# EQ-128S FXS Analog VoIP Gateway

# User Manual V1.0

# Preface

## Welcome

Thanks for choosing the **EQ-128S Analog Gateway for VoIP**! We hope you will make full use of this rich-feature FXS VoIP Gateway.

## About This Manual

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

## Intended Audience

This manual is aimed primarily at the following people:

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

## Revision Record

| Document Name | Document Version | Firmware Version |
|---|---|---|
| EQ-128S FXS Analog VoIP Gateway User Manual V1.0 | V1.0 | |

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# 1 Product Introduction

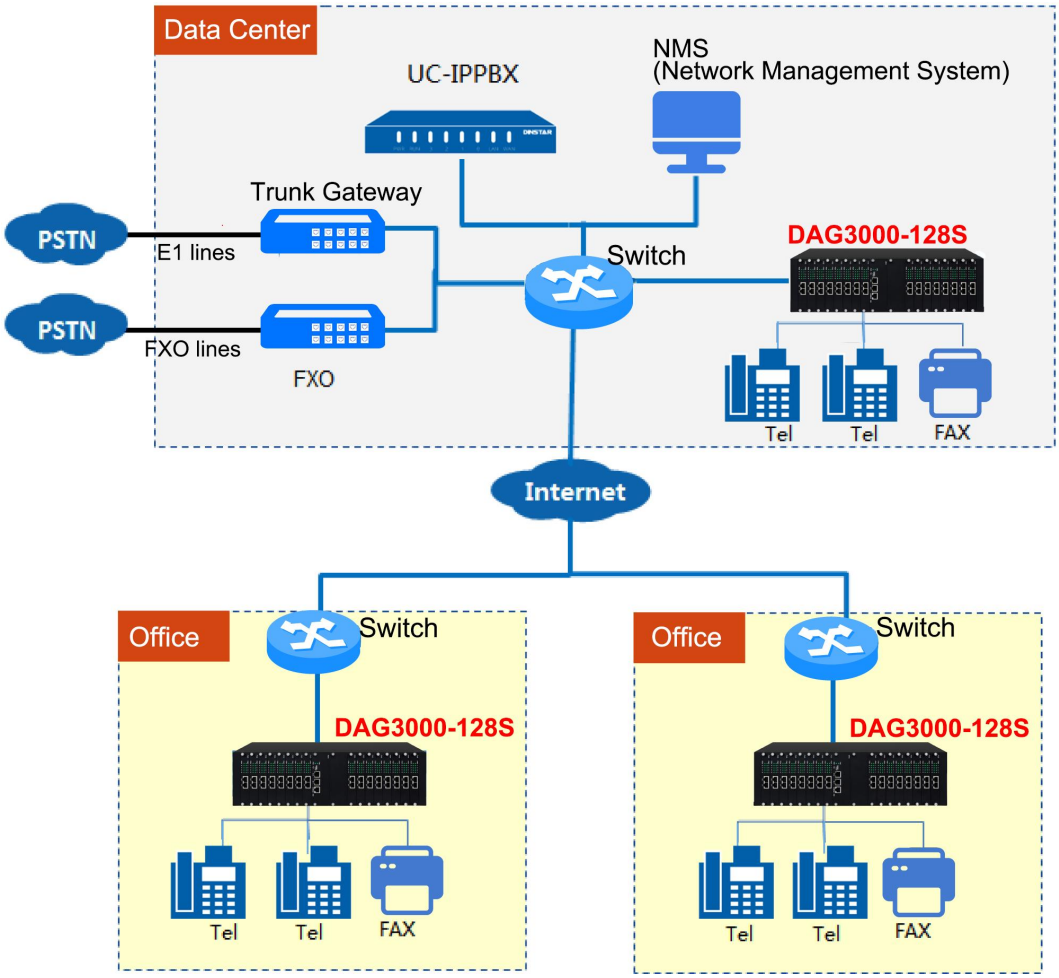
## 1.1 Overview

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

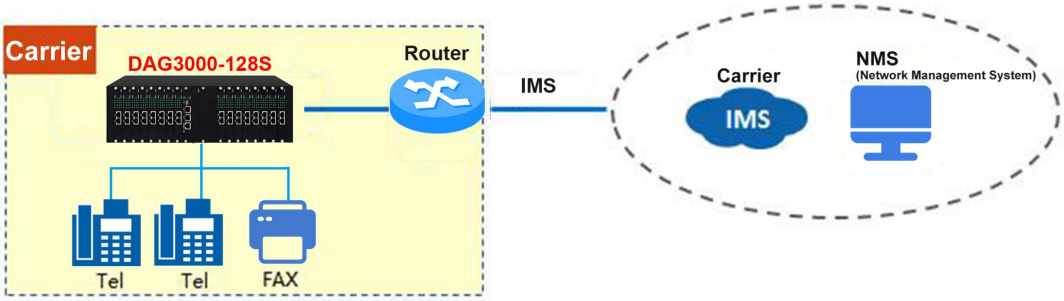
EQ supports the standard SIP protocol and it's compatible with leading IMS/NGN platforms and SIP-based IP telephony systems. It is ideally suited for small and medium businesses, call centers and multi-location environments that need VoIP services.

## 1.2 Application Scenario

The general application scenarios of EQ-128S for enterprises are as follows:




The general application scenarios of EQ-128S for carrier are as follows:



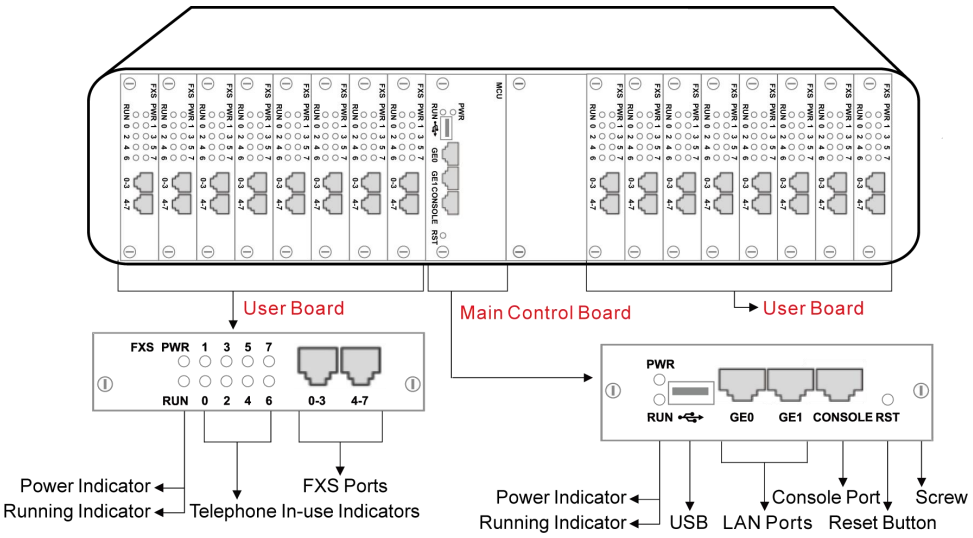

## 1.3 Product Appearance

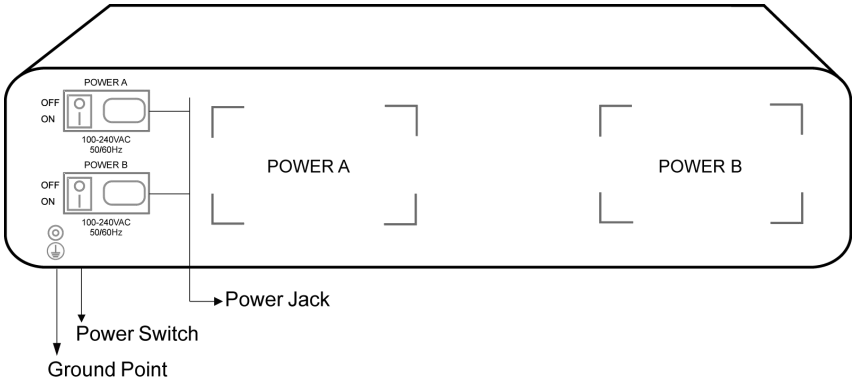
### 1.3.1 Appearance of EQ-128S

Front View:

Back View:

## 1.3.2 Ports and Connector






The description of interfaces of EQ-128S

| Port Name | Connector | Description |
|---|---|---|
| Power Jack | Power Jack | To connect 100-240V AC 50-60HZ power supply |
| LAN Port (GE0/GE1) | RJ45 | To connect to the IP network over a DSL modem or Router or a LAN switch |
| FXS Ports 0~7 | RJ45 | FXS ports to connect standard analog phone or FAX machine or a PBX |
| Console Port | RJ48 | Console port is used to carry out maintenance-related configurations |
| Reset Button | Reset Button | The button is used to restart EQ-128S |

The description of indicators of EQ-128S

| Indicator | Definition | Status | Description |
|---|---|---|---|
| PWR | Power Indicator | On | Power supply is normal . |
| | | Off | There is no power supply or power supply is abnormal. |
| RUN | Running Indicator | Slow Flashing | The device is running properly |
| | | Fast Flashing | SIP account is registered successfully. |
| | | On dull | The device is running improperly. |
| FXS 0~7 | Telephone | On | FXS port is currently occupied by a call |

| | | | |
|---|---|---|---|
| | In-use Indicator | Off | FXS port is idle or faulty |
| LAN Port (GE0/GE1) | Link (Green) | Flashing | The gateway is properly connected to network |
| | | Off | The gateway is not connected to network or network connection is improper |

## 1.4 Features & Functions

### ➢ Key Features

- High density gateway, up to 128 FXS
- Modularized design, easy to expand
- Support IPv4 and IPv6
- 5KM Maximum Cabling Length
- Multiple codecs: G.711A/U,G.723.1,G.729A/B, iLBC
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

### ➢ Physical Interfaces

- Capacity

  Range from 8 to 128 FXS

  Support 16 user board slots
- User Board

  2* RJ45 connectors with 8 FXS
- MCU Board

  1*RS232, 115200bps

  2*10/100/1000Mbps, RJ45

  1* USB 2.0

### ➢ Voice Capabilities & Fax

- G.711A/U law, G.723.1, G.729A/B, G.726,iLBC
- Silence Suppression
- Comfort Noise Generation(CNG)
- Voice Activity Detection(VAD)
- Echo Cancellation(G.168), with up to 128ms
- Adaptive (Dynamic) Jitter Buffer

- Hook Flash
- Programmable Gain Control
- T.38/Pass-through
- Modem/POS
- DTMF mode: Signal/RFC2833/INBAND
- VLAN 802.1P/802.1Q
- Layer3 QoS and DiffServ

## ➢ FXS

- Connector: RJ45 with 4 FXS
- Dial Mode: DTMF and Pulse
- Pulse: 10 and 20 PPS
- Caller ID: DTMF/FSK CLI Presentation
- Max Cable Length: 5KM
- Reversed Polarity
- Programmable Call Progress Tone

## ➢ VoIP

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

## ➢ Software Features

- Hunting Group
- Web ACL
- Telnet ACL
- Action URL
- PPPoE/IPv4/IPv6

- Digitmap
- Bandwidth Optimization
- Routing Rules based Prefixes
- Caller/Called Number Manipulation

## ➢ Supplementary Services

- Call Waiting
- Blind Transfer
- Attend Transfer
- Call Forward on Busy
- Call Forward on No Reply
- Unconditional Call Forward
- Warm/Immediately Hotline
- Call Hold
- Do-not-disturb
- 3-Way Conference
- Message Waiting Indicator

## ➢ Environmental

- 1+1 Power Supply: 100-240VAC, 50-60 Hz
- Power Consumption:105W MAX
- Operating Temperature:0 ℃ ~45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity:10%-90% Non-Condensing
- Dimensions(W/D/H): 440*300*135mm(3U)
- Unit Weight: 9.3kg
- Compliance: CE, FCC

## ➢ Maintenance

- SNMP
- TR069
- Auto Provisioning
- Web/Telnet
- Configuration Backup/Restore
- Firmware Upgrade via Web
- CDR
- Syslog

- Ping/Tracert Test
- Network Capture
- Outward Test(GR909)
- NTP/Daylight Saving Time
- IVR local Maintenance
- Cloud-based Management

# 2 Quick Installation

## 2.1 Installation Attentions

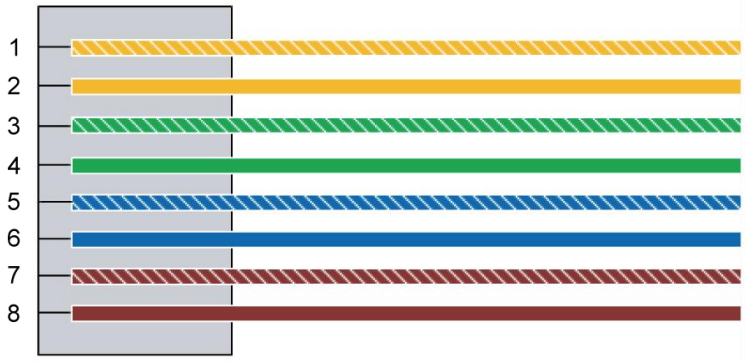
To avoid unexpected accident or device damage, please read the following instructions before installing the EQ device:

- EQ-128S is equipped with RJ45 ports;
- Anti-jamming: to reduce the interference with telephone calls, it's highly recommended that telephone lines connected to the gateway should be placed away from power cables;
- Power supply: the gateway accepts AC input voltage of 100-240V. Please ensure safe and stable power supply;

- Network bandwidth: please ensure there is enough network bandwidth so as to guarantee stabilized running of the gateway;
- Ventilation: to avoid overheating, please do not pile up the gateway with other devices and make sure the gateway has good ventilation around.
- Temperature and humidity: to avoid any accident that might cause malfunction, it's advised to install the gateway in an equipment room where temperature and humidity are appropriate;
- Mechanical load: please make sure the gateway is placed steadily to avoid damage. It' s highly advised to horizontally place the gateway on a flat surface or a cabinet.

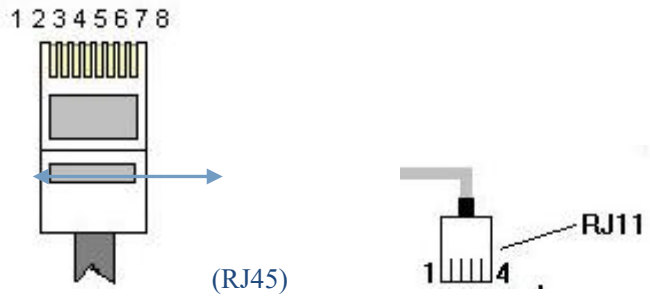
## 2.2 RJ45 Wire Sequence

EQ-128S is equipped with RJ45 interface as FXS port. The internal wire sequence of RJ45 cable is shown as follows:




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

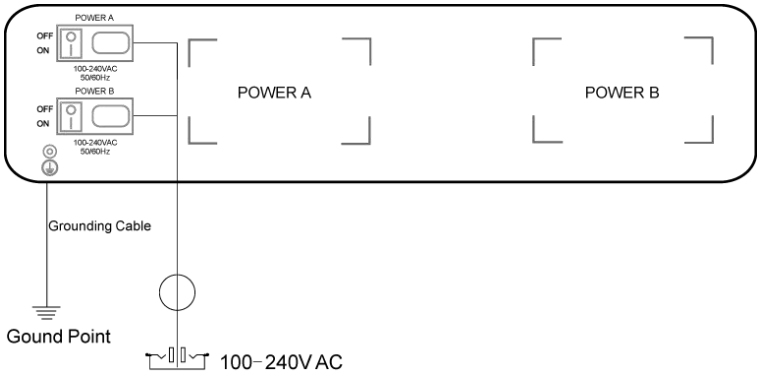
The other end uses the RJ11 interface to connect to the analog phone. Each interface can output 4 FXS voice interfaces, as shown in the figure below,



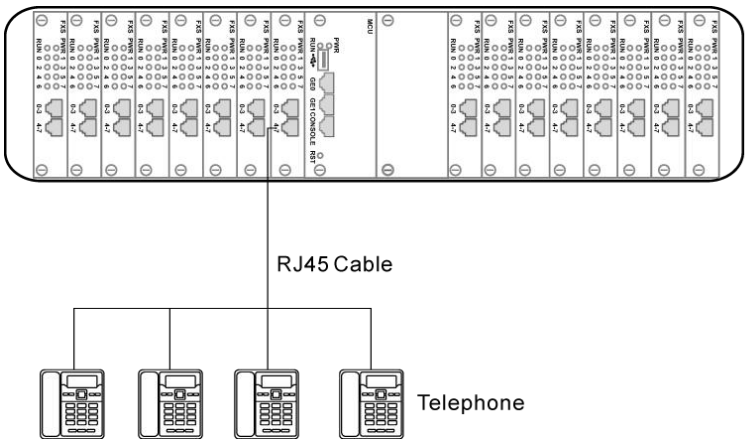

(RJ45)

## 2.3 Installation Steps

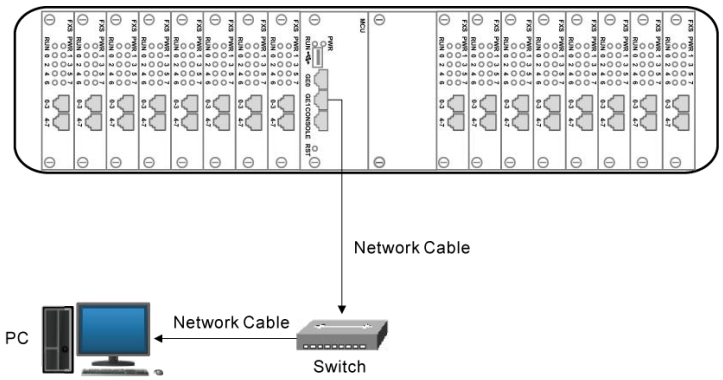
- **Connect the power adapter to the power jack of the EQ device;**




- **Connect telephone line to the FXS port(s);**




- **Connect network cable to the GE0/GE1 port;**



# 3 Basic Operation

## 3.1 Methods to Number Dialing

There are two methods to dial telephone number or extension number:

- Dial the called number and wait for 4 seconds for dialing timeout, or dial the called number directly (the system will judge whether the dialing is completed according to Digitmap and Regular Expression dialplans).
- Dial the called number and press #.

## 3.2 Direct IP Calls

The EQ-128S gateway allows users to call directly through IP address. Under this circumstance, the user only needs an analog phone connected to a FXS port of the gateway, and calls can be established without register.

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

1) Both the EQ-128S and other VoIP device have public IP addresses;
2) The EQ-128S and other VoIP device use private IP addresses of a same LAN;
3) The EQ-128S 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 4 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 EQ-128S since they are using the same IP addresses. Call through IP address is only routed to the default destination port 5060.

## 3.3 Call Holding

If a calling party places a call to a called party which is otherwise engaged, and the called party has the call holding feature enabled, the called party is able to switch to the new incoming call while keeping the current call holding on by pressing the **flash** button/ **hook flash**.

When the called party presses the **flash** button/ **hook flash** once again, he or she will switch back to the first call.

## 3.4 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 called party will hear three beeps if waiting tone is enabled.

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

## 3.5 Call Transfer

### 3.5.1 Blind Transfer

Blind transfer is a call transfer in which the transferring party connects the call to a third party without notifying the third party.

Example: A gives a call to B and B wants to blindly transfer the call to C. Operation instructions are as follows:

1. A dials the extension number of B;
2. The extension of B rings, and B picks up the phone. Then A and B go into conversation;

3. B presses the **FLASH** button (or hook flash), and dial *87* after hearing a dialing tone to trigger blind transfer. Then B dials the extension number of C (end up with #).
4. The extension of C rings, B hangs up the phone and C picks up the phone. Then C and A goes into conversation.

**Note:**

a. On the **Advanced → Feature Code** page, blind transfer should be enabled.
b. If B hears continuous busy tones after he dials the extension number of C, it means the call has timed out.

### 3.5.2 Attended Transfer

Attended transfer is a call transfer in which the transferring party connects the call to a third party after he confirms that the third party agrees to answer the call.

Example: A gives a call to B and B wants to attended transfer the call to C. Operation instructions are as follows:

1. A dials the extension number of B;
2. The extension of B rings, and B picks up the phone. Then A and B go into conversation;
3. B presses the **FLASH** button (or hook flash), and then dials the extension number of C (end up with #).

; Then one of the following situations will happen:

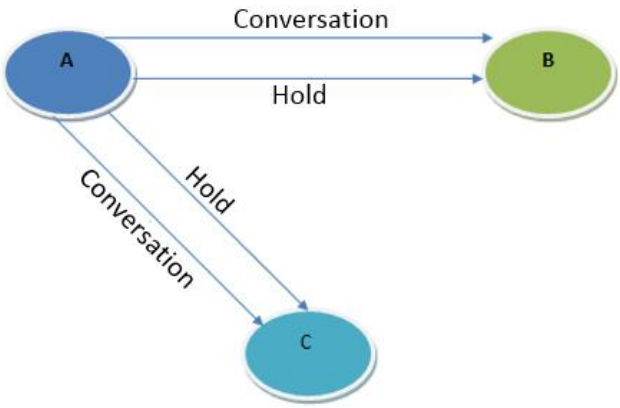
**a.** If C answers the call and accepts the transfer, B will hand up the phone, and then C and A go into conversation.

**b.** If the extension of C cannot be reached or if C rejects the call, B needs to press the **FLASH** button to resume the call with A.

## 3.6 Three-way Calling

Three-way calling:

1) A calls B,B picks up the phone, then A and B goes into conversation;
2) 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.
3) 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.




## 3.7 Description of Feature Code

EQ-128S provides convenient telephone functions. Connect a telephone to the FXS port and dial a specific feature code, and you can query corresponding information.

| Code | Corresponding Function |
|---|---|
| ***158#** | Dial *158# to query LAN IP |
| ***114#** | Dial *114# to query the phone number of a FXS port |
| ***115#** | Dial *115# to query the phone number of a FXS port group |
| ***168#** | Dial *168# to query the register status of a FXS port |
| ***157*** | Dial *157*0 to set route mode; dial *157*1 to set bride mode |
| ***150*** | Dial *150*1 to set IP address as static IP address; dial *150*2 to set IP address as DHCP IP address |
| ***152*** | Dial *152* to set IPv4 address, for example:<br>Dial *152*192*168*1*10# to set IPv4 address as 192.168.1.10 |
| ***156*** | Dial *156* to set IPv4 gateway, for example:<br>Dial *156*192*168*1*1# to set IPv4 gateway as 192.168.1.1 |
| ***153*** | Dial *153* to set IPv4 netmask, for example:<br>Dial *153*255*255*0*0*# to set IPv4 netmask as 255.255.0.0 |
| ***170#** | Dial *170# to increase the sound volume of a FXS port |
| ***171#** | Dial *171# to decrease the sound volume of a FXS port |
| ***160*** | Dial *160*1# to allow HTTP WAN access, Dial *160*0# to deny HTTP WAN access |
| ***165*** | Dial *165*000000# to restore username/password and network configuration to factory defaults |

| | |
|---|---|
| ***111#** | Dial *111# to restart the device |
| ***47*** | Dial *47* to allow call through IP address, for example:<br>Dial *47*192*168*1*1# to allow to call through the IP address of 192.168.1.1 |
| ***51#** | Dial *51# to enable the call waiting service |
| ***50#** | Dial *50# to disable the call waiting service |
| ***87*** | Dial *87* to trigger blind transfer, for example:<br>Dial *87*8000#, and you can blind transfer to the extension number 8000 |
| ***72*** | Enable unconditional call forwarding service. Example: Dial *72*8000, and calls will be unconditionally forwarded to extension number 8000 |
| ***73#** | Disable unconditional call forwarding service |
| ***90*** | Enable the 'call forwarding on busy' service. Example: Dial *90*8000, and calls will be forwarded to extension number 8000 when the called number is busy |
| ***91#** | Disable the 'call forwarding on busy' service |
| ***92*** | Enable the 'call forwarding on no reply' service. Example: Dial *92*8000, and calls will be forwarded to extension number 8000 when there is no reply from the called number |
| ***93#** | Disable the 'call forwarding on no reply' service |
| ***78#** | Enable the 'No Disturbing' service |
| ***79#** | Disable the 'No Disturbing' service |
| ***200#** | Dial *200# to access voicemail |

**Note：** A voice prompt indicating successful configuration will be played after each configuration procedure. Please do not hang up the phone until hearing this voice prompt.

