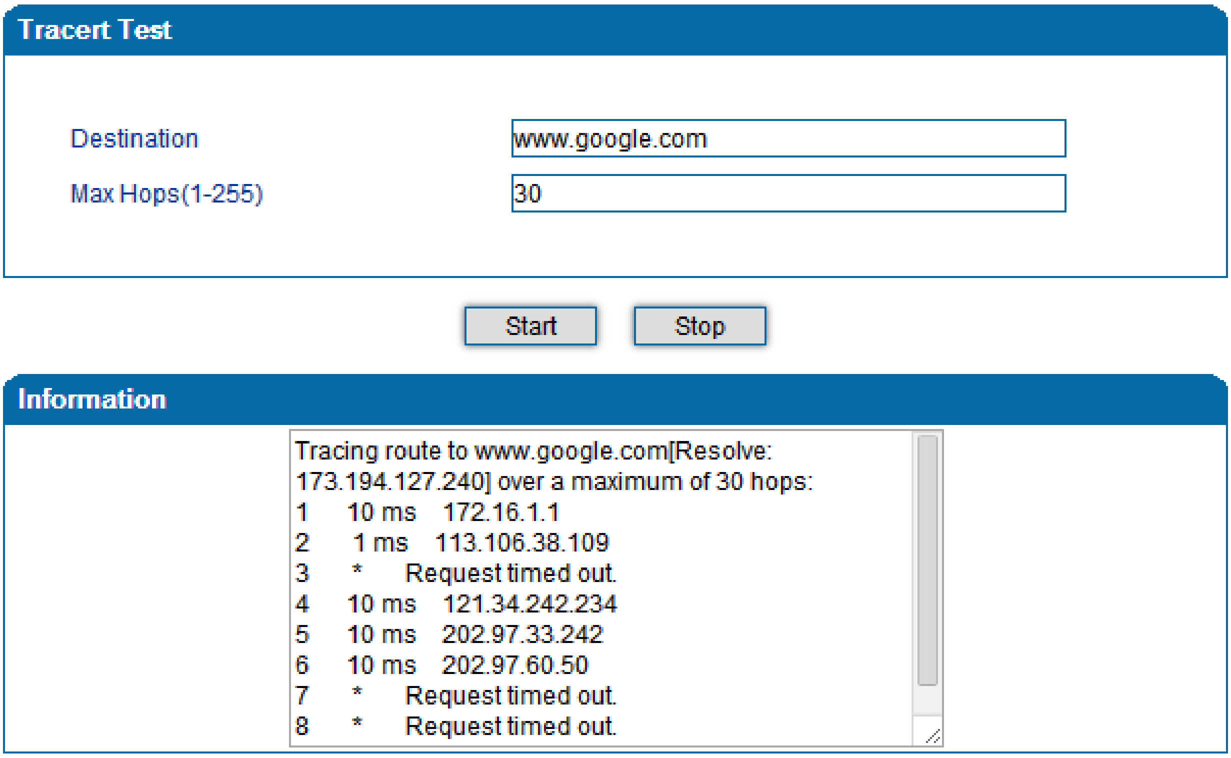


Figure 3.16-8 Tracert Test

3.19.6 Outward Test

Outward test enable user to diagnose the physical phone lines which follow GR909 standards. To start outward test, select the ports to be tested and click 'start'. Testing costs a few minutes.

