Synway SMG-D Series Analog Gateway

SMG1004D

SMG1008D

SMG1016D

SMG1032D

Analog Gateway

User Manual

Version 2.0.0

Synway Information Engineering Co., Ltd

www.synway.net

Content

Copyright Declaration

Revision History

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Chapter 1 Product Introduction

Thank you for choosing Synway SMG-D Series Analog Gateway!

The Synway SMG-D series analog gateway products (hereinafter referred to as ‘SMG-D analog gateway’) are mainly used for connecting traditional phone sets, fax machines and PBXes with the IP telephony network or IP PBX. It provides a powerful, reliable and cost-effective VoIP solution for such occasions as IP call centers and multi-branch agencies.

1.1 Typical Application

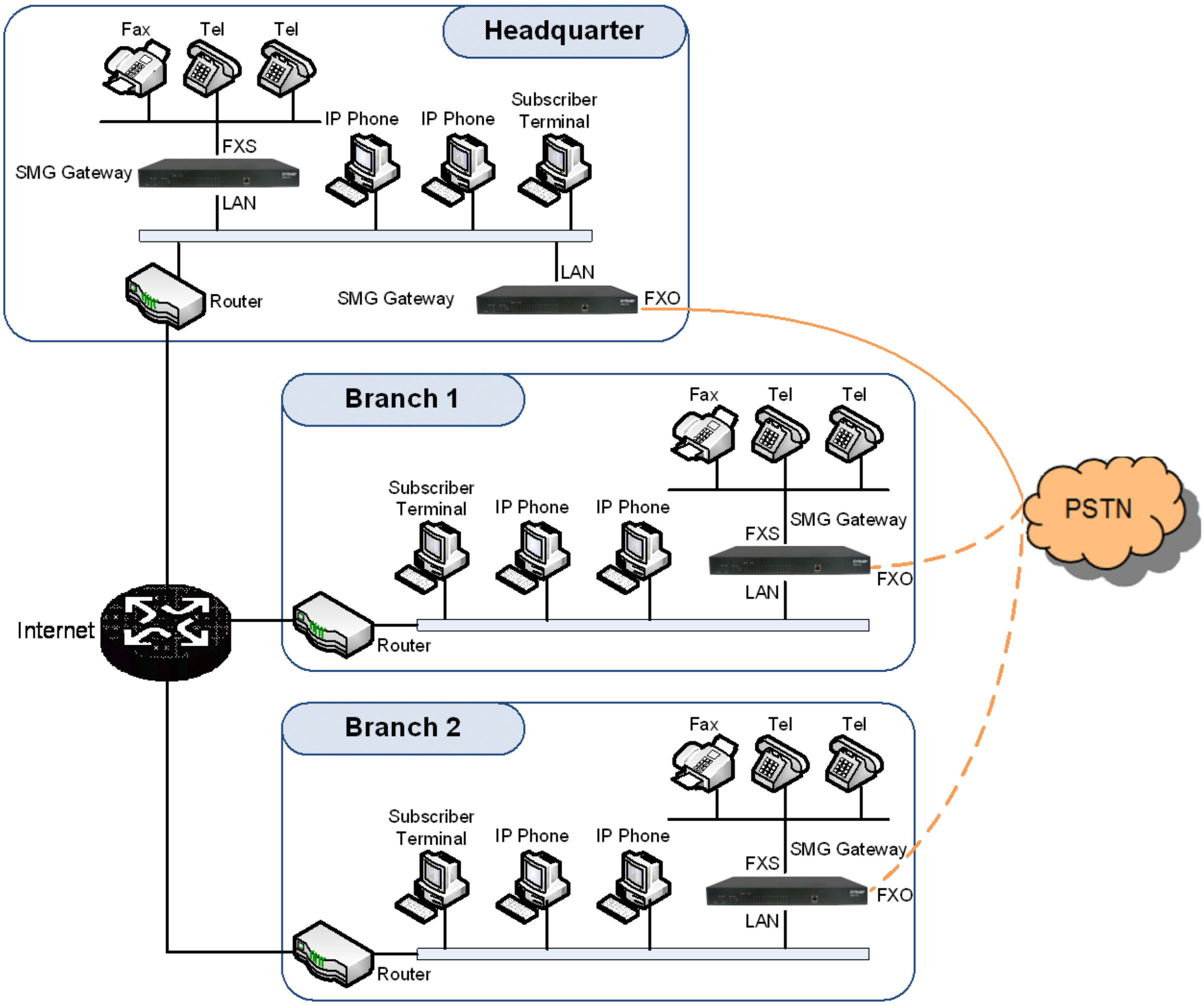


Figure 1-1 Typical Application for SMG-D Series Gateway

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call

	destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy and No Reply.
Call Waiting	When an FXS channel receives another call while it is in conversation, it will have the newly received call keep waiting. Once the current call is finished, the new one will ring the FXS channel and wait for its answer.
Auto Dial	If there is no dialing operation in a designated time period after pickup, the preset auto dial number will be called.
Do Not Disturb	Rejects all the incoming calls to the channel.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the FXS/FXO channel.
TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.
Fax	Provides multiple fax parameters: fax mode, maximum fax rate, fax train mode, error correction mode, etc.
Communication without Network	Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout.
Send Polarity Reversal Signal	Sends the polarity reversal signal to a corresponding FXS channel when the called party pick-up behavior is detected.
Detect Polarity Reversal Signal	Turns a corresponding channel into the talking state when the FXO port detects the polarity reversal signal.
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.
IMS Network	Registers the gateway to a server under IMS network.
SIP Station	Supports a SIP terminal to be registered to the gateway and become a SIP station.
Group Ringing	Rings all the idle FXS ports in a port group.
Ringing by Turns	Rings the FXS ports in a port group by turns according to the *Rule for Ringing by Turns*.