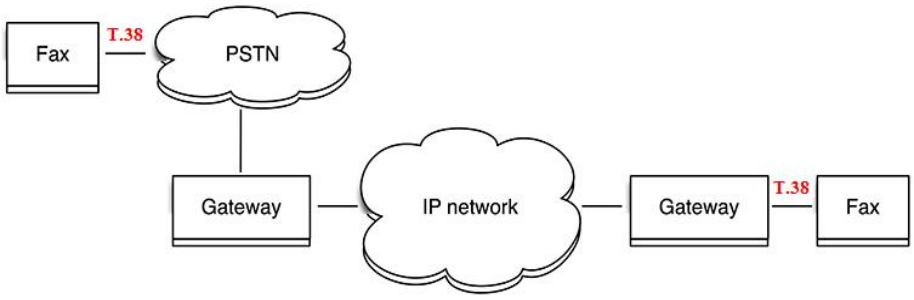
# 3.8 Send or Receive Fax

## 3.8.1 Fax Mode Supported

- T.38（IP-based）
- T.30（Pass-Through）
- Modem
- Adaptive Fax Mode (automatically match with the peer fax mode)

## 3.8.2 Explanation of T.38 and Pass-through

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




**T 3.0 (Pass-through)**: Under the pass-through mode, fax signal is not converted and fax traffic is carried in RTP packets. It uses the G.711 A or G711U codec in order to reduce the damage to fax signal.

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

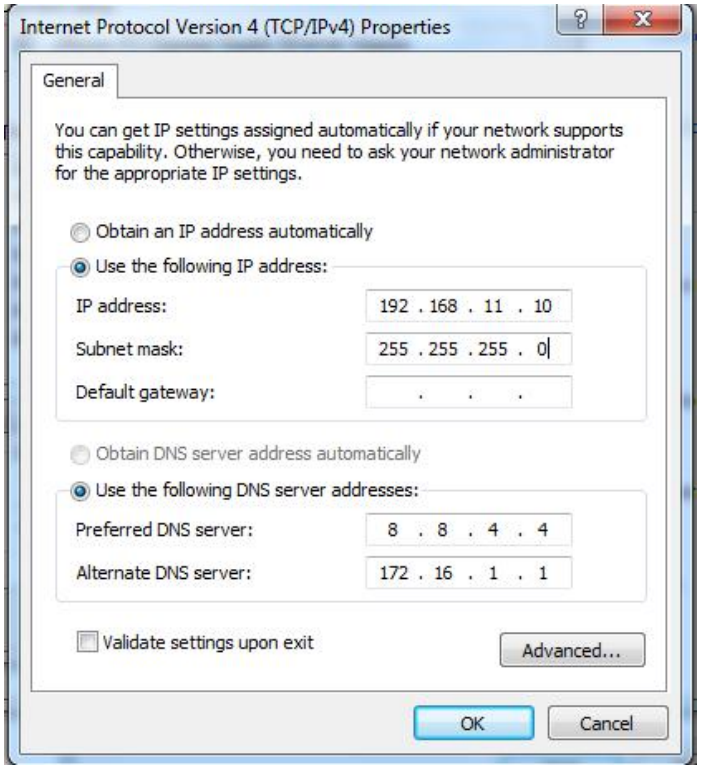
# 4 Configurations on Web Interface

## 4.1 Preparations for Login

### 4.1.1 Network Connection

Firstly, connect the device to network according to the above network diagrams, and connect a telephone to the FXS port. Then dial *158# to query the IP address of the gateway.

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



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

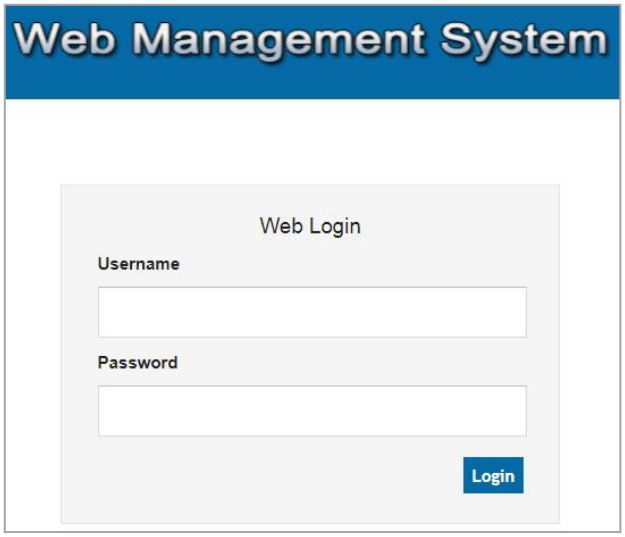
### 4.1.2 Log In Web Interface

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

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

It is suggested that you should modify the username and password for security consideration.

Figure 4- 1 Login GUI

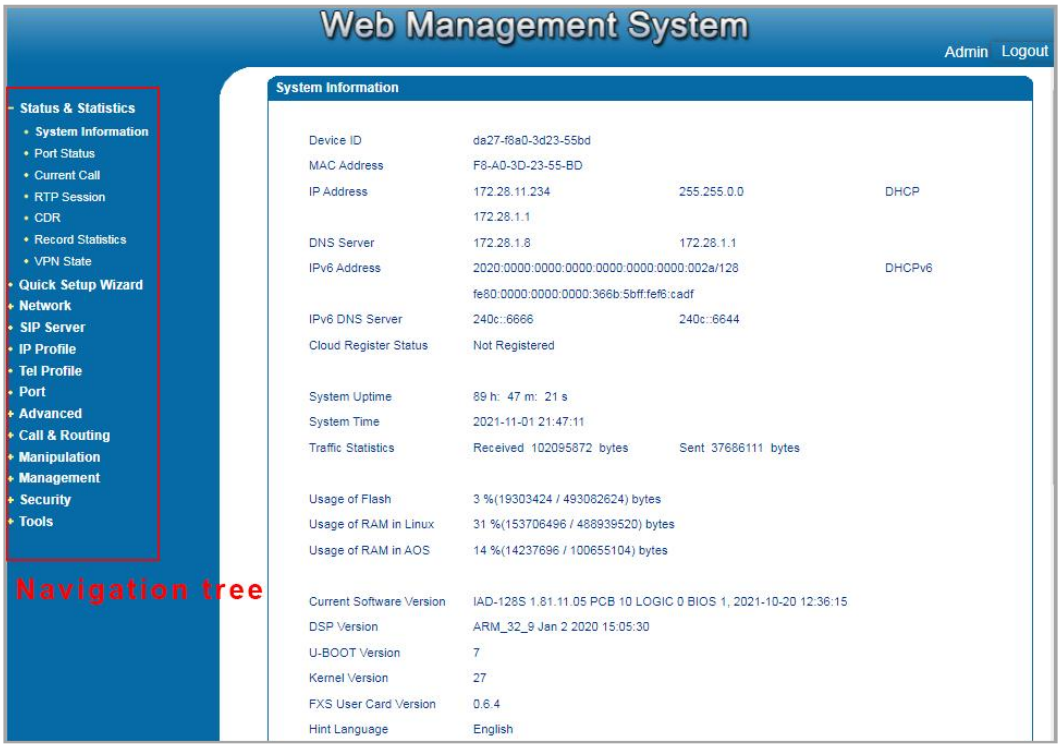



## 4.2 Navigation Tree

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

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

Figure 4- 2 Navigation Tree of Web Interface




## 4.3 Status & Statistics

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

### 4.3.1 System Information

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

Figure 4- 3-1 System Information

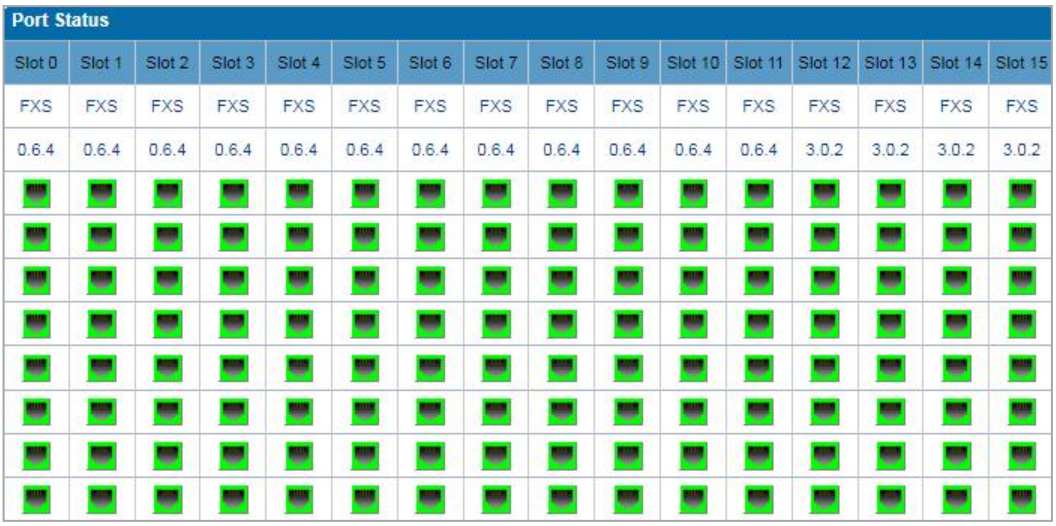



Table 4.3.1 Explanation of Items on System Information Interface:

| | |
|---|---|
| **Device ID** | A unique ID of each device. This ID is used for warranty and cloud server authentication. |
| **MAC address** | Hardware address of the WAN port |

| | |
|---|---|
| **IP Address** | There are three kinds of IP address for the WAN port and LAN port:<br><br>**DHCP: Obtain IP address automatically**. EQ 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 EQ from a defined range of numbers.<br><br>**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.<br><br>If you choose static IP address, you need to fill in the following information:<br><br>● IP Address: the IP address of the WAN port of the EQ;<br>● Subnet Mask: the netmask of the router connected the EQ;<br>● Default Gateway: the IP address of the router connected the EQ;<br><br>**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.<br><br>If you choose PPPoE, you need to fill in to fill in the following information:<br><br>● Username: the account name of PPPoE<br>● Password: the password of PPPoE<br>● Server Name: the name of the server where PPPoE is placed |
| **DNS Server** | IP addresses of primary DNS server and standby DNS server are displayed. |
| **Cloud Register Status** | Whether the EQ device is registered to cloud or not. |
| **System Uptime** | The running time of the EQ device since it is powered on. |
| **NTP Status** | Succeed: the EQ device is sync to NTP server successfully;<br><br>Failed: the EQ device fails to be sync to NTP server. Then you should check network connection and the NTP server. |
| **WAN Traffic Statistics** | Total bytes of message received and sent by WAN port. |
| **Usage of Flash** | Detailed usage of Flash memory |
| **Usage of RAM in Linux** | detailed RAM usage of Linux core |
| **Usage of RAM in** | Detailed RAM usage of AOS |

| | |
|---|---|
| **AOS** | |
| **Current Software Version** | The software version that runs on the EQ device. Model name, version number and the software development date are displayed. |
| **Backup Software Version** | Backup software is for the purpose of backup. When the current software fails, the backup software version will work. |
| **U-boot Version** | U-boot version |
| **Kennel version** | Linux Kennel version |
| **FS Version** | File system version |
| **Hint Language** | The current language of the EQ device |

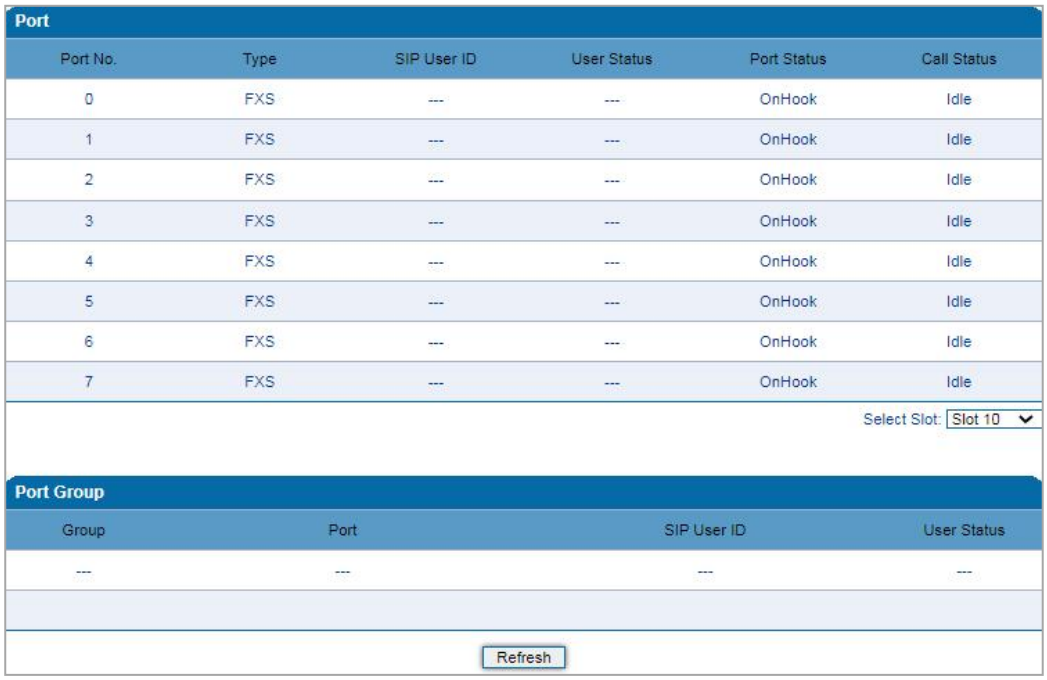
### 4.3.2 Port Status

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

The description of ports and port groups is as follows:

Port: An FXS port

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



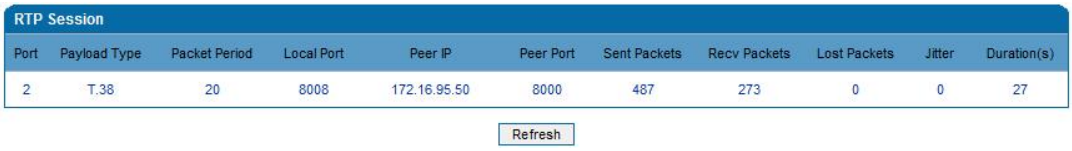
### 4.3.3 Current Call

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

### 4.3.4 RTP Session

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

Figure 4- 3 Real-time RTP Session Information

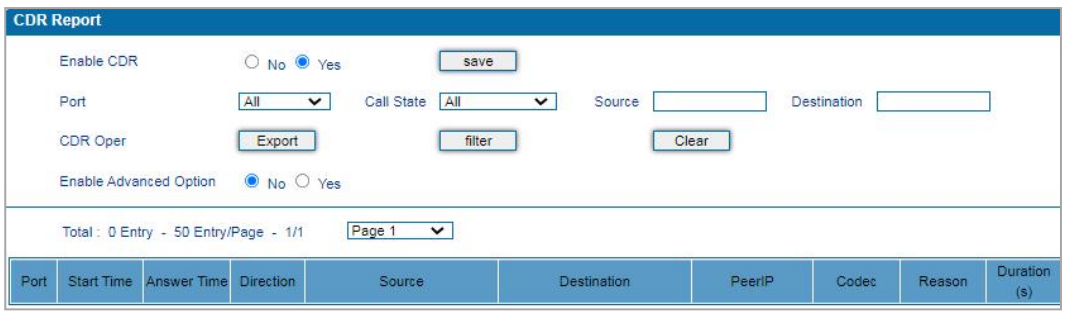



### 4.3.5 CDR

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

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

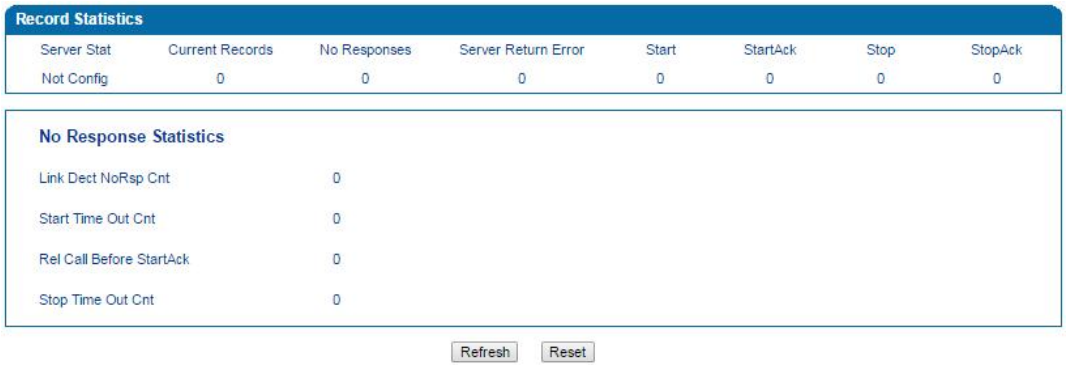
Figure 4- 4 CDRs of FXS Ports



### 4.3.6 Record Statistics

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

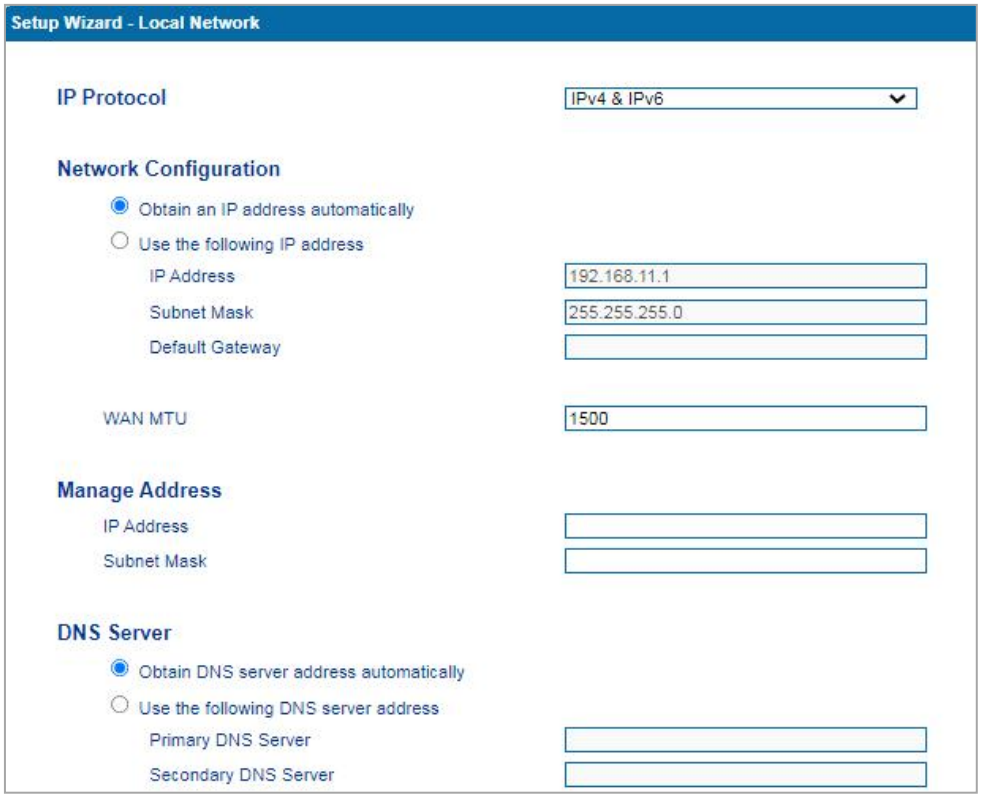
Figure 4- 5 Record Statistics




## 4.4 Quick Setup Wizard

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

For the configurations of network, SIP server and SIP port, please refer to 4.5 , 4.6 and 4.9 .




# 4.5 Network

## 4.5.1 Local Network

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

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

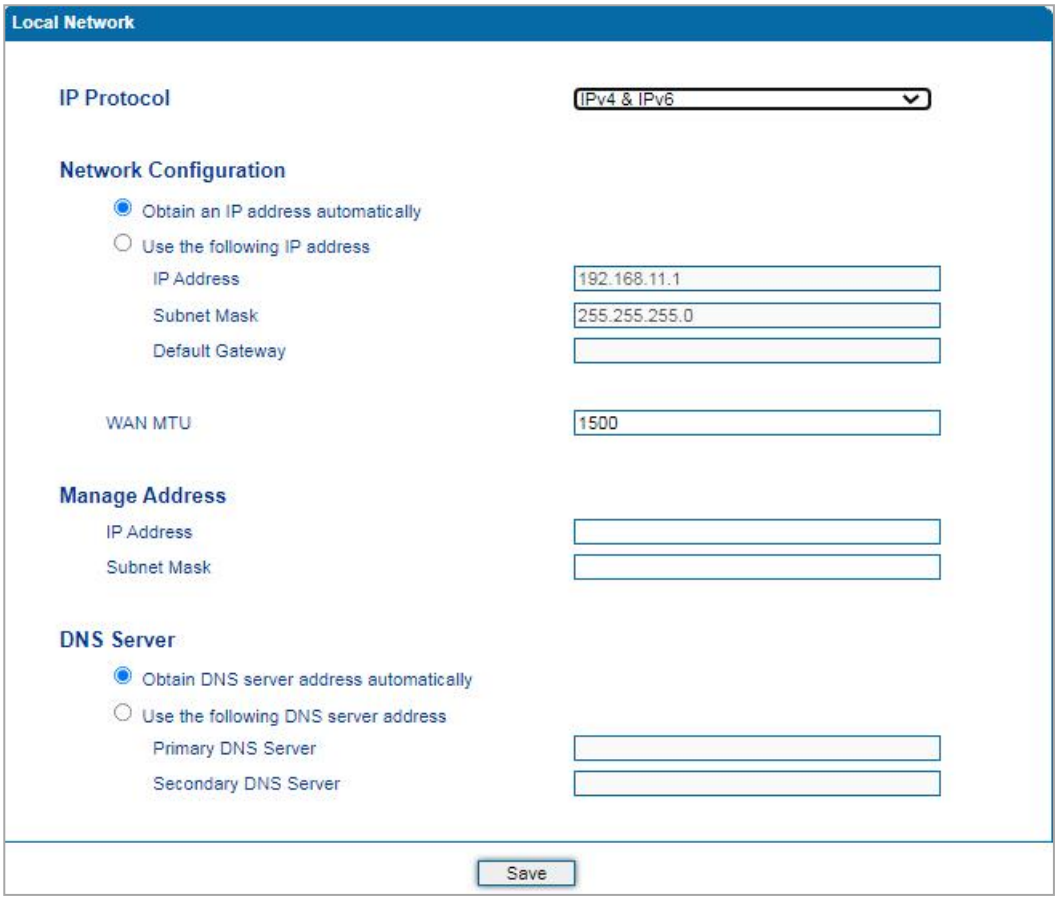
**Note:**

a. When it is configured to "obtain an IP address automatically", it is necessary to ensure that there is a DHCP Server in the network and it is working normally.

b. The management IP address and the network IP address cannot be in the same network segment.

c. After the configuration is complete, you need to restart the device to make the configuration take effect.