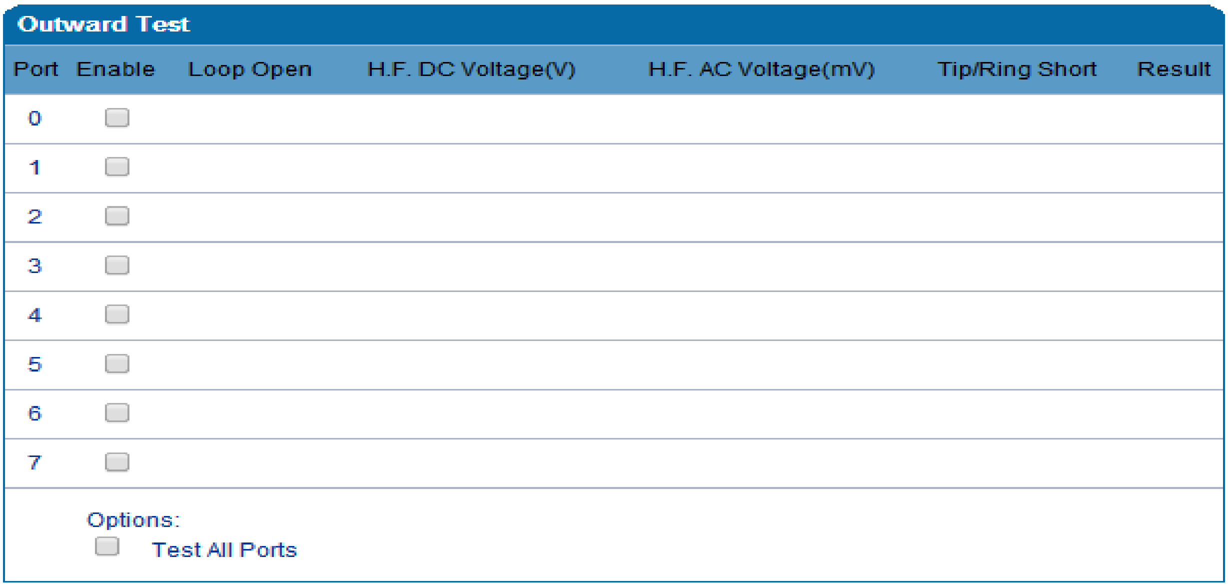


Figure 3.16-9 Outward Test

Test results

OK: the analog phone set and phone line are working well

FAIL: analog phone doesn't connect to FXS port or there's something wrong in phone set

3.19.7 Network Capture

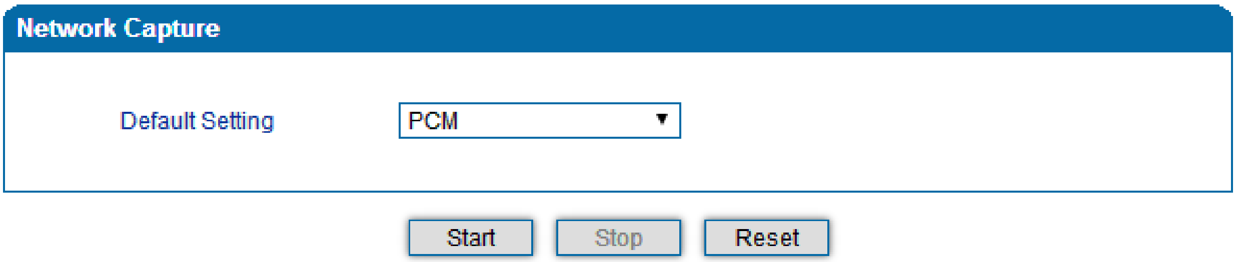
Network capture is a very important diagnostic tool for maintenance. It can be used to capture data packages of the available network ports.

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:

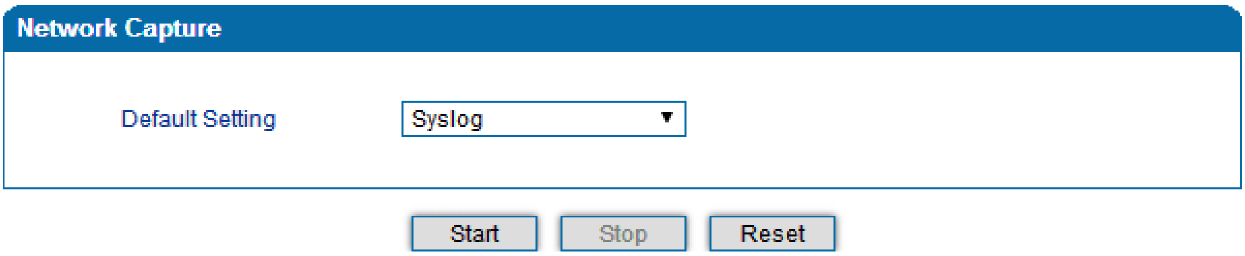
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	-->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