Preemptive Answer	When a channel in a port group is ringing, another channel in the same port group can press the preemptive answer keyboard shortcut to transfer the call from the ringing channel to the current channel.
Centralized Manage	The gateway can register to Synway DCMS and accept the management of the platform.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.
Voice	CODEC: G.711A, G.711U, G.729A/B DTMF Mode: RFC2833, SIP INFO, INBAND
Network	**Description**

Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.
Static IP	IP address modification support.
DHCP	IP address dynamic allocation support.
DNS	Domain Name Service support.
Security	**Description**
Admin Authentication	Supports admin authentication to guarantee the resource and data security.
System Monitor	Monitors the running status of the system and the server.
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface.
Language	Chinese, English.
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.
Tracking Test	Support of Ping and Tracert tests based on WEB.
SysLog Type	Three options available: ERROR, WARNING, INFO, DEBUG.

1.3 Hardware Description

The SMG D-type analog gateway integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 4/8/16/32 voice ports (FXS/FXO) and 2 LANs on the chassis. Each voice port can be configured on demand to serve as an FXS or FXO interface; however, the respective amount of FXS and FXO interfaces must be multiples of 2. The SMG-8D analog gateway adopts an external 12V power supply. See below for product appearance.

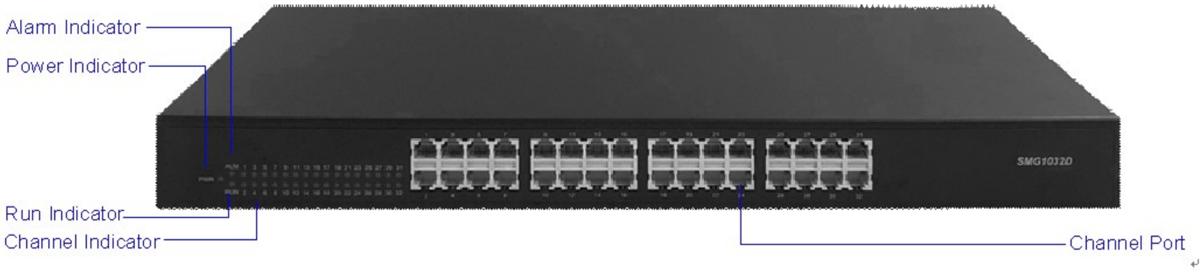


Figure 1-2 O-port Product Front View

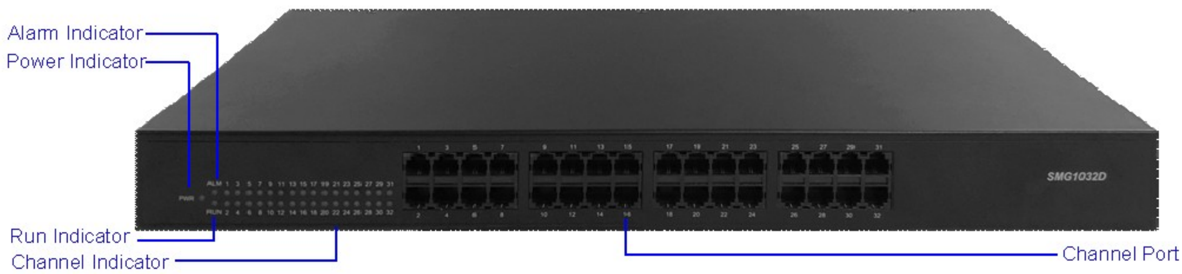


Figure 1-3 S-port Product Front View

Figure 1-4 Rear View

Figure 1-5 Left View

The table below gives a detailed introduction to the interfaces, buttons and LEDs illustrated above:

Interface	Description
LAN	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
FXS/FXO	Amount: 4/8/16/32
	Type: RJ-11
	Maximum Transmission Distance: 1500m
	Charge Mode: Negative Anti-billing Supported
Console Port	Amount: 1
	Type: USB-to-Serial
	Baud Rate: 115200bps
	Connector: MINI USB Connector
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
Button	**Description**
Reset Button	Restore the gateway to factory settings.
LED	**Description**
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
Run Indicator	Indicates the running status. For more details, refer to Alarm Info.
Alarm Indicator	Alarms the device malfunction. For more details, refer to Alarm Info.
Link Indicator	The green LED, indicating the network connection status.