Figure 4- 6 Network Setting




## 4.5.2 VLAN (Virtual Local Area Network)

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

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

Figure 4- 7 Configure VLAN

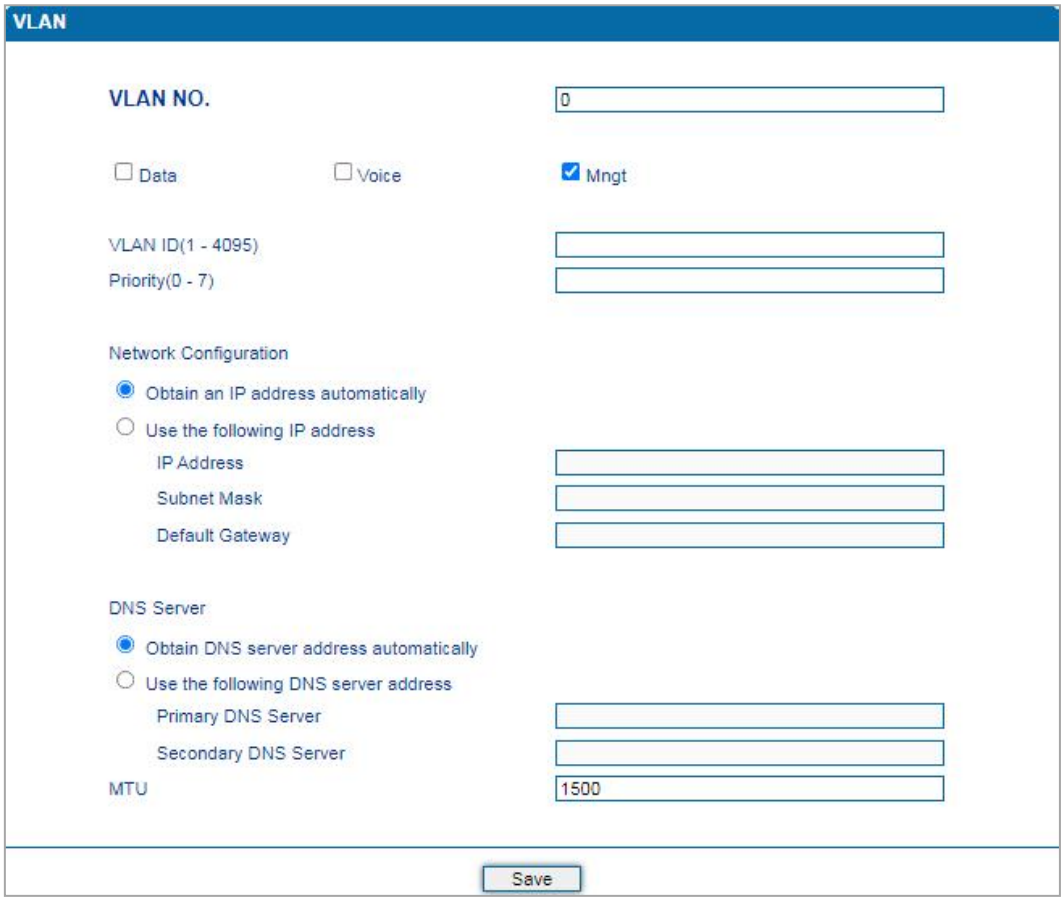



Table 4- 1 Explanation of VLAN Parameters

| | |
|---|---|
| **Data/Voice/Management** | Select what kind of messages are allowed to go through this VLAN. For example, if the checkbox on the left of data is selected, it means data messages are subject to the following network setting of this VLAN. |
| **VLAN ID(0-4095)** | Set an ID to identify a VLAN based on 802.1Q protocol. Range is from 0 to 4095. |
| **Priority（0-7）** | Set the priority of a VLAN based on 802.1P protocol. 0 is the highest priority. |
| **Network Setting** | Set a DHCP IP address or static IP address for a VLAN, and set the IP address of the DNS server used by the VLAN. |

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

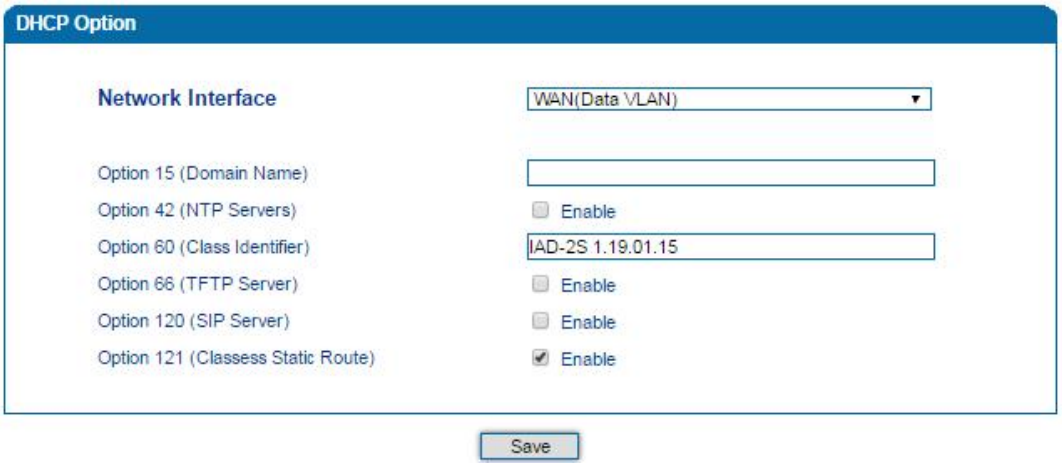
### 4.5.3 DHCP Option

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

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

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

Figure 4- 8 Configure DHCP Option

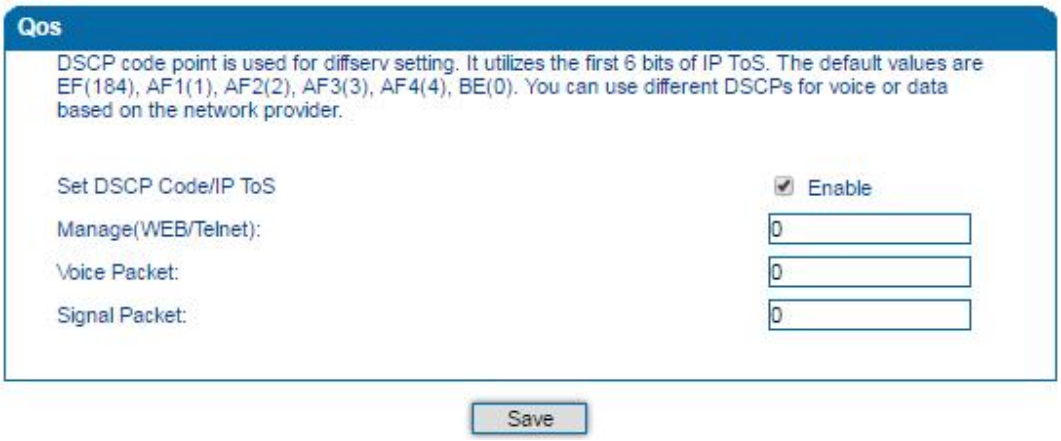



Network Interface: choose which VLAN to send request to DHCP server (or to receive information from DHCP server).

### 4.5.4 QoS

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

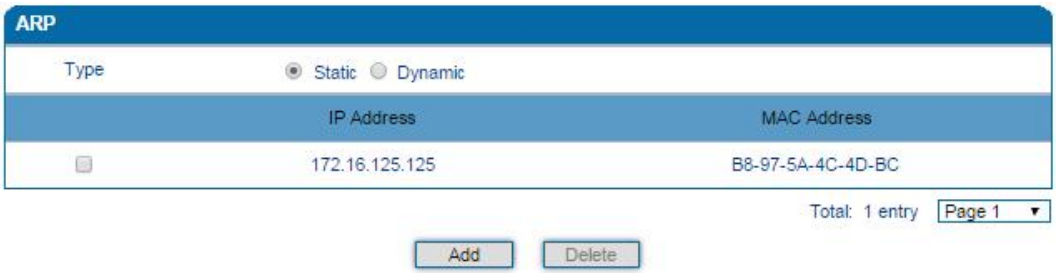
Figure 4- 9 Qos




## 4.5.5 ARP

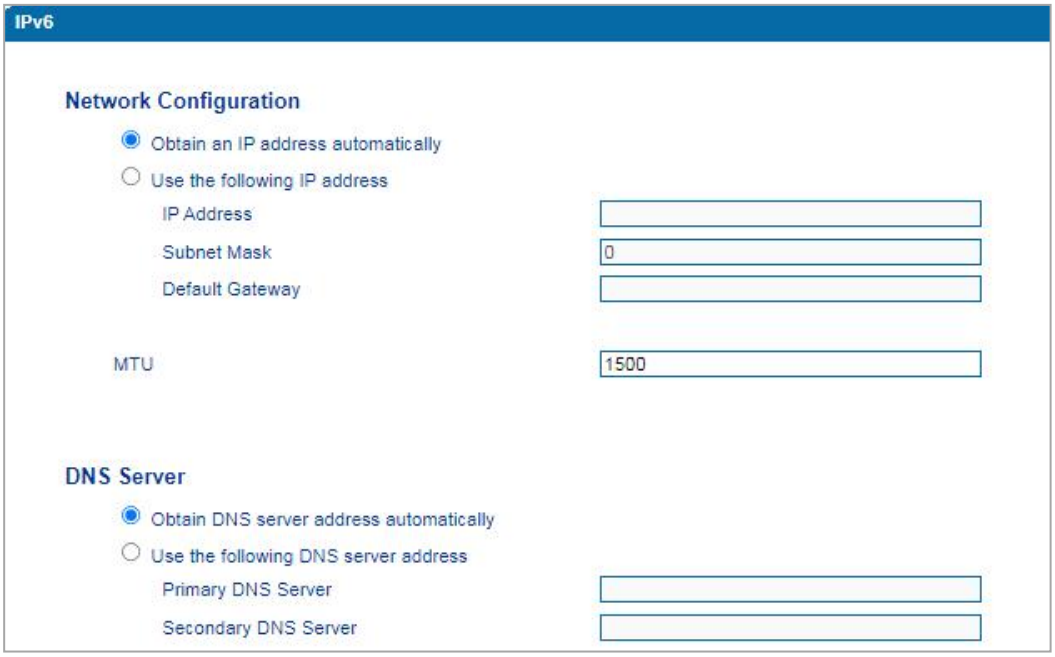
ARP is address resolution protocol, which helps to get the MAC address of a device through its IP address. Under TCP/IP network environment, each host is assigned with a 32-bit IP address, but MAC address needs to be known for message transmission in the physical network. In the above case, ARP can help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

Table 4- 2 Explanation of Parameters for ARP




## 4.5.6 IPv6 Network

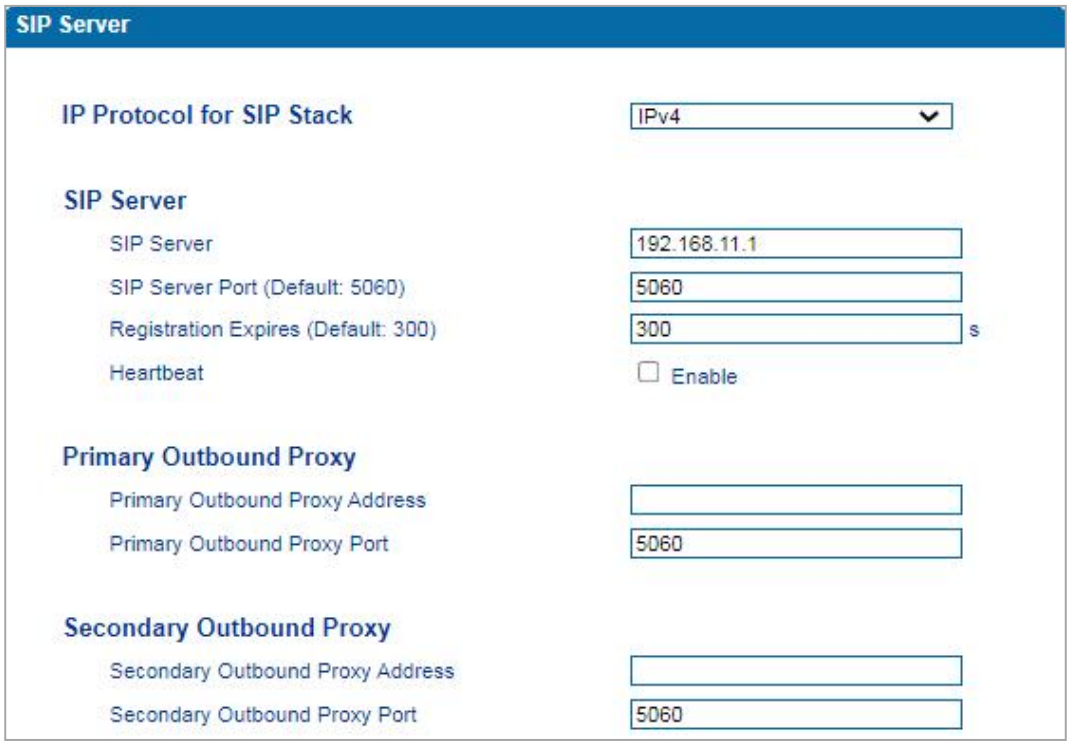
When IPv4&IPv6 is selected as the IP protocol in the local network, information such as IPv6 address, subnet mask, default gateway, and primary/secondary DNS servers can be configured. The IPv6 network configuration interface is shown in Figure 4.4-6:




## 4.6 SIP Server

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

Figure 4- 10 Configure SIP Server Information



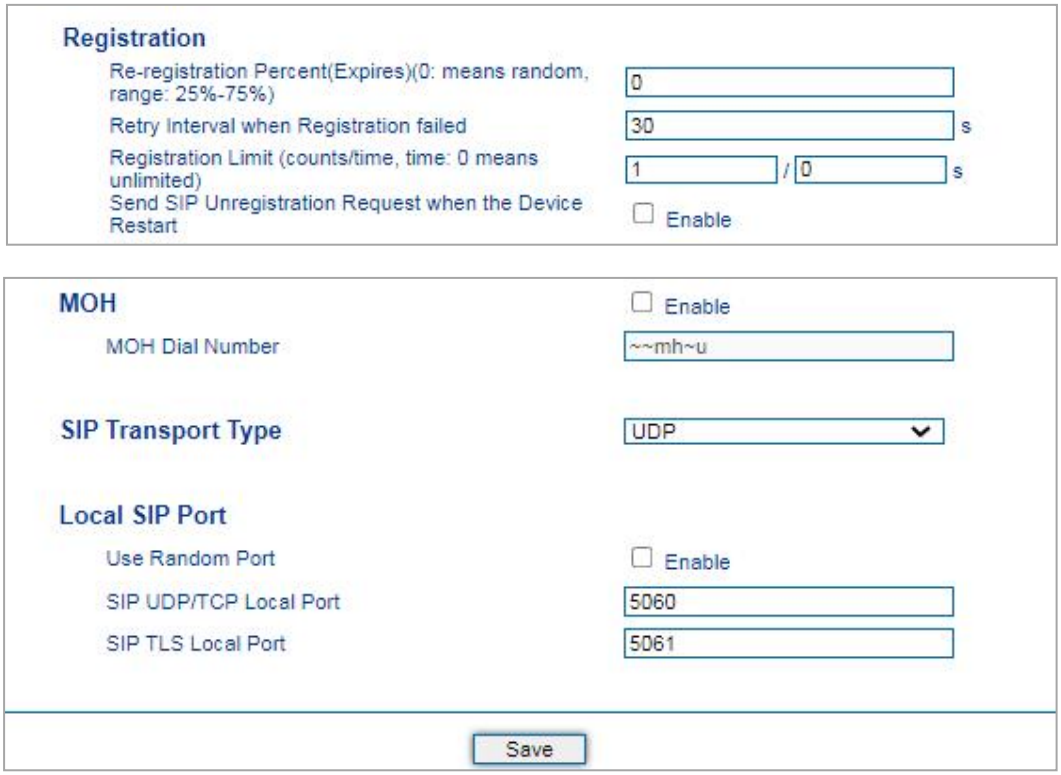



Table 4- 3 Explanation of Parameters for SIP Server

| | |
|---|---|
| **IP Protocol for SIP Stack** | Choose SIP protocol stack, support IPv4 and IPv6, please choose the corresponding protocol according to the actual connected server |
| **SIP Server Address** | The IP address or domain name of the SIP server. It is provided by VoIP service provider. |
| **SIP Server port** | The service port of the SIP server. It is 5060 by default. |
| **Registration Expires** | It is used to avoid excessively frequent registrations.<br>When the time that is set expires, the EQ device will send register request to the SIP server. The time is 300s by default. |
| **Heartbeat** | Heartbeat is used to check the connection between the EQ device and SIP server. |
| **Outbound Proxy Address** | The IP address or domain name of outbound proxy server, which is provided by VoIP service provider. |
| **Outbound Proxy Port** | Service port of outbound proxy server. It is 5060 by default. |
| **Retry Interval when Registration failed** | The retry interval after a registration fails. Default: 30s |
| **Registration times per 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, TLS or Automatic. Default: UDP. |
| **Use Random Port** | If this parameter is selected, the local port of the EQ device for using SIP services is chosen by random. |
| **SIP UDP/TCP Local Port** | The UDP/TCP port of EQ device for using SIP services. Default SIP UDP/TCP local is 5060. |
| **SIP TLS Local Port** | The TLS port of EQ device for using SIP services. Default SIP TLS local port is 5061. |

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

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

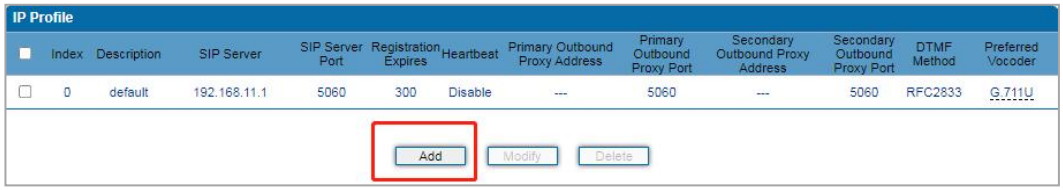
SIP server based on windows platform, such as :mini SipServer、Brekeke，VoIPswitch etc.

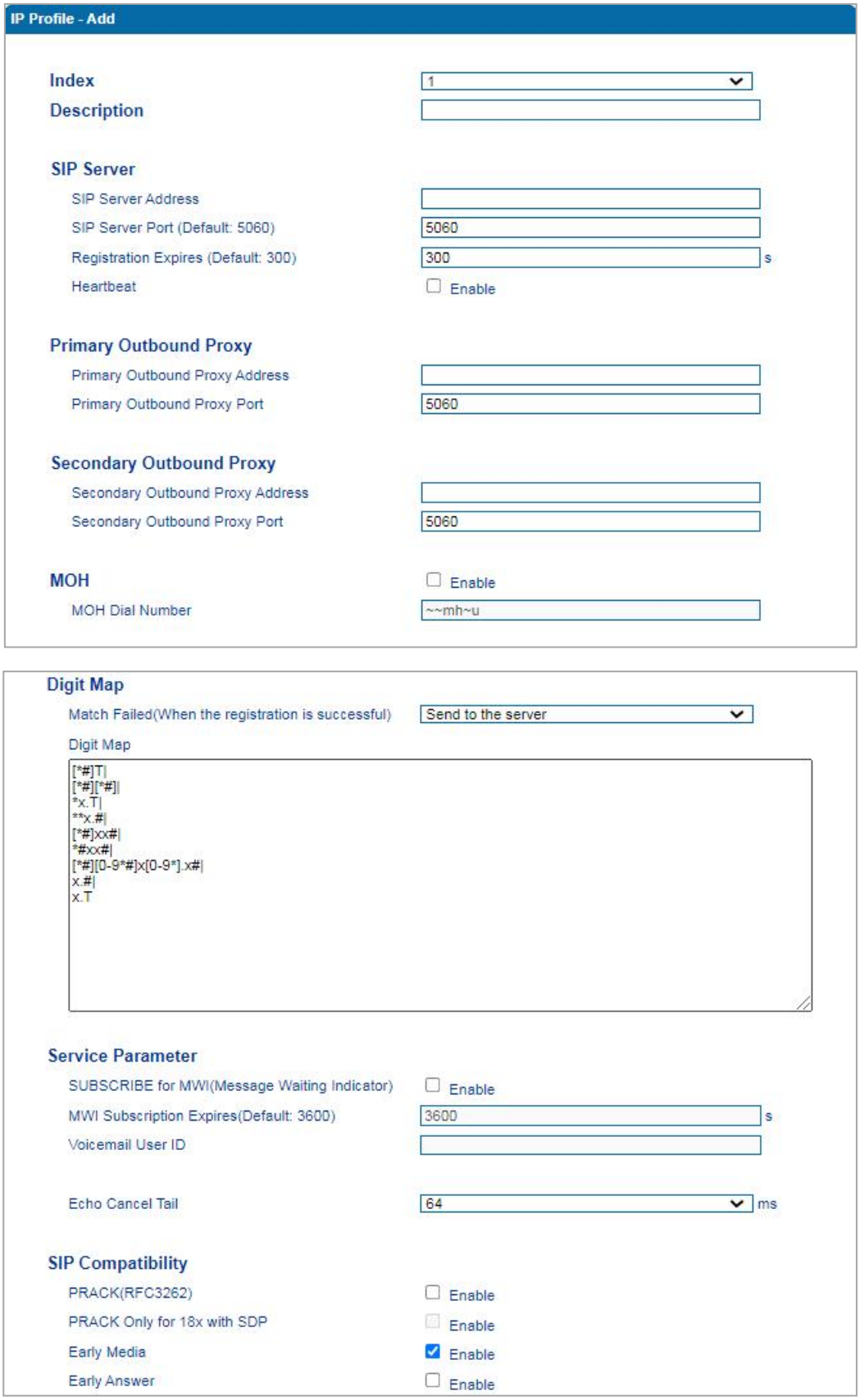
Carrier-grade soft switch platform, such as Cisco, Huawei, ZTE etc.