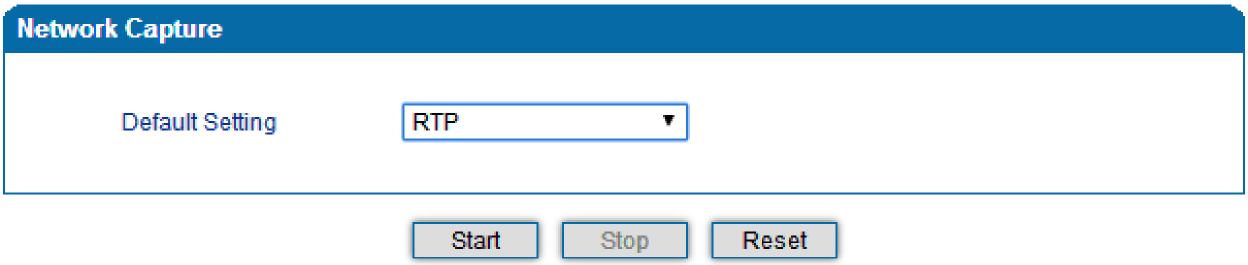
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1>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

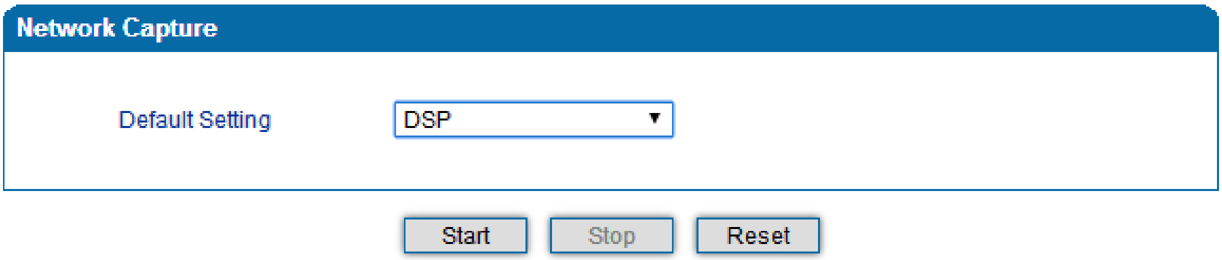
No.	Time	Source	Destination	Protocol	Length	Info
176	7.020000	172.16.221.228	116.204.105.50	SIP	565	Request: REGISTER sip:116.204.105.50 |
178	7.030000	116.204.105.50	172.16.221.228	SIP	411	Status: 200 OK (1 bindings) |
244	11.610000	172.16.221.228	58.56.64.101	SIP/SDP	814	Request: INVITE sip:201@58.56.64.101 |
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