ACT Indicator	The orange LED, whose flashing tells data are being transmitted.
Channel Indicator	FXS and FXO channels are respectively marked by green and red LED after power on. 1. When the channel is idle, the LED Lights up; 2. When the channel is off-hook, the LED flashes slowly; 3. When the channel is ringing, the LED flashes fast.

For other hardware parameters, refer to Appendix A Technical Specifications.

1.4 Alarm Info

The SMG-D analog gateway is equipped with two indicators denoting the system's running status: Run Indicator (green LED) and Alarm Indicator (red LED). The table below explains the states and meanings of the two indicators.

LED	State	Description
Run Indicator	Go out	System is not yet started.
	Light up and flash fast	System is starting.
	Flash slowly	System is normal.
Alarm Indicator	Go out	System is normal.
	Light up	Upon startup: System is normal. In runtime: System is abnormal.
	Flash	System is abnormal.

Note:

- The startup process consists of two stages: System Booting and Gateway Service Startup. The system booting costs about 1 minute and once it succeeds, both the run indicator and the alarm indicator light up. Then after the gateway service is successfully started and the device begins to work normally, the run indicator flashes and the alarm indicator goes out.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Go to Appendix D Technical/sales Support to find the contact way.

Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the SMG analog gateway in the shortest time.

Step 1: Confirm that your packing box contains all the following things.

- SMG Series Analog Gateway *1
- Angle Bracket *2, Rubber Foot Pad *4, Screw for Angle Bracket *8
- 220V Power Cord *1, External 12V Power Adapter *1 for SMG-8D gateway
- Warranty Card *1
- Installation Manual *1

Step 2: Properly fix the SMG analog gateway.

If you do not need to place the gateway on the rack, simply fix the 4 rubber foot pads. Otherwise, you should first fix the 2 angle brackets onto the chassis and then place the chassis on the rack.

Step 3: Connect the power cord.

Make sure the device is well grounded before you connect the power cord. Check if the power socket has the ground wire. If it doesn't, use the grounding stud on the rear panel of the device (See Figure 1-4) for earthing.

Step 4: Connect the network cable.

Step 5: Connect the telephone line. The line from PSTN should be connected to FXO port; the line from station should be connected to FXS port.

These series products provide RJ11 interfaces. You can use a common telephone line directly or construct a telephone line by yourself according to Figure 2-1. Note that only the middle two cores in the RJ11 jack are valid for use.

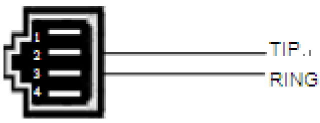


Figure 2-1

Step 6: Log in the gateway.

Enter the original IP address (LAN1: 192.168.1.101) of the SMG analog gateway in the browser to go to the WEB interface of the gateway. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to System Login. We suggest you change the initial username and password via 'System Tools → Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to Change Password. After changing the password, you are required to log in again.

Step 7: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'System Tools → Network' on the WEB interface to put it within your company's LAN. Refer to Network for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 8: Make phone calls.

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration, that is, SIP initiates calls in a point-to-point mode.

Situation 1: Call from a station to another (Tel→Tel)

The gateway allows two FXS ports to call each other by default. Just use a station connected with an FXS port to dial the number of the destination FXS port and you can make a Tel→Tel call. The default number of an FXS port is 80XX, among which XX represents the corresponding port number. For example, the default number corresponding to Port 1 is 8001, and that corresponding to Port 32 is 8032.

Actually a Tel→Tel call on the gateway is accomplished via the routing of Tel→IP→IP→Tel. For detailed introductions and configuration guide, refer to Q2 in Appendix B.

Situation 2: Call from a station to an IP phone (Tel→IP)

1. Go to 'Advanced Settings → Dialing Rule' on the WEB interface and click the 'Add New' button to add a new dialing rule. Refer to Dialing Rule for detailed instructions. Enter either a particular number or a string of 'x's to represent several random numbers. For example, 'xxx' denotes 3 random numbers. You may use the default value of 'Index' and are required not to leave 'Description' empty.

   Example: Set **Index** to **99**, fill in **Description** with **test** and configure **Dial Rule** to **123**.

2. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add FXS ports which are connected with stations to it. Refer to Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the FXS port which is connected with a station is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

3. Go to 'Route Settings → Tel→IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to Tel→IP for detailed instructions. Select the port group created in Step2 as 'Source Port Group' and fill in 'Destination IP' and 'Destination Port' with the IP address and the Port number you plan to