## 4.7 IP Profile

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

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





IP Profile - Add

Index: 1

Description:

SIP Server

SIP Server Address:

SIP Server Port (Default: 5060): 5060

Registration Expires (Default: 300): 300 s

Heartbeat: Enable

Primary Outbound Proxy

Primary Outbound Proxy Address:

Primary Outbound Proxy Port: 5060

Secondary Outbound Proxy

Secondary Outbound Proxy Address:

Secondary Outbound Proxy Port: 5060

MOH: Enable

MOH Dial Number: ~~mh~u

Digit Map

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

Digit Map:

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

Service Parameter

SUBSCRIBE for MWI(Message Waiting Indicator): Enable

MWI Subscription Expires(Default: 3600): 3600 s

Voicemail User ID:

Echo Cancel Tail: 64 ms

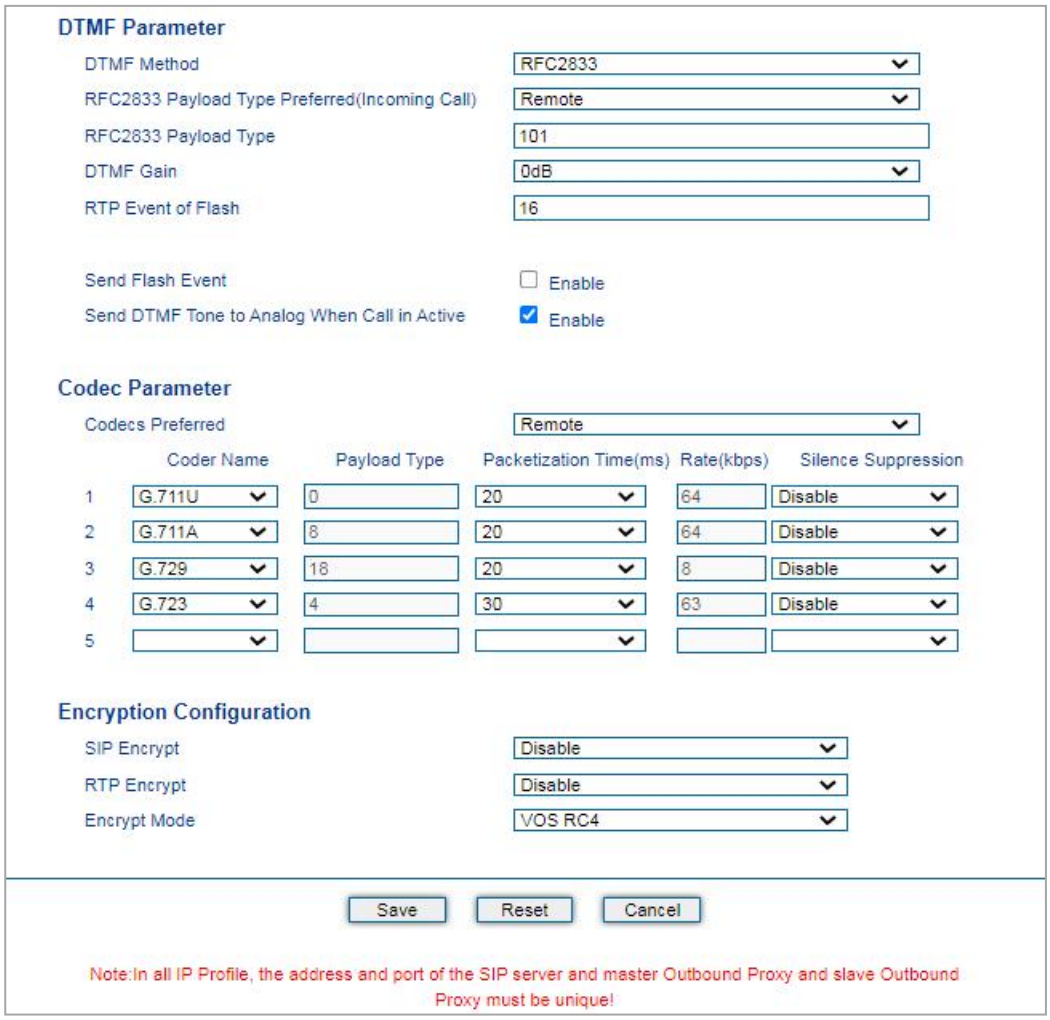
SIP Compatibility

PRACK(RFC3262): Enable

PRACK Only for 18x with SDP: Enable

Early Media: Enable

Early Answer: Enable

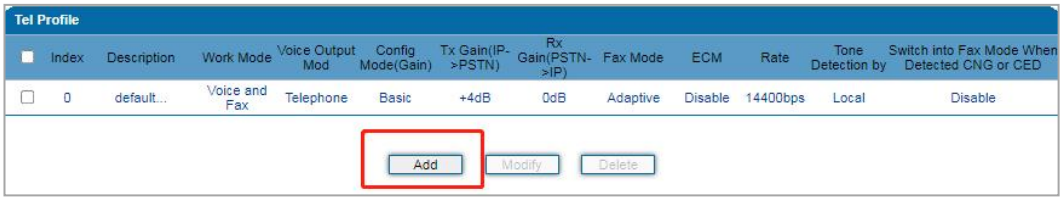


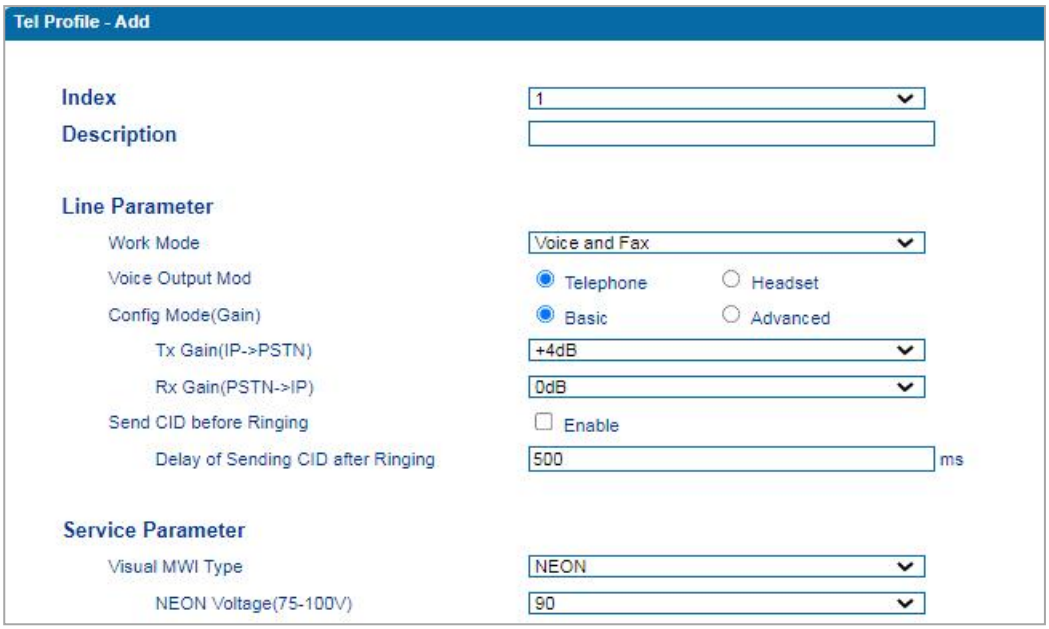

## 4.8 Tel Profile

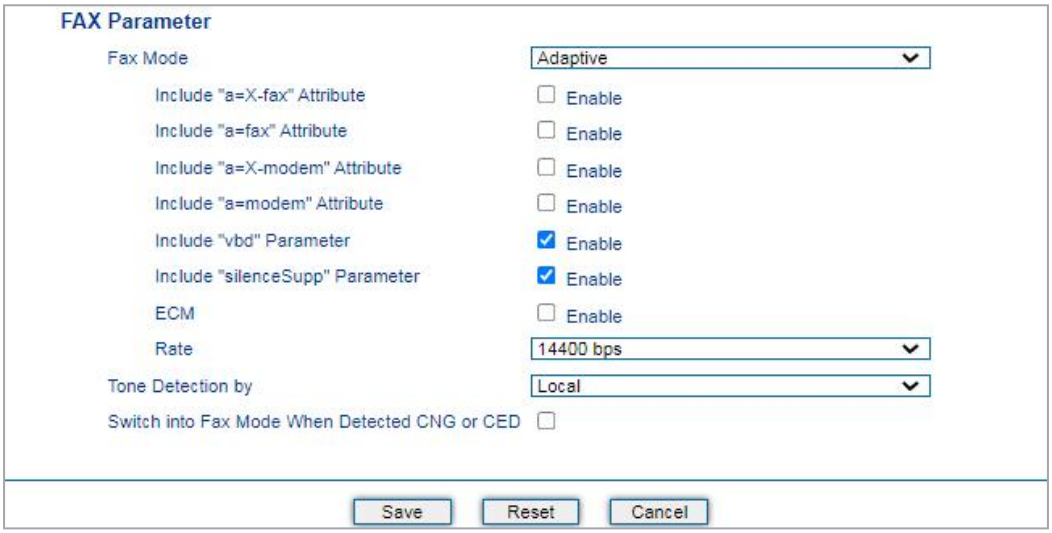
The device supports setting different values for the line parameters to each port. Different ports can be configured with different gains and fax parameters as needed. Tel profile is used to create a configuration group of line parameters, service parameters, and fax parameters for the port. During port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

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

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




## 4.9 Port

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

Figure 4- 11 Configure SIP Account for Port Registration

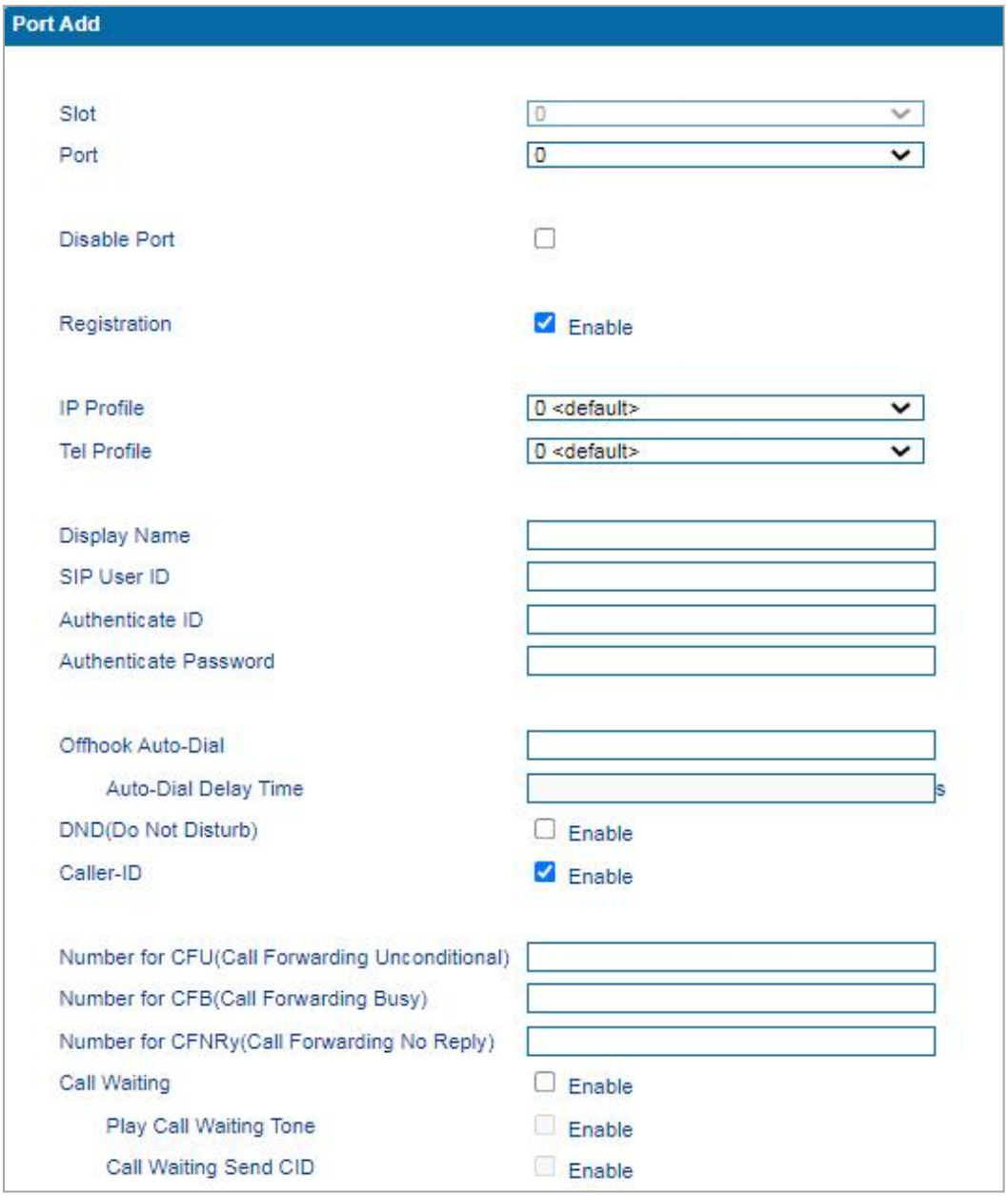



Table 4- 4 Explanation of Parameters Related to SIP Registration

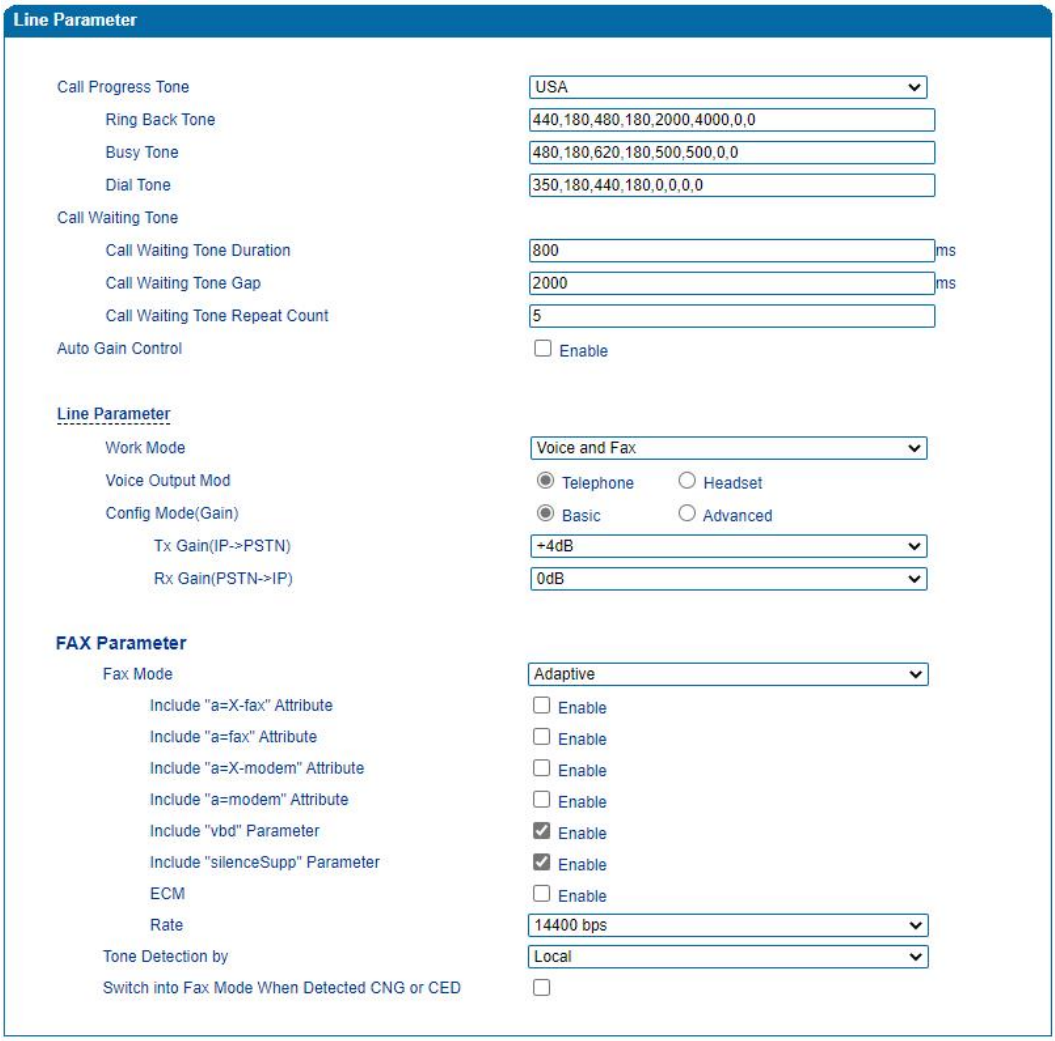
| | |
|---|---|
| **Port** | The FXS port corresponding to this account |
| **Disable port** | Whether to disable port temporally |
| **Registration** | Whether to enable registration for the port |
| **IP Profile** | IP profile (need to be created in advance, refer to 4.6 IP profile configuration) |
| **Tel Profile** | IP policy (need to be created in advance, refer to 4.6 IP policy configuration) |

| | |
|---|---|
| | Tel profile(need to be created in advance, refer to 4.7 Tel profile Configuration) |
| **SIP Display Name** | Description of SIP account. It is used to identify the SIP account. |
| **SIP User ID** | User ID of the SIP account, which is provided by VoIP service provider (ITSP) for registration. Usually it is in the form of digits similar to phone number or an actual phone number. |
| **SIP Authenticate ID** | SIP service subscriber's authenticate ID used for authentication of registration. It can be identical to or different from SIP User ID. |
| **Authenticate password** | SIP service subscriber's authenticate ID used for authentication of registration |
| **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 if this feature is 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. |

# 4.10 Advanced

## 4.10.1 Line Parameter

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




| | |
|---|---|
| **Call Process Tone** | The signal tone standard after a phone is picked up. Choose national standards from the drop-down box. Default value is USA. |
| **Call Waiting Tone** | Set the duration, interval and number of repetitions for the call waiting tone |
| **Auto Gain Control** | Whether to enable automatic gain control |
| **Work Mode** | To set the FXS ports work in both Voice and Fax mode. There are several configure options:<br>• Voice and FAX: to be able to make call and use FAX service<br>• Voice Only: allows to make call only, Fax doesn't work if you connect a fax machine<br>• Fax Only: allows to make Fax call only.<br>• POS only: allows to connect POS terminal only |
| **Gain mode** | IP to PSTN(RX): adjust gain value to analog phone |

| | |
|---|---|
| | PSTN to IP(TX): adjust gain value from analog phone |
| **FAX Parameter** | The EQ-128Sdevice supports the three fax modes: T.38 (IP-based), T.30 (Pass-Through) and Adaptive Fax Mode (automatically match with the peer fax mode). |
| **Fax Mode** | There are three fax modes: T.38, T.30(Pass-through), and Adaptive. |
| **Include "a=X-fax" Attribute** | If this parameter is enabled, "a=X-fax" attribute will be carried in SDP |
| **Include "a=fax" Attribute** | If this parameter is enabled, "a=fax" attribute will be carried in SDP |
| **Include "a=X-modem" Attribute** | If this parameter is enabled, "a=X-modem" attribute will be carried in SDP |
| **Include "a=modem" Attribute** | If this parameter is enabled, "a=modem" attribute will be carried in SDP |
| **ECM** | Whether to enable 'Error Correction Mode'（ECM）. |
| **Rate** | The rate of sending or receiving fax, default value is 14400bps. |
| **Tone Detection by** | Fax sound is detected by caller, callee or automatically. |
| **Switch into Fax Mode When Detect CNG or CED** | If this parameter is enabled, the system will switch into fax mode when CNG or CED is detected. |

### 4.10.2 FXS Parameter

On the **Advanced→ FXS/FXO** page, you can configure FXS parameters which include send polarity reversal, detect hook flash, CID type and so on.

Figure 4- 12 Configure FXS Parameters

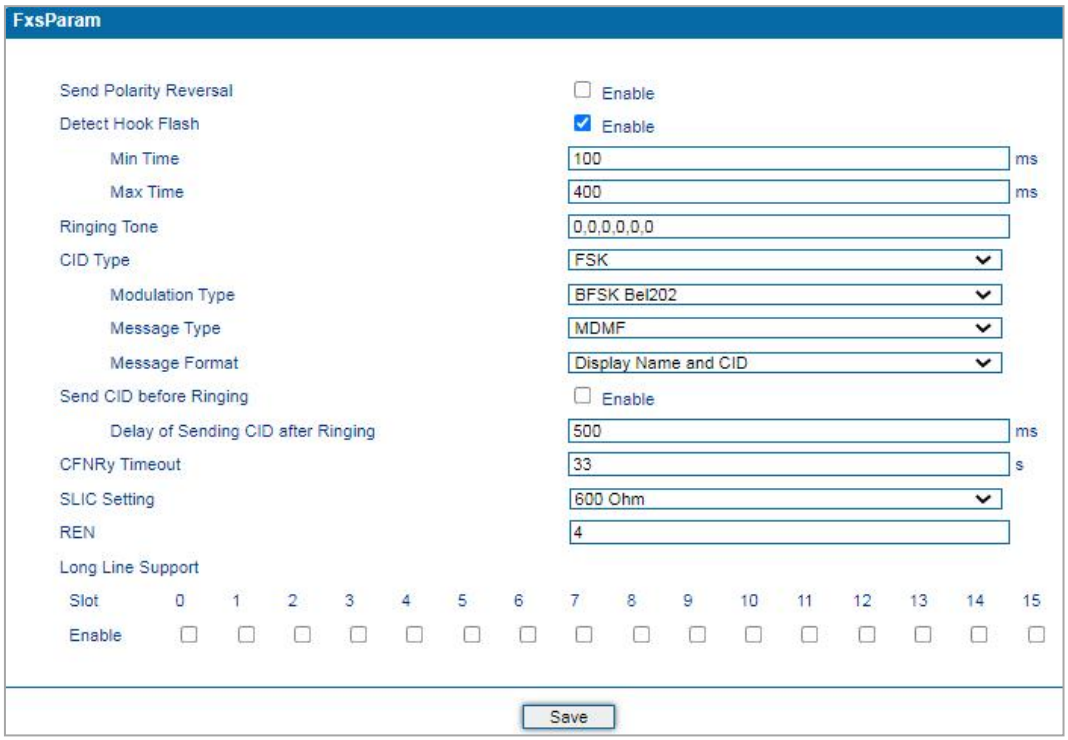



Table 4- 5 Explanation of FXS Parameters

| | |
|---|---|
| **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 device 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. |
| **REN** | The maximum number of extensions that can be connected to a single FXS port. If this parameter is configured, you need to restart the device for the configuration to take effect. |
| **Long Line Support** | Whether to enable 'Long Analog Extension Line'. |

## 4.10.3 Media Parameter

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

Figure 4- 13 Configure Media Parameters

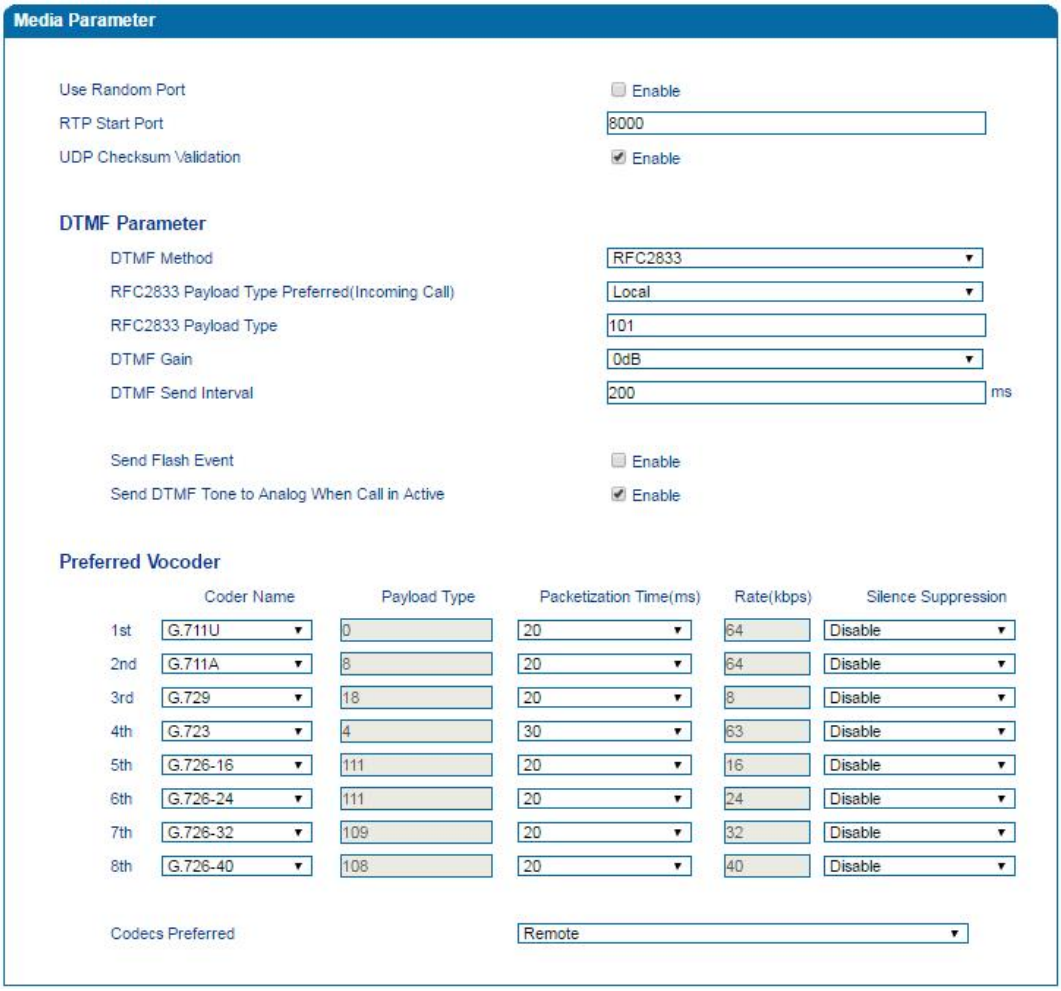


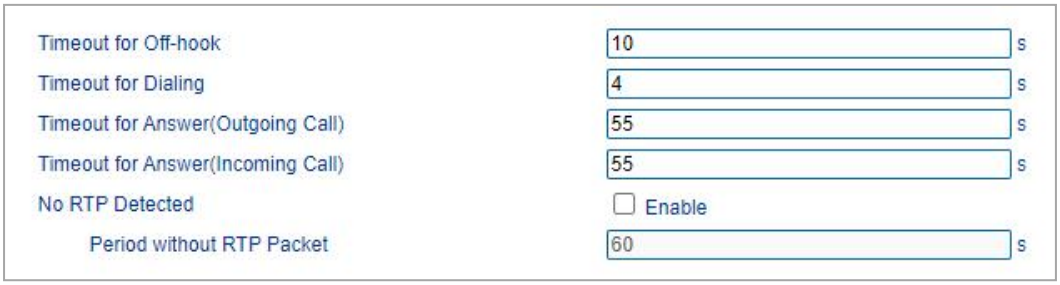

Table 4- 6 Explanation of Media Parameters

| | |
|---|---|
| **Use Random Port** | If this parameter is enabled, the EQ device will choose a port by |

| | |
|---|---|
| | random as the start port for RTP. |
| **RTP Start Port** | When 'Use Random Port' is not selected, you need to configure a start port for RTP. Default RTP start port is 8000 |
| **UDP Checksum Validation** | Choose whether to enable header checksum of UDP |
| **DTMF Method** | Include SINGAL, INBAND and RFC2833 |
| **RFC2833 Payload Type Preferred (Incoming Call)** | For an incoming call, choose local or remote RFC2833 payload type as the preferred payload type |
| **RFC2833 Payload Type** | Local payload value, default value is 101 |
| **DTMF Gain** | Default value is 0 DB |
| **DTMF Send Interval** | The interval for sending DTMF signal. The default value is 200ms. |
| **Send Flash Event** | If this parameter is enabled, the EQ device will send flash-hook event to remote terminal, and thus user does not need to handle it locally |
| **Send DTMF Tone to Analog When Call in Active** | If this parameter is enabled, DTMF tone will be sent to analog phone when there is a call |
| **Coder Name** | The device supports G.729, G.711U, G.711A, G.723, G.726-16/24/32/40. When outgoing calls are made, G.729 will be used. |
| **Payload Type** | Each kind of coding has a unique load value, refer to RFC3551. |
| **Packetization Time** | The time for voice packaging |
| **Rate** | Voice data flow rate; It is defaulted by system. |
| **Silence Suppression** | Default value is 'disabled'. If this parameter is enabled, VoIP transmission bandwidth can be saved, and meanwhile network congestion can be avoided. |
| **Codecs Preferred** | Choose local or remote codec as the preferred codec |