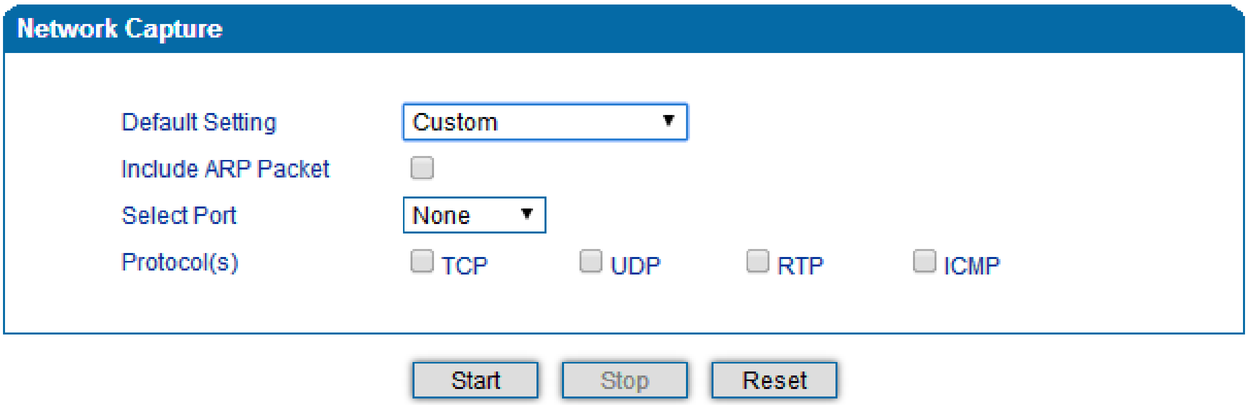
The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of RTP capture as below:

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**

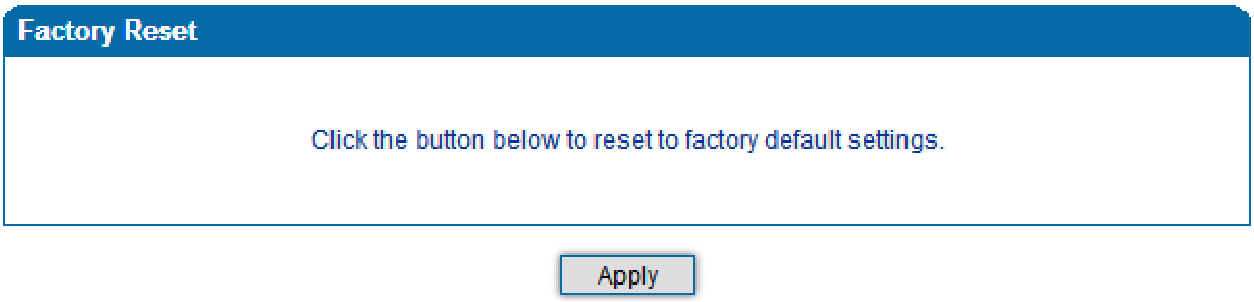
- **Getting start to custom capture**

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



3.19.8 Factory Reset

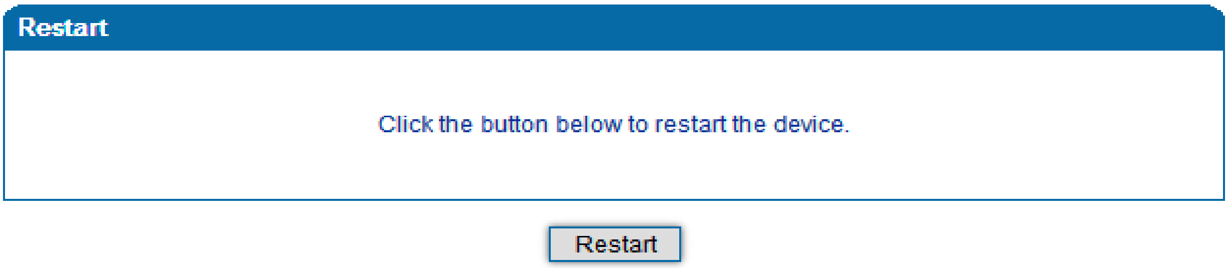
Click 'Apply' to restore the factory settings.



Factory Reset

3.19.9 Device Restart

After saving all the configurations or changes to the equipment, user can restart the DAG2000-16S gateway for the changes to take effect.



Restart Gateway

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
- FXS: Foreign Exchange Station
- FXO: Foreign Exchange Office