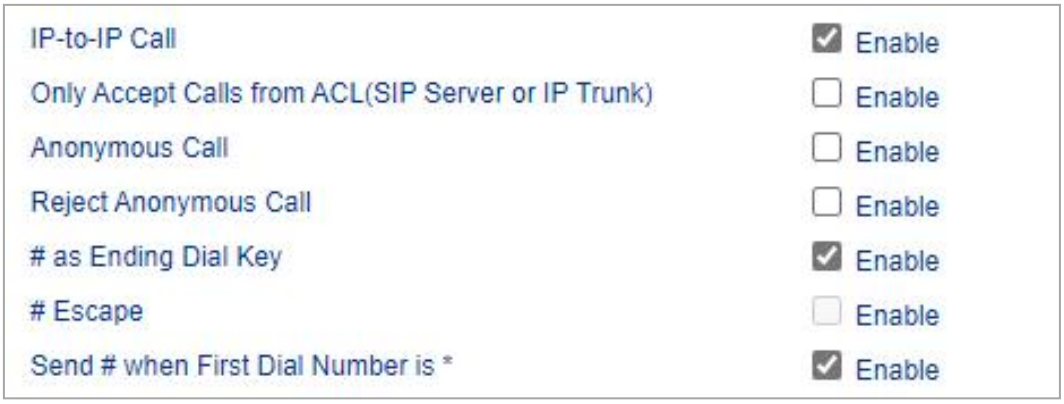
### 4.10.4 Service Parameter

Service parameters include tiemout for dialing, digitmap,MWI message and so on.




| | |
|---|---|
| Timeout for off-hook | Mainly used to define a timer that when the user is off hook an analog phone without dial any digits |
| Timeout for dialing | With the help of dialing timeout, you can limit the time between two digits while users are typing the digits of a number through an extension. If the timeout expires, the gateway will consider the dialing has finished and will try to send message to SIP server. Default value is 4 seconds. |
| Timeout for answer(Outgoing call) | This parameter determines how long the caller party will wait for answer when making outgoing calls through a phone. |
| Timeout for answer(Incoming call) | This parameter determines how long the phone rings when there are incoming calls |
| No RTP Detected | If this parameter is enabled, the situation will be detected when there is no RTP packets received during the set time period. |
| Period without RTP Packet | The time period when there is no RTP packets received. |

| | |
|---|---|
| **SUBSCRIBE for MWI (Message Waiting Indicator)** | MWI is aimed to notify user that there is new voicemail. It is realized in the way of NOTIFY. |
| **MWI Subscription Expires** | The expiry time of MWI subscription; Default value is 3600s. |
| **Voicemail User ID** | The user ID used to access to voicemail |
| **Visual MWI Type** | There are two visual MWI Type, namely NEON and FSK |

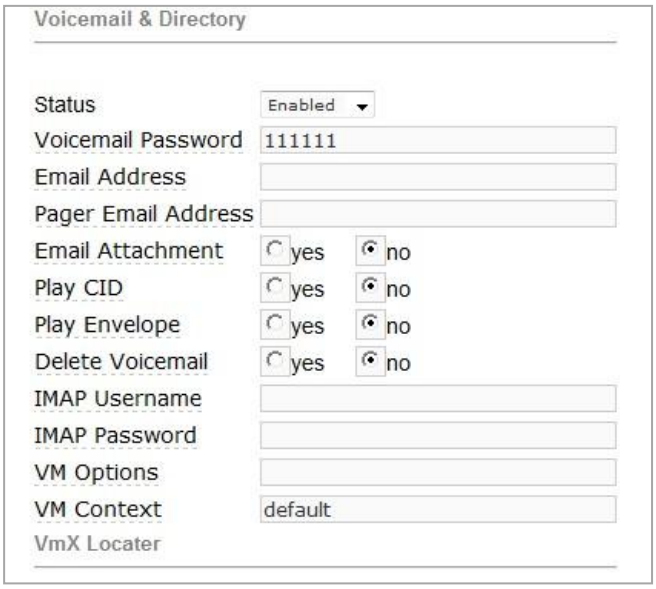



| | |
|---|---|
| **IP-to-IP Call** | If this parameter is enabled, user can dial IP address through a phone to call destination gateway. |
| **Only Accept Call from ACL (SIP server or IP Trunk)** | If this parameter is enabled, the device only accepts incoming call from SIP server only. Default value is 'not enable'. |
| **Anonymous Call** | If this parameter is enabled, 'anonymous' will be included in SIP message. |
| **Reject Anonymous Call** | If this parameter is enabled, all anonymous calls will be rejected. Default value is 'not disable'. |
| **# as ending Dial Key** | If this parameter is enabled, '#' 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. |
| **Send '#' when First Dial Number is '*'** | If this parameter is enabled, '#' will be sent when first dialed digit is '*'. |

## Voicemail instructions:

Here takes the EQ-128S device together with Elastix as the example to introduce how voicemail works in the device.

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




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

(3) On the Web interface of EQ-128S, click **Advanced → SIP Parameter** in the navigation tree and then enter voicemail User ID.

(4) 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:

(5) Dial *200# on the extension which is connected to EQ-128S, and then dial voicemail user ID and password for authentication. After that user will hear voice message.

Digitmap is used for number dialing of calls through FXS ports of the EQ device.

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

## 4.10.5 SIP Compatibility

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

Figure 4- 14 Configure SIP Parameters

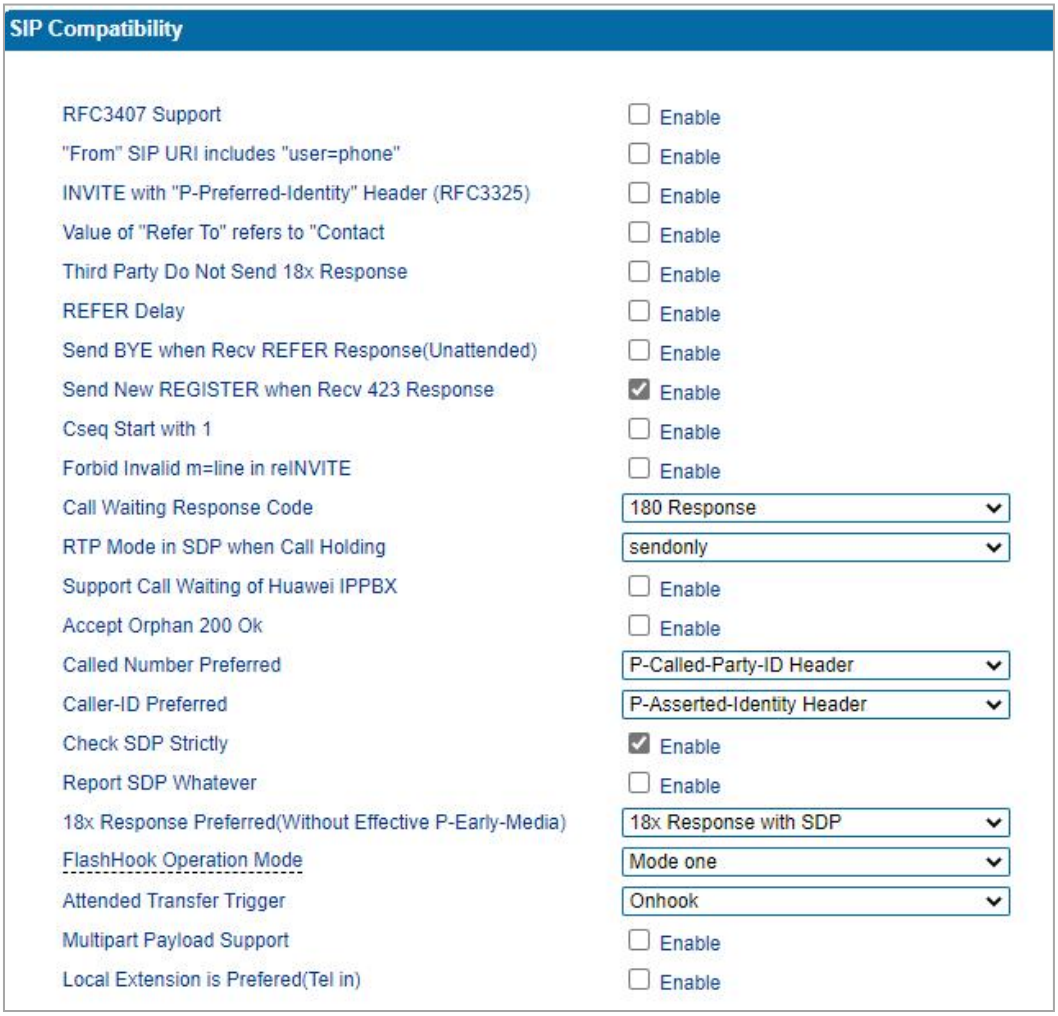



Table 4- 7 Explanation of SIP Parameters

| | |
|---|---|
| **RFC3407 Support** | Whether to enable RFC3407 support. If this parameter is enabled, the device will support RFC3407 which defines the SDP capability of backward compatibility. |
| **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"** | If this parameter is enabled, "P-Preferred-Identity" header will be added in INVITE message for anonymous call (Support |

| | |
|---|---|
| **Header (RFC3325)** | RFC3325). |
| **Only Accept Call from ACL (SIP server or IP Trunk)** | If this parameter is enabled, the device only accepts incoming call from SIP server only. Default value is 'not enable'. |
| **Value of "Refer To" refers to "Contact"** | If this parameter is enabled, 'contract header' needs to be filled in in the 'refer to' field of a SIP message. |
| **Third Party Do Not Send 18x Response** | If this parameter is enabled, the third party will not send 18x response during an attended transfer. |
| **Send BYE when Recv REFER Response (Unattended)** | If this parameter is enabled, the third party will send BYE to release session after receiving REFER during a blind transfer. |
| **Send New REGISTER when Recv 423 Response** | If this parameter is enabled, the value of 'expires' header will be automatically updated and REGISTER will be re-sent after receiving of 423 response. |
| **CSeq Start with 1** | If this parameter is enabled, the value of CSeq starts with '1'. |
| **Forbid Invalid m=line in reINVITE** | If this parameter is enabled, the device will prevent 'invalid m=line' from being carried in the SDP of re-INVITE. |
| **Call Confirm Tone** | If this parameter is enabled, ring-back tone will be played when a call does not receive 180x response. |
| **Call Waiting Response Code** | User can choose 180 or 182 as call waiting response code |
| **RTP Mode in SDP when Call Holding** | Use 'send only' or 'inactive' as RTP mode during call holding. |
| **Support Call Waiting of Huawei IPPBX** | If this parameter is enabled, the device will support call waiting of Huawei IPPBX. |
| **Accept Orphan 200 OK** | If this parameter is enabled, the EQ device will support different 'to-tag 200 OK' in an INVITE session. |
| **Called Number Preferred** | Choose P-Called-Party-ID header or Request-Line |
| **Caller-ID Preferred** | Choose P-Asserted-Identity header or From Header |
| **Report SDP Whatever** | If this parameter is enabled, SDP will be reported anytime |
| **18x Response Preferred** | Choose '18x Response with SDP', 'Last 18x Response' or 'Local Ring Tone Only' |
| **Flashhook Operation Mode** | Choose Mode one, Mode two or Mode three |
| **Attended Transfer Trigger** | Choose 'Onhook' or 'Flashhook +4' |

Figure 4- 15 Configure Default SIP Parameters & Early Media

Table 4- 8 Explanation of Default SIP Parameters & Early Media Parameters

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

Figure 4- 16 Configure Timer in SIP Protocol

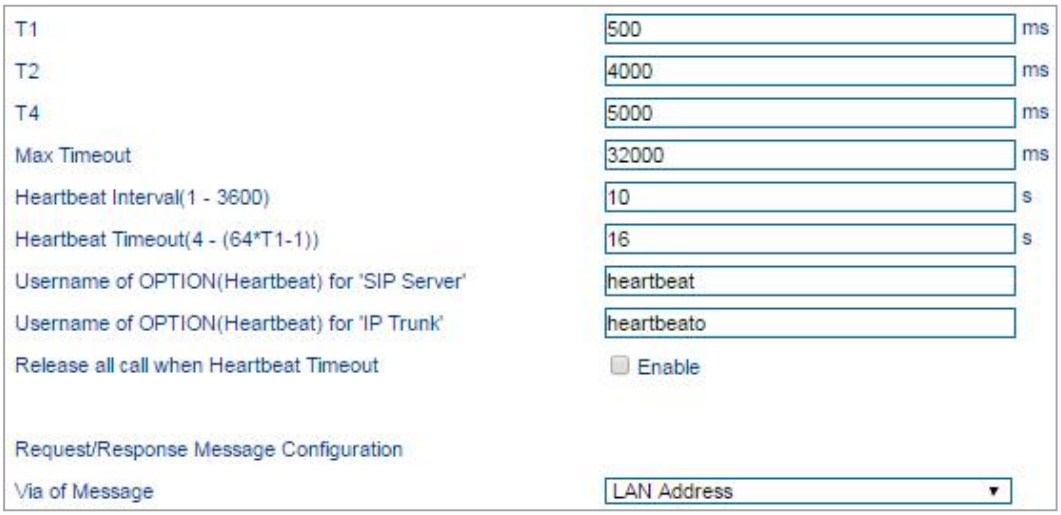
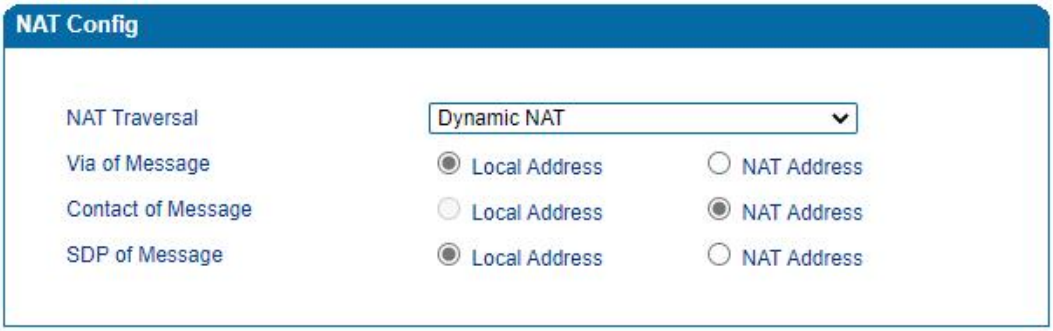



Table 4- 9 Explanation of Timer Parameters in SIP Protocol

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

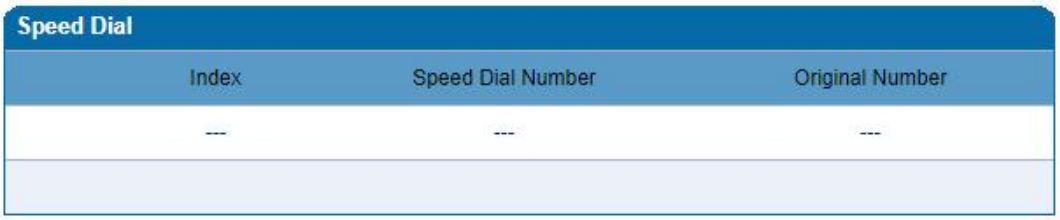
## 4.10.6 NAT Parameter



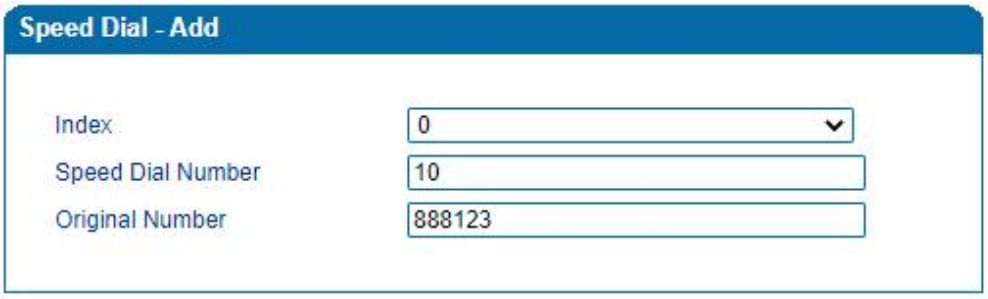

**NAT Traversal (Network Address Translator Traversal)** is a computer networking technique of establishing and maintaining Internet protocol connections across gateways that implement network address translation (NAT). NAT breaks the principle of end-to-end connectivity originally envisioned in the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

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

## 4.10.7 Speed dial




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



**Speed Dial**

| | Index | Speed Dial Number | Original Number |
|---|---|---|---|
| ☐ | 0 | 10 | 888123 |

Total: 1 Entry

## 4.10.8 Feature Code

**Feature Code**

| Feature | Codes | Use Default | Status |
|---|---|---|---|
| **Device Function** | | | |
| Inquiry LAN IP | *158# | ☑ | Enable |
| Inquiry Phone Number | *114# | ☑ | Enable |
| Inquiry PortGroup Number | *115# | ☑ | Enable |
| Remove Login Limit | *154# | ☑ | Enable |
| Setting IP Mode | *150* | ☑ | Enable |
| Configure IP Address | *152* | ☑ | Enable |
| Network Subnet Mask Configure | *153* | ☑ | Enable |
| Network Gateway Configure | *156* | ☑ | Enable |
| Port Voice Up | *170# | ☑ | Enable |
| Port Voice Down | *171# | ☑ | Enable |
| Reset Basic Configuration | *165* | ☑ | Enable |
| Reset Factory Configuration | *166* | ☑ | Enable |
| Restart Device | *111# | ☑ | Enable |
| **Call Function** | | | |
| Call by IP | *47* | ☑ | Enable |
| Call Waiting Activate | *51# | ☑ | Enable |
| Call Waiting Deactivate | *50# | ☑ | Enable |
| Blind Transfer | *87* | ☑ | Enable |
| Call Forward Unconditional Activate | *72* | ☑ | Enable |
| Call Forward Unconditional Deactivate | *73# | ☑ | Enable |
| Call Forward Busy Activate | *90* | ☑ | Enable |
| Call Forward Busy Deactivate | *91# | ☑ | Enable |
| Call Forward No Reply Activate | *92* | ☑ | Enable |
| Call Forward No Reply Deactivate | *93# | ☑ | Enable |
| Do Not Disturb Activate | *78# | ☑ | Enable |
| Do Not Disturb Deactivate | *79# | ☑ | Enable |
| Dial Voicemail | *200# | ☑ | Enable |
| **DTMF Function** | | | |
| Call Holding | *# | ☑ | Enable |
| Call Switch | ## | ☑ | Enable |

Table 4- 10 Explanation of Feature Code

| | |
|---|---|
| Inquiry LAN port IP address | Dial*158# to obtain device's LAN port IP address |
| Inquiry Phone Number | Dial*114# to obtain port account |
| Inquiry PortGroup Number | Dial *115# to obtain port group number |
| Setting IP Mode | *150*0#, means pppmodem, *150*1#, means static IP, *150*2#, means obtain IP address by DHCP, *150*3#, means pppoe. |
| Configure IP Address | *152*+IP, set gateway IP address |
| Network subnet mask configure | *153*+subnet mask, set gateway subnet mask |
| Network Gateway Configure | *156*+gateway IP, set gateway |
| 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 | *91#, forbid call forward busy |

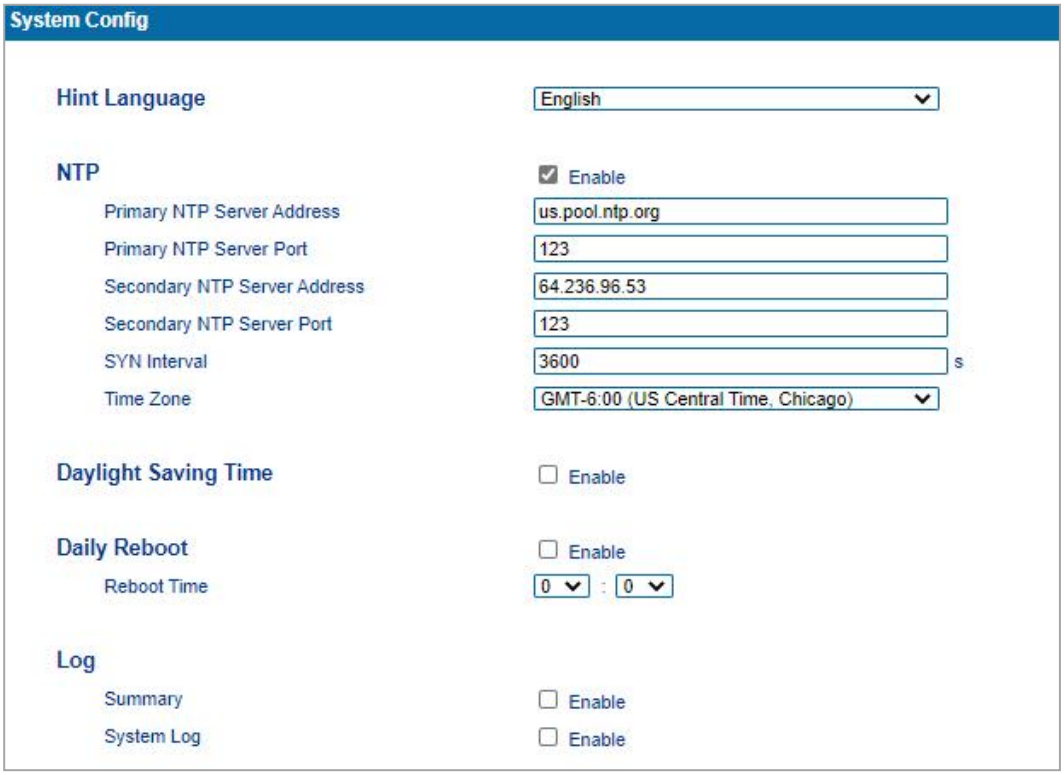
| | |
|---|---|
| Deactivate | |
| Call Forward No Reply Activate | *92*+ forward no reply number# |
| Call Forward No Reply Deactivate | *93#, close this function |
| Do Not Disturb Activate | *78#, enable DND function |
| Do Not Disturb Deactivate | *79#, close DND function |
| Dial Voicemail | *200#, visit voice mail box |

### 4.10.9 System Parameter

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

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

Figure 4- 17 Configure System Parameters



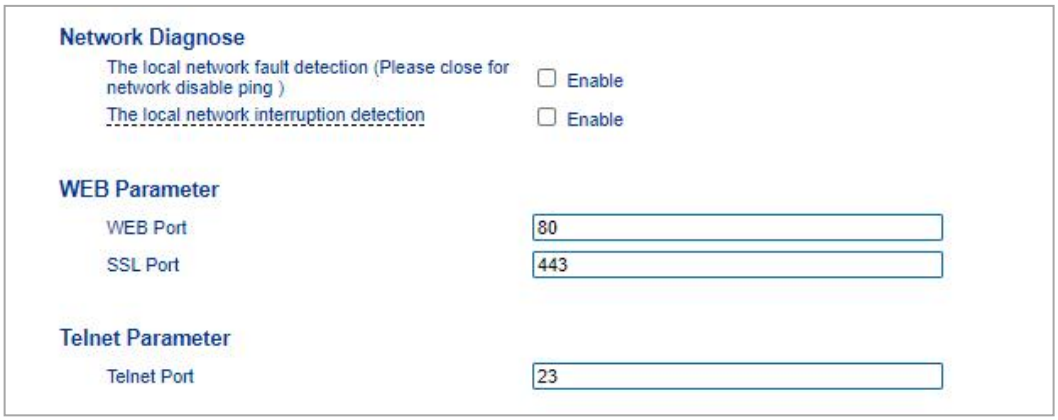



Table 4- 11 Explanation of System Parameters

| | |
|---|---|
| **NTP** | To enable or disable NTP |
| **Primary NTP server address** | The IP address of primary NTP server; default IP address is us.pool.ntp.org. |
| **Primary NTP server port** | The service port of primary NTP server; default port is 123. |
| **Secondary NTP server address** | The IP address of secondary NTP server ; Default IP address is 64.236.96.53 |
| **Secondary NTP server port** | The service port of secondary NTP server; Default port is 123 |
| **SYN Interval** | The interval to synchronize the time of the EQ-128S. Default value is 3600s. |
| **Time Zone** | The time zone of the device; Default configuration is United States central time, Chicago. |
| **Daylight Saving Time** | Enable or disable daylight saving time |
| **Daily Reboot** | Whether to enable daily reboot |
| **Reboot time** | The time to reboot the device daily |
| **WEB Port** | The web port of the device; Default port is 80 |
| **SSL Port** | The SSL port; Default is 443 |
| **Telnet port** | Listening port of telnet service; Default port is 23 |

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

# 4.11 Call & Routing

## 4.11.1 Port Group

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

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

Figure 4- 18 Add Port Group

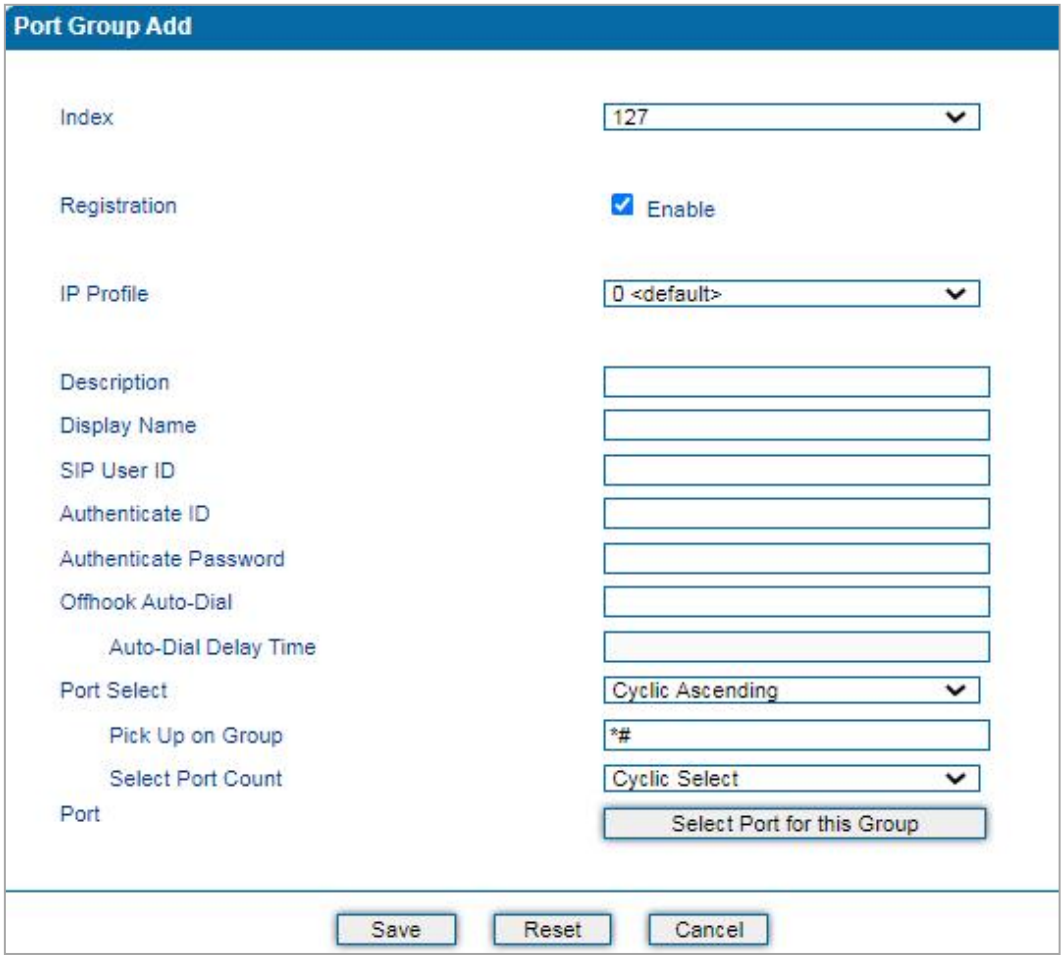



Table 4- 12 Parameter Explanation of Port Group

| | |
|---|---|
| **Index** | The NO. of the port group; It uniquely identifies a route. |
| **Description** | The description of the port group; it is used to identify the port group. |

| | |
|---|---|
| **Display Name** | Display name of the port group, which will be used in SIP message, for example:<br>INVITE sip:bob@biloxi.com SIP/2.0<br>Via: SIP/2.0/UDPpc33.atlanta.com;branch=z9hG4bK776asdhds<br>Max-Forwards: 70<br>To: Bob <sip:bob@biloxi.com><br>From: Alice <sip:alice@atlanta.com>;tag=1928301774<br>Here Bob and Alice is the display name |
| **SIP User ID** | User ID of this SIP account, which is provided by VoIP service provider (ITSP). It is usually in the form of digit similar to phone number or an actual phone number. |
| **Authenticate ID** | SIP service subscriber's ID for authentication; it can be identical to or different from SIP User ID. |
| **Authenticate Password** | SIP service subscriber's password for authentication |
| **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 auto-dialing will be delayed |
| **Port Select** | It specifies the policy for selecting a port for ringing in the port group<br>• Ascending: the device always selects a port from the minimum number.<br>• Cyclic ascending: the device always selects a port from a number next to the number selected last time. If the maximum number was selected last time, the next selected number is the minimum number. The sequence moves in cycles like this.<br>• Descending: the device always selects a port from the maximum number.<br>• Cyclic descending: the device always selects a port from a number next to the number selected last time. If the minimum number was selected last time, the next selected number is the maximum number. The sequence moves in cycles like this.<br>• 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 |

### 4.11.2 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 4- 19 Configure IP Trunk

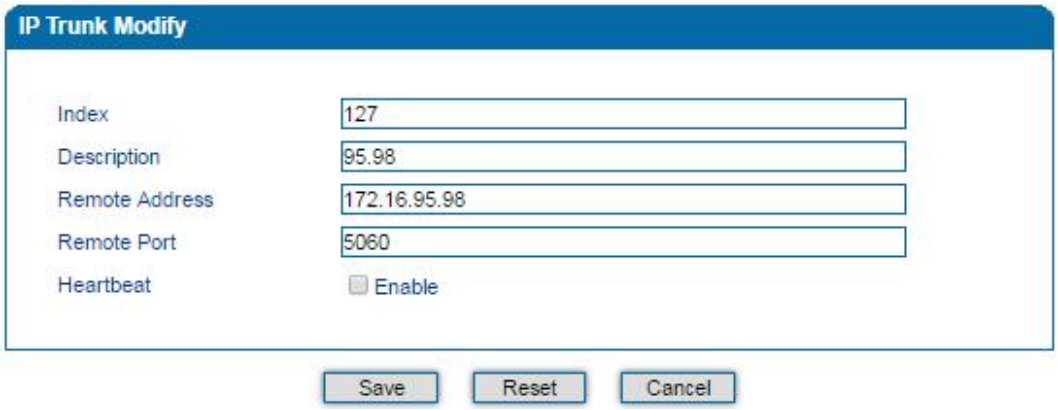



Table 4- 13 Explanation of IP Trunk Parameters

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

### 4.11.3 Routing Parameter

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

Figure 4- 20 Configure Routing Parameter

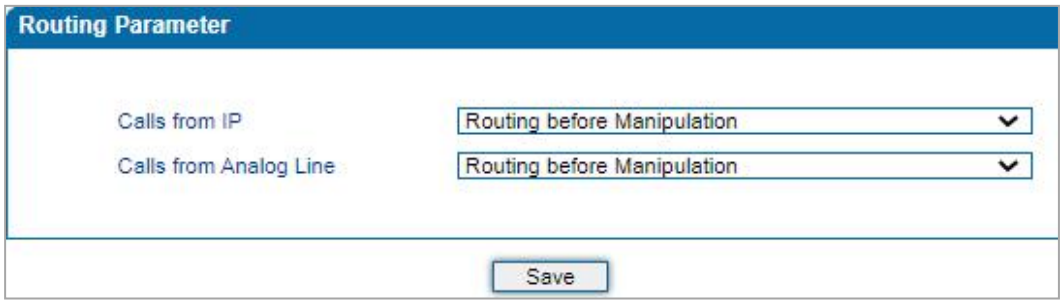


Table 4- 14 Explanation of Routing Parameters

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

### 4.11.4 IP → Tel Routing

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

Figure 4- 21 Add IP →Tel Route

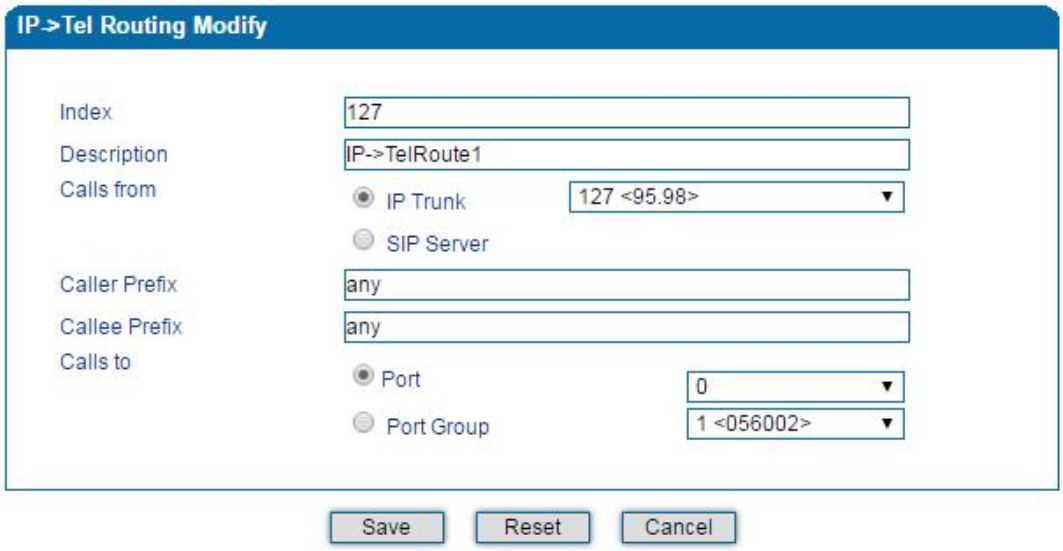



Table 4- 15 Parameter Explanation of IP →Tel Routes

| | |
|---|---|
| **Index** | Index of the IP →Tel routing; range is from 0 to127; 0 is the highest priority. |
| **Description** | Description of the IP →Tel routing; it is used to identify the IP → Tel routing. |
| **Calls from** | Choose calls from IP trunk or SIP server; 'any' means any IP addresses. |
| **Caller Prefix** | The prefix of the caller number, which helps match routing exactly. Its length is less than or equal to the caller number. For example, if caller number is 2001, the caller prefix can be 200 or 2. 'Any' means the prefix matches any caller number. |
| **Callee Prefix** | The prefix of the called number, which helps match routing exactly. Its length is less than or equal to the called number. If the called number is 008675526456659, the called prefix can be 0086755 or 00.,"any" means the prefix matches any called number |
| **Calls to** | Which port or port group to which calls are routed. |

## 4.11.5 Tel → IP/Tel Routing

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

Figure 4- 22 Add Tel →IP/Tel Route

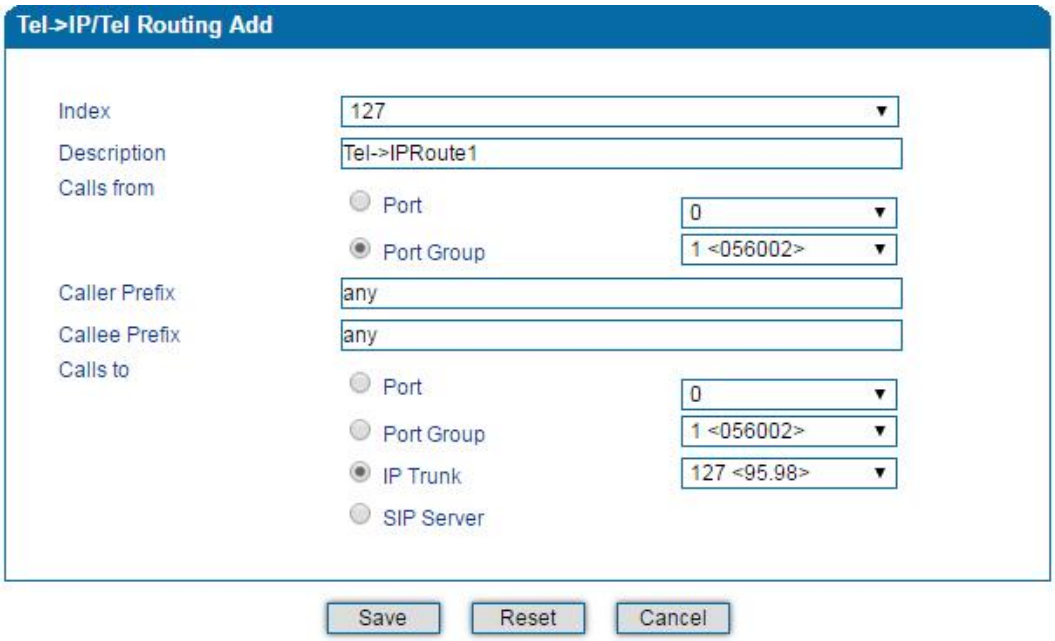



Table 4- 16 Explanation of Tel →IP/Tel Route

| | |
|---|---|
| **Index** | The index of this Tel → IP/Tel routing; range is from 0 to 127. Each index cannot be used repeatedly. Routing priority: 0 is the highest priority. |
| **Description** | The description of this Tel → IP/Tel routing; it is used to identify the routing. |
| **Calls From** | Choose calls are from a port or a port group |
| **Caller Prefix** | The prefix of the caller number, which helps match routing exactly. Its length is less than or equal to the caller number. For example, if caller number is 2001, the caller prefix can be 200 or 2. 'any' means the prefix matches any caller number. |
| **Callee Prefix** | The prefix of the called number, which helps match routing exactly. Its length is less than or equal to the called number. If the called number is 008675526456659, the called prefix can be 0086755 or 00. 'any' means the prefix matches any called number. |
| **Calls to** | Choose calls are routed to a port, port group, IP trunk or SIP server |

# 4.12 Manipulation

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

## 4.12.1 IP → Tel Called

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

Figure 4- 23 Add IP →Tel Called Number Manipulation

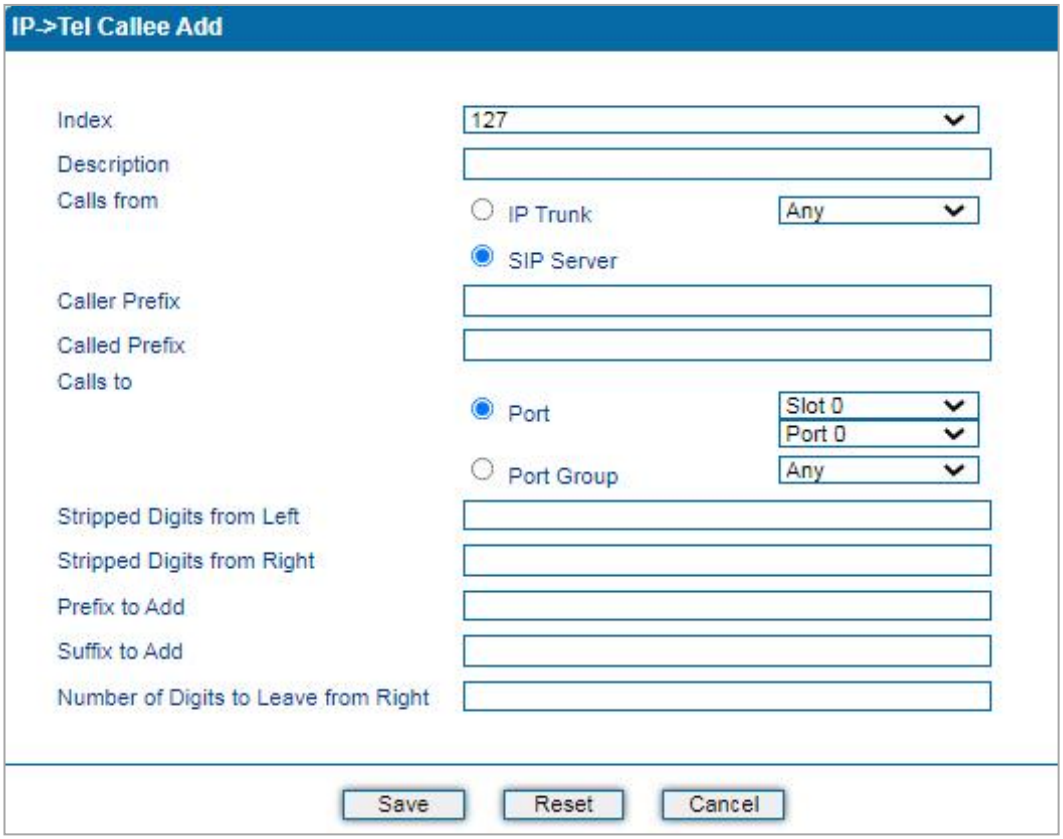



Table 4- 17 Explanation of Parameters for IP →Tel Called Number Manipulation

| | |
|---|---|
| **Index** | The index of this manipulation; range is from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority |
| **Description** | Description of this manipulation; it is used to identify this manipulation. |
| **Calls From** | Determine the calls come from IP trunk or SIP server |
| **Caller Prefix** | Set a prefix for caller number. The prefix's length is less than or equal to that of the caller number, which helps to match the caller number of this call. If caller number is 2001, the caller prefix can be 200 or 2. "any" |

| | |
|---|---|
| | means match any caller number. |
| **Called Prefix** | Set a prefix for called number. The prefix's length is less than or equal to called number, which helps to match the called number. If called number is 008675526456659, the called prefix can be 0086755 or 00. "any" means match any called number. |
| **Calls to** | Determine the call is routed to a port or a port group. |
| **Stripped Digits from Left** | The number of digits which are lessened from the left of the called number |
| **Stripped Digits from Right** | The number of digits which are lessened from the right of the called number |
| **Prefix to Add** | The prefix added to the called number after its digits are lessened. |
| **Suffix to Add** | The suffix added to the called number after its digits are lessened. |

### 4.12.2 Tel → IP/Tel Caller

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

Figure 4- 24 Add Tel → IP/Tel Caller Number Manipulation

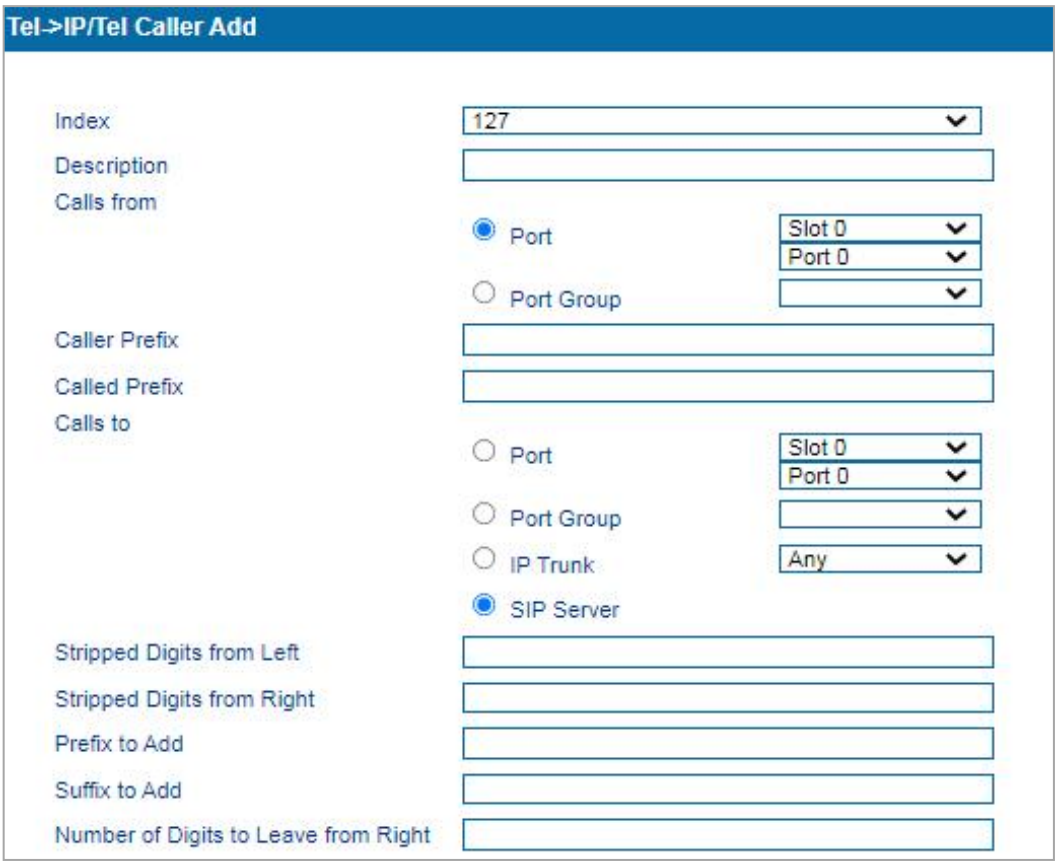



Table 4- 18 Explanation of Parameters for IP →Tel Called Number Manipulation

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

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

### 4.12.3 Tel → IP/Tel Called

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

Figure 4- 25 Add Tel → IP/Tel Called Number Manipulation

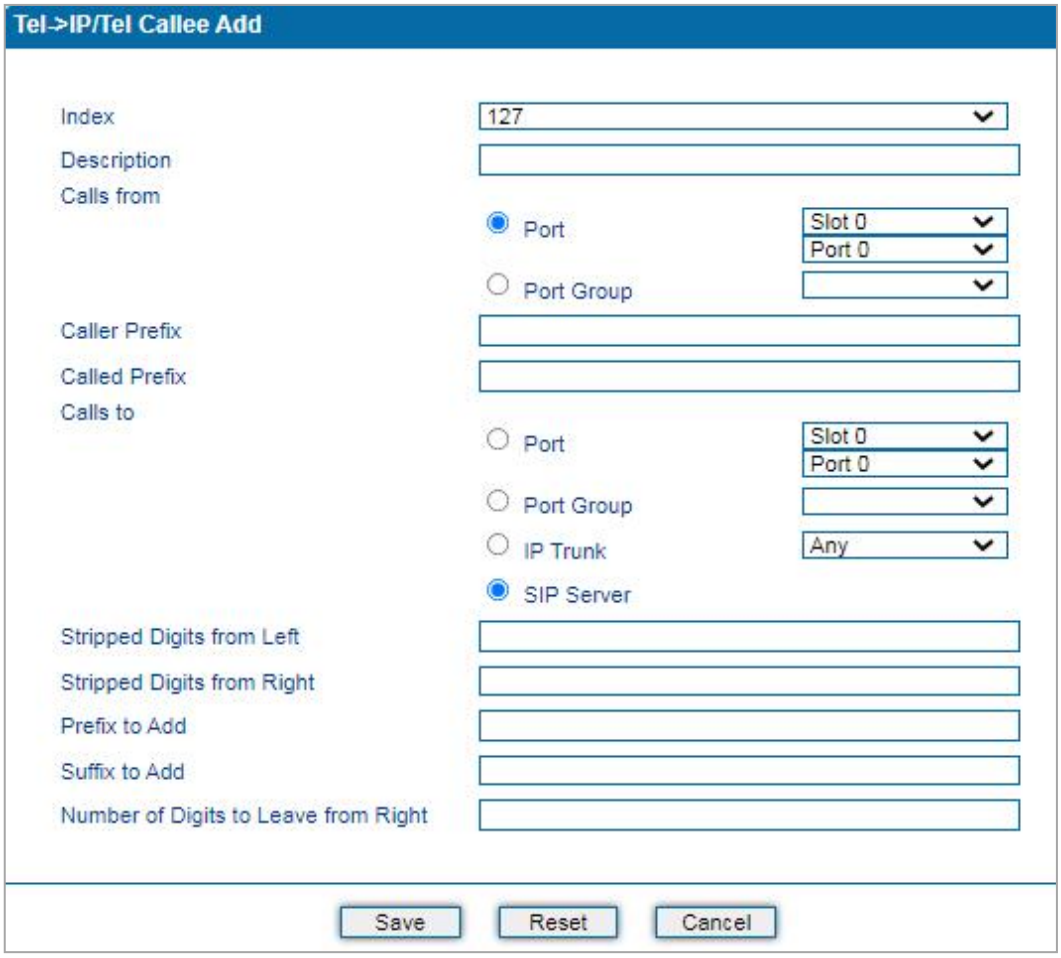



Table 4- 19 Explanation of Parameters for Tel → IP/Tel Callee Number Manipulation

| | |
|---|---|
| **Index** | The index of this manipulation; range is from 0 to 127. Each index cannot be used repeatedly. 0 is the highest priority |
| **Description** | Description of this manipulation; it is used to identify this manipulation. |

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

## 4.13 Management

### 4.13.1 TR069

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

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

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

Username and password are used for ACS authentication.

Figure 4- 26 Configure TR069 Parameter

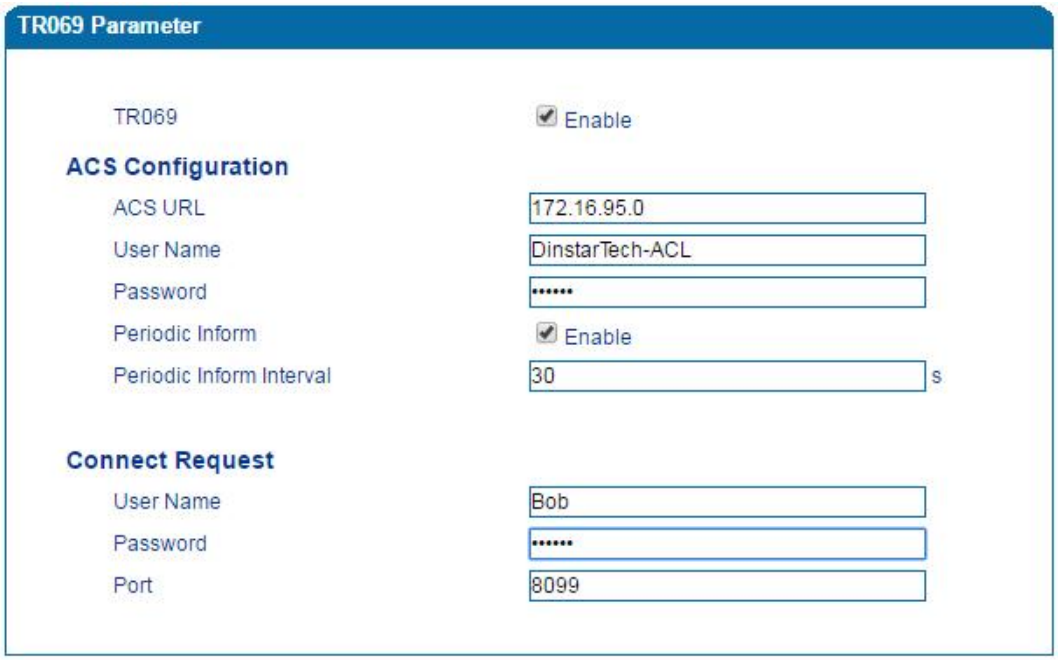



Table 4- 20 Explanation of TR069 Parameters

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

### 4.13.2 SNMP

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

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

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

Figure 4- 27 Configure SNMP Parameters

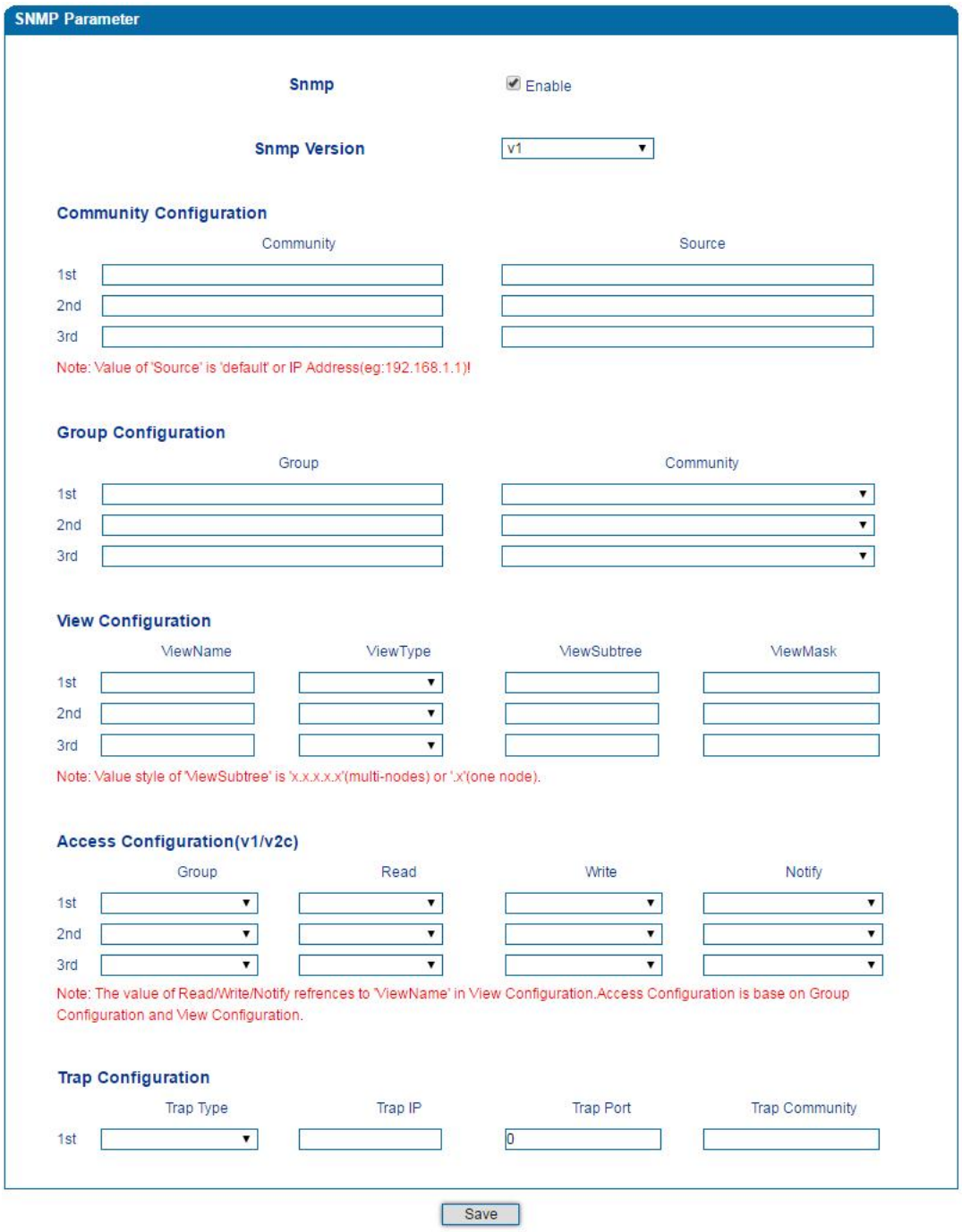


Table 4- 21 Explanation of SNMP Parameters

| | |
|---|---|
| SNMP | The EQ device supports three versions of SNMP, namelyV1、V2C and V3. |
| **Community Configuration** | Community configuration exists in V1 and V2C.<br>**Community:** fill in a community name used to read through SNMP protocol; it is a character string.<br>**Source:** The IP address of SNMP server.<br>SNMP server cannot identify the packets sent from EQ unless the community configured in EQ matches with the community configured in SNMP server. |
| **Group Configuration** | Group configuration exists in V1 and V2C and V3.<br>**Group:** fill in a group name which is used to identify the group; it's a character string.<br>**Community:** fill in a community which means this community has joined in the group.<br>In the following, access permission of read, write and notify is configured for each group. |
| **View Configuration** | View configuration exists in V1, V2C and V3.<br>**ViewName:** fill in a view name which is used to identify this view.<br>**ViewType:** choose 'Included' or 'Excluded'. 'Included' means the view includes the OID of the corresponding ViewSubtree, while 'Excluded' means the OID of the corresponding ViewSubtree is excluded from this view.<br>**ViewSubtree:** fill in the OID of the view subtree.<br>**ViewMask:** it is used to withdraw a row of a table, such as an Ethernet port. |
| **Access Configuration** | Access configuration exists in V1, V2C and V3, under which permission of read, write or notify is configured for a community group.<br>**Group:** choose a group name that has been configured.<br>**Read:** Choose a 'read' view for the group.<br>**Write:** Choose a 'write' view for the group.<br>**Notify:** Choose a 'notify' view for the group. |
| **Trap Configuration** | Trap configuration exists in V1, V2C and V3, which is aimed to send trap alarm.<br>**Trap Type:** Choose V1, V2C and Inform.<br>**Trap IP:** the IP address of the destination SNMP server where trap alarm is sent.<br>**Trap Port:** the port of the destination SNMP server, which will receive trap alarm.<br>**Trap Community:** the community configured in the destination SNMP server. |
| **User Configuration** | User configuration exists in V3. When V3 transmits SNMP packets in an encryption way, this item needs to be configured. |

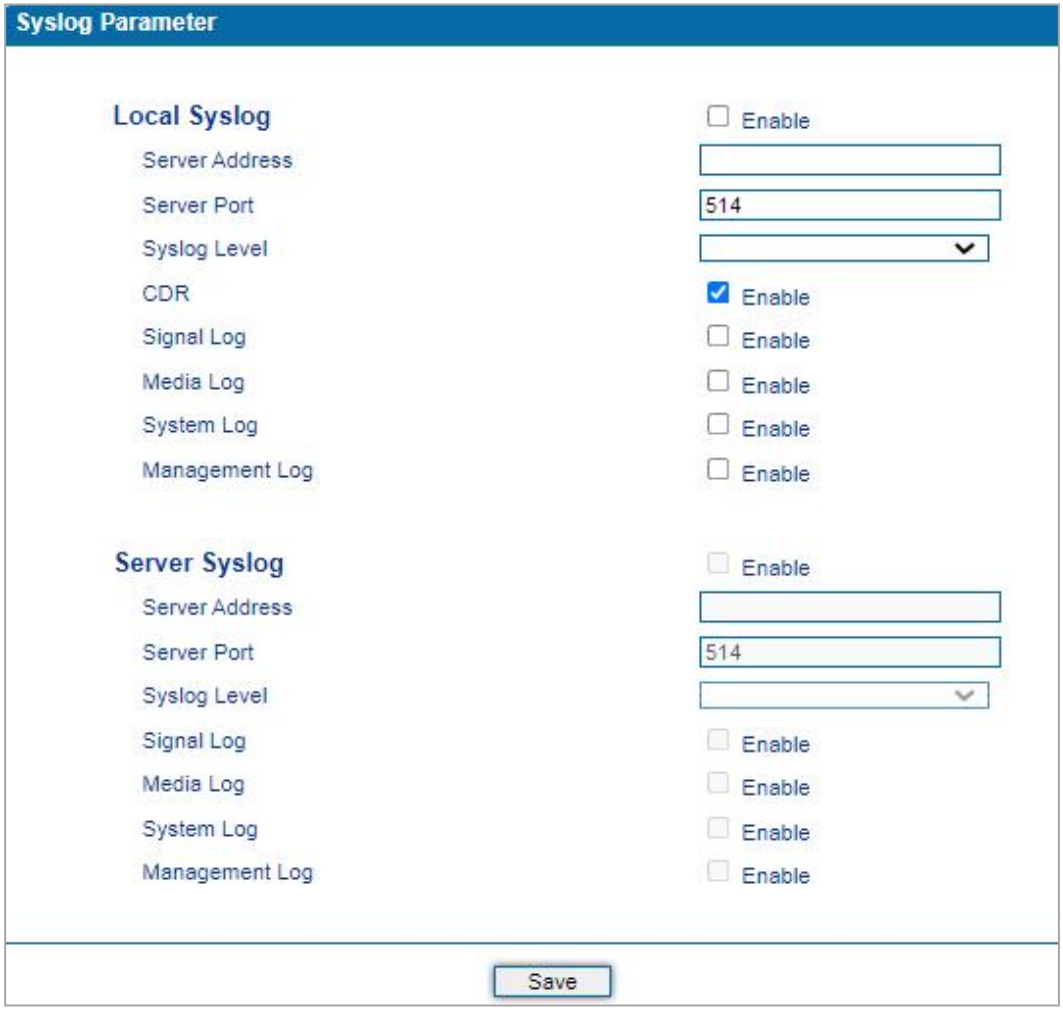
| | |
|---|---|
| | **User:** fill in a user name used to authenticate.<br>**AuthType:** choose MD5 or SHA as authentication type.<br>**AuthPassword:** the password used to authenticate.<br>**Privacy Type:** Choose DES, AES or AES 128 as encryption type.<br>**Privacy Password:** the encryption password. |

### 4.13.3 Syslog

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

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

Figure 4- 28 Configure Syslog Parameters



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

### 4.13.4 Provision

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

Figure 4- 29 Provision

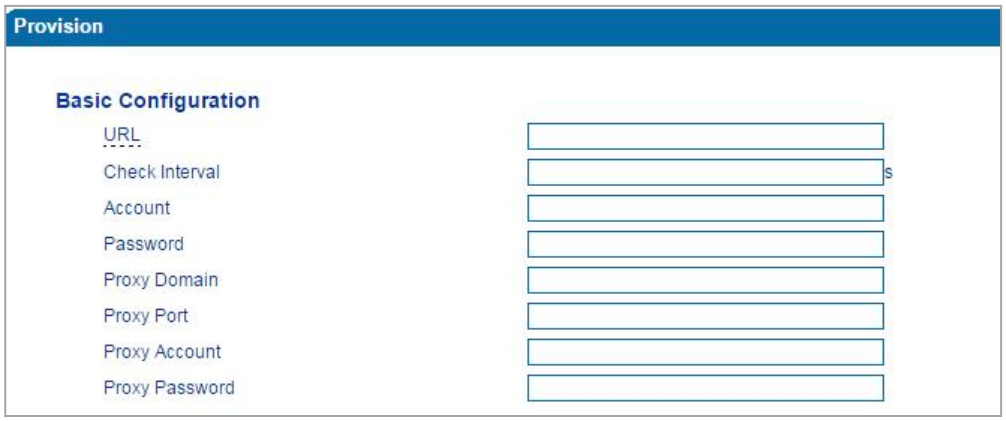



Table 4- 22 Explanation of Provision Parameters

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

### 4.13.5 Cloud server

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

Figure 4- 30 Configure Cloud Server

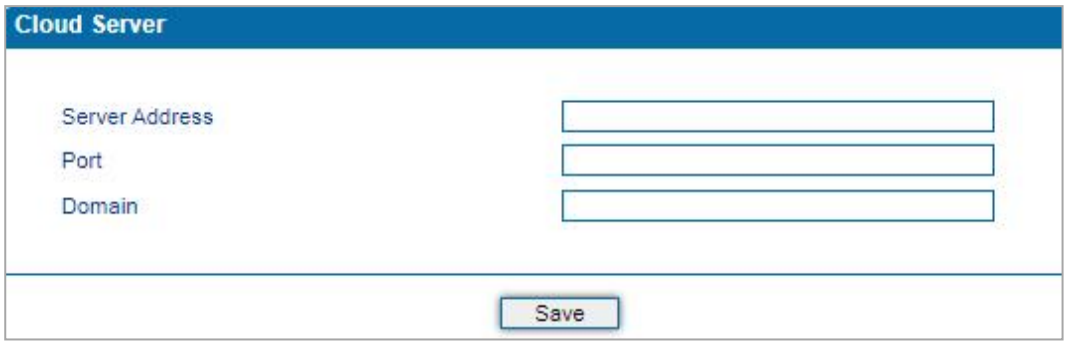


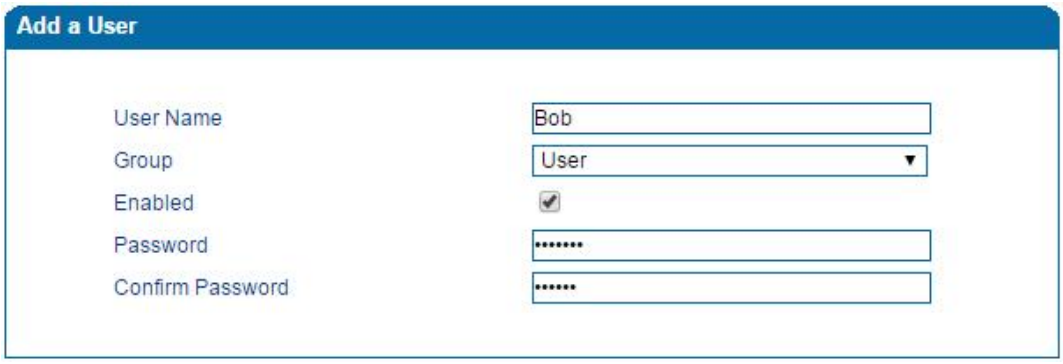

Table 4- 23 Explanation of Parameters for Cloud Server

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

## 4.13.6 User Manage

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

Figure 4- 31 Modify Username and Password




## 4.13.7 Remote Server

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

Figure 4- 32 Configure Remote Server

## 4.13.8 Record Parameter

After configuring the record server, you can upload the call voice of the FXS port of the device to the record server. The record configuration page is shown in Figure 4.12-8:

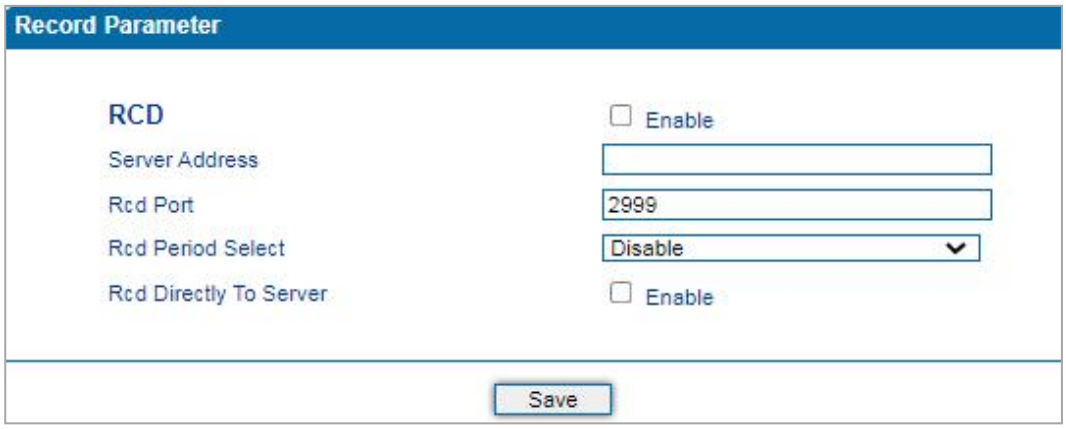



Table 4- 24 Explanation of Record Parameters

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

## 4.13.9 Radius Parameter

After the Radius server is configured, you can log into the gateway after successful Radius authentication. The Radius server configuration page is shown in Figure 4.12-9:

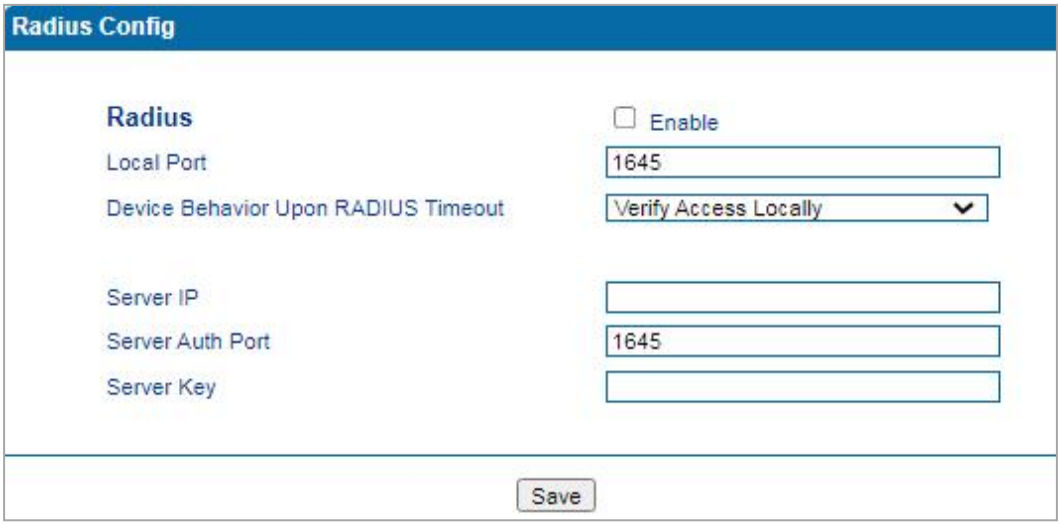



Table 4- 25 Explanation of Parameters for Radius Config

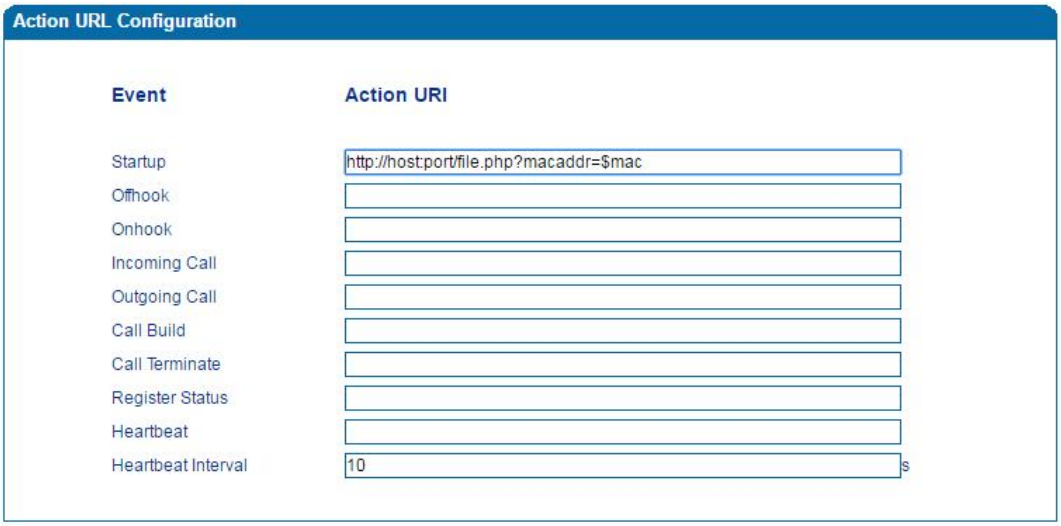
| | |
|---|---|
| **Radius** | Enable or disable Radius |
| **Local Port** | The port of the local Radius client |
| **Device Behavior Upon RADIUS Timeout** | Processing after Radius authentication timeout.<br>Verify Access Locally : After the timeout, when verifying the user name and password of the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

## 4.13.10 Action URL

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

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

Figure 4- 33 Configure Action URL




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

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

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

### 4.13.11 SIP PNP

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

1) The gateway sends request packet for SIP subscription to the multicast at intervals
2) The gateway receives the Notify message and reads the URL address of the deployment server in the message
3) Initiate the Provision to the URL to restore the profile or upgrade the software version

Figure 4- 34 Configure SIP PNP

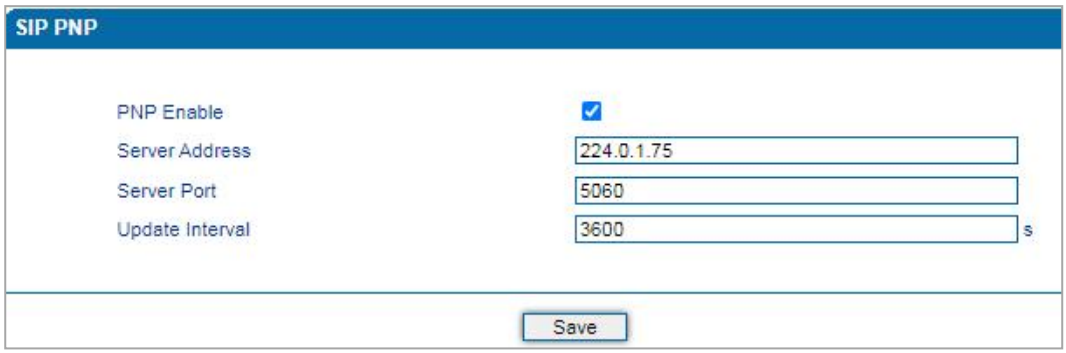



Table 4- 26 Explanation of Parameters for SIP PNP

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

### 4.13.12 NMS Configuration

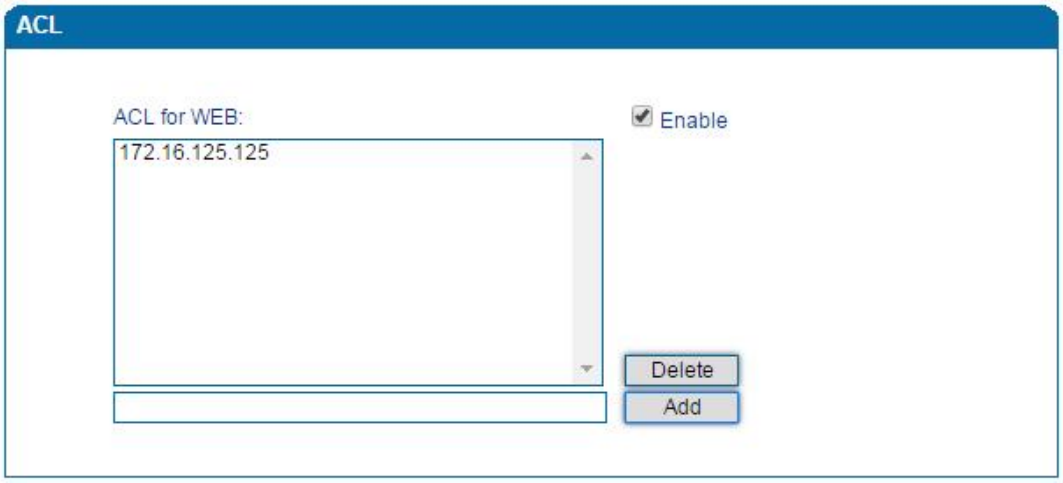
NMS is a public/private cloud-based device management platform. After users register with the NMS, you can perform batch upgrades, status monitoring, and view alarm information for the devices. To apply for cloud management server(NMS) , please contact your local distributor.

## 4.14 Security

### 4.14.1 WEB ACL

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

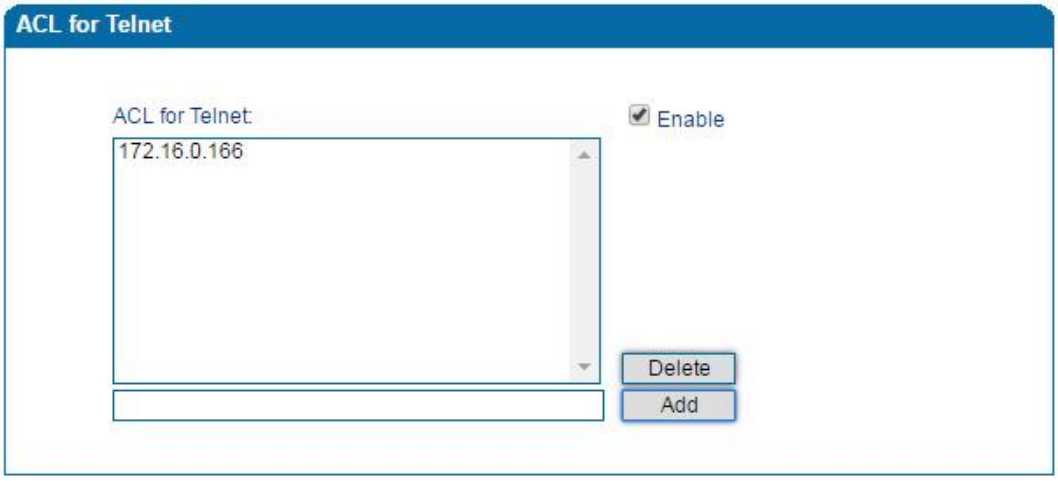
Figure 4- 35 Add IP Address to Web ACL




## 4.14.2 Telnet ACL

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

Figure 4- 36 Add IP Address to Telnet ACL

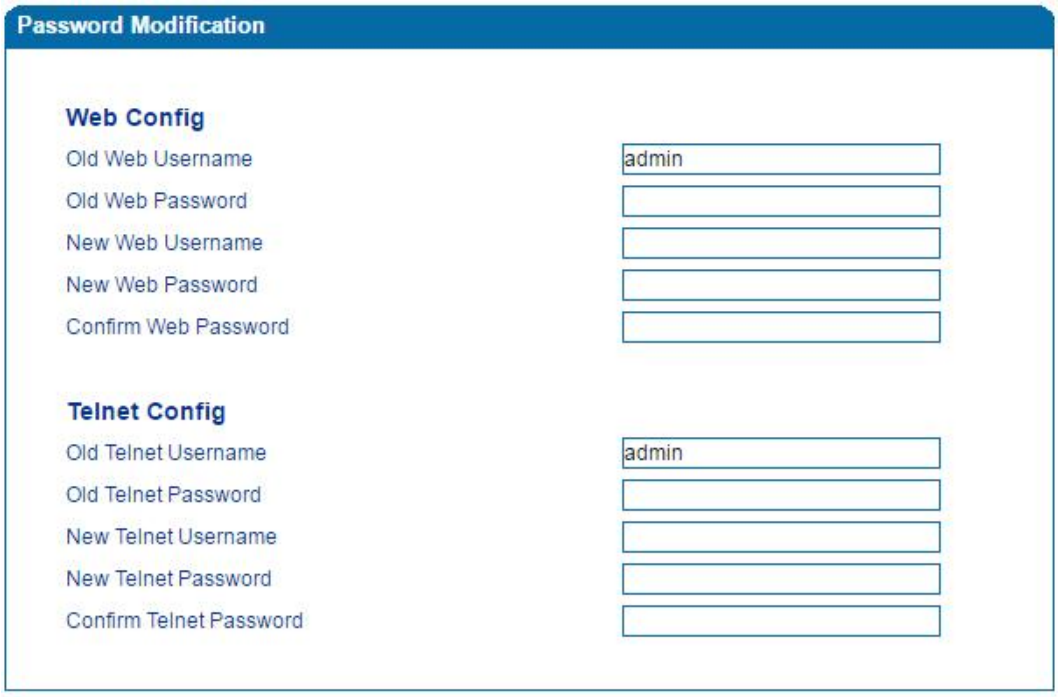



## 4.14.3 Passwords

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

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

Figure 4- 37 Modify Username and Password

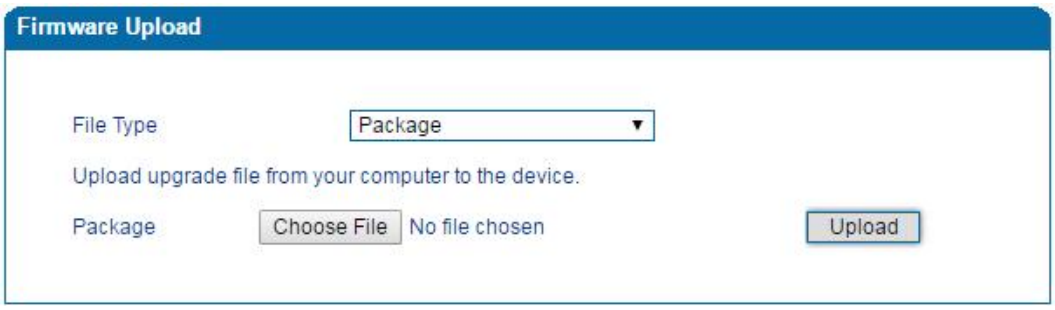



# 4.15 Tools

## 4.15.1 Firmware Upload

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

Figure 4- 38 Upload Firmware




Steps of Firmware Uploading:

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

Step 2. Prepare firmware package.

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

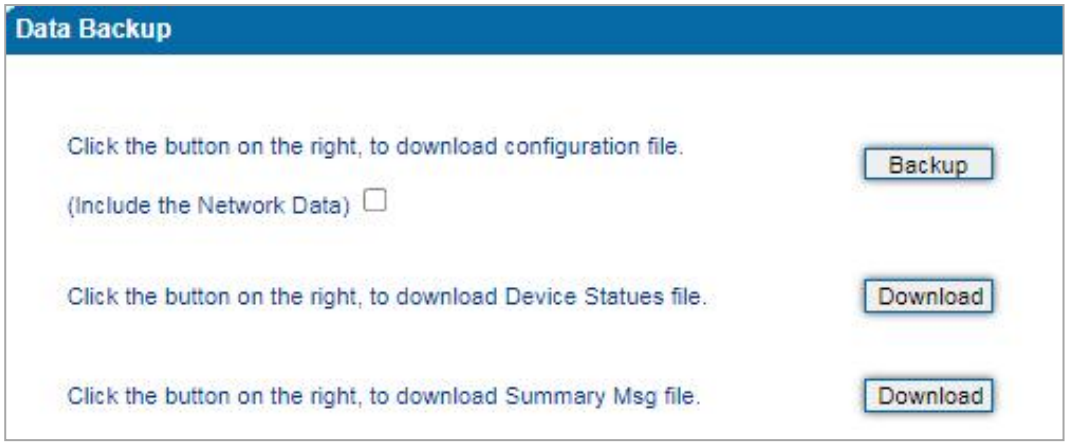
Step 4. Keep waiting until it prompts 'Software loaded successfully!'

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

## 4.15.2 Data Backup

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

Figure 4- 39 Backup Data




## 4.15.3 Data Restore

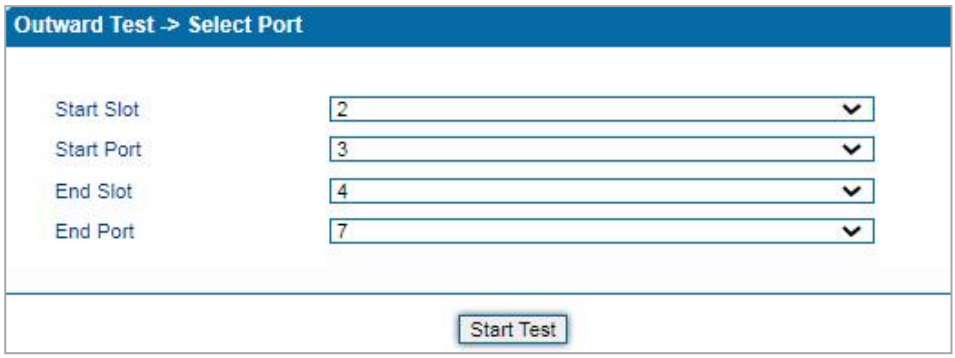
On the **Tools → Data Restore** page, you can restore configuration data through uploading a data file from local computer. The restored configurations will take effect after the device is restarted.

Figure 4- 40 Restore Data

## 4.15.4 Outward Test

Outward test enables you to diagnose the physical function of FXS port which follow the GR909 standard. To start outward test, select the FXS ports to be tested and click 'Start'. The testing may cost a few minutes.

Figure 4- 41 Execute Outward Test




**Test Results:**

OK: the physical function of the tested FXS ports is working well;

FAIL: There's something wrong with the physical function of the tested FXS ports.

## 4.15.5 Ping Test

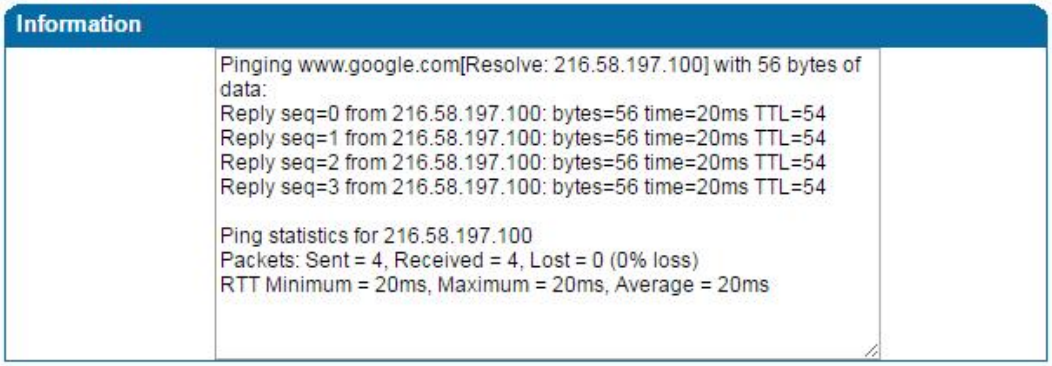
**Ping** is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

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

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

Figure 4- 42 Execute Ping Test



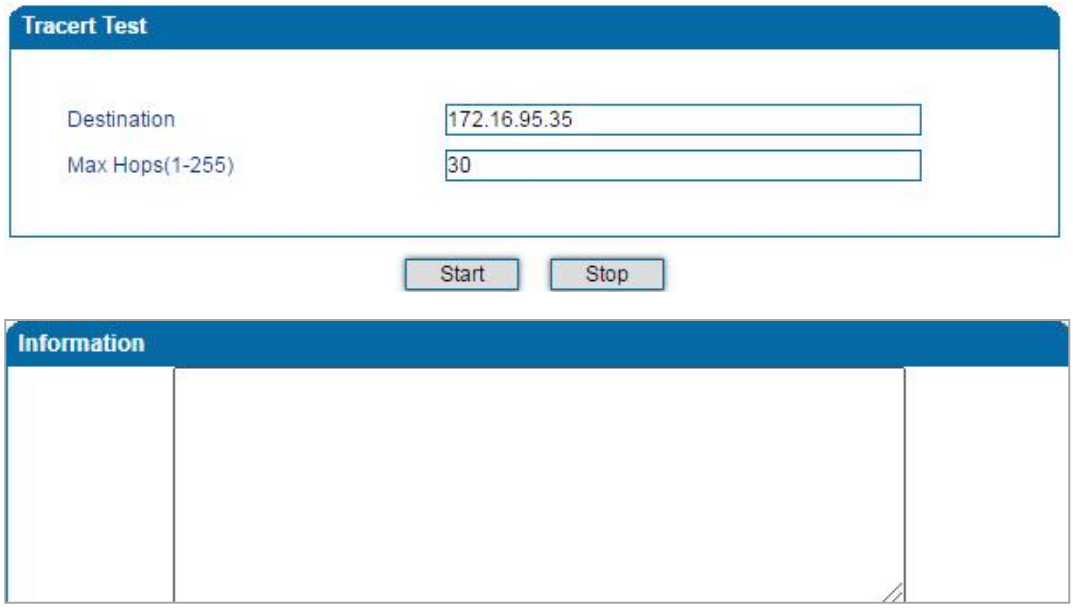

### 4.15.6 Tracert Test

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

Instruction for using Traceroute:

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

Figure 4- 43 Execute Tracert Test




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

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

2. View the route information from the returned message.

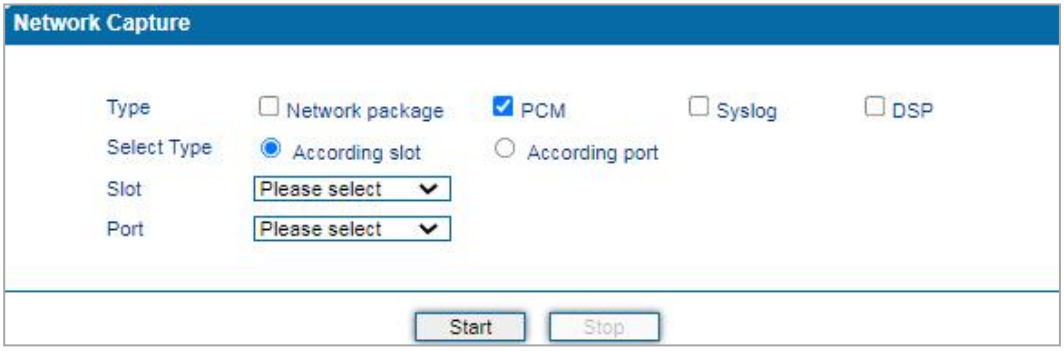
## 4.15.7 Network Capture

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

**PCM Capture:**

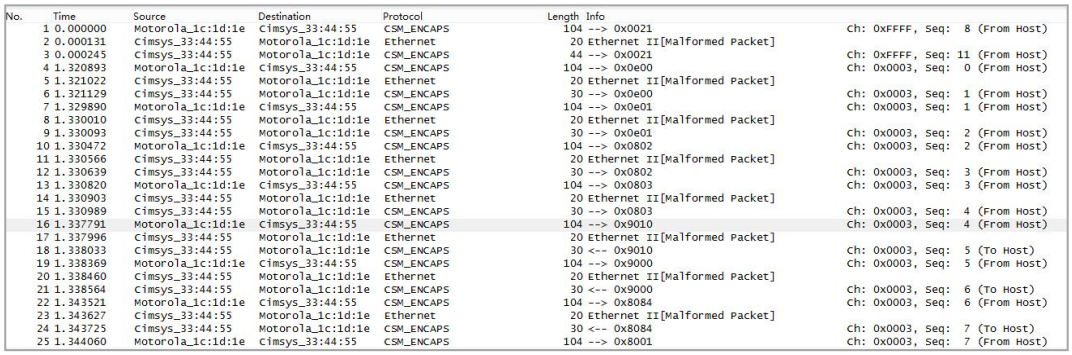
PCM capture helps to analysis voice stream between analog phone and DSP chipset.

Figure 4- 44 Capture PCM Packages




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

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



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

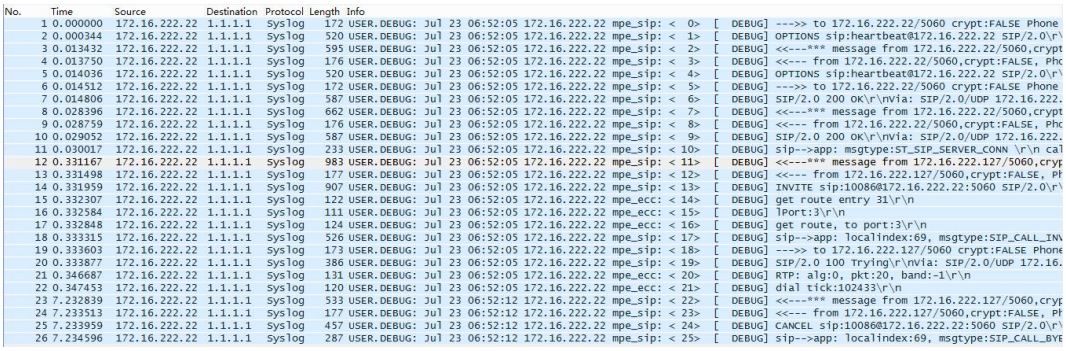
**Syslog Capture:**

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

Figure 4- 45 Capture Syslog Packages

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

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



| No. | Time | Source | Destination | Protocol | Length | Info |
|---|---|---|---|---|---|---|
| 1 | 0.000000 | 172.16.222.22 | 1.1.1.1 | Syslog | 172 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [ DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone |
| 2 | 0.000344 | 172.16.222.22 | 1.1.1.1 | Syslog | 520 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1> [ DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r\ |
| 3 | 0.013432 | 172.16.222.22 | 1.1.1.1 | Syslog | 595 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 2> [ DEBUG] <<---*** message from 172.16.222.22/5060,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:100860172.16.222.22:5060 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:100860172.16.222.22:5060 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 |

**DSP Capture:**

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

Figure 4- 46 Capture DSP Packages

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

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

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

## 4.15.8 Factory Reset

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

Figure 4- 47 Reset Device to Factory Default Setting

## 4.15.9 Device Restart

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

Figure 4- 48 Restart Device

# 5 Glossary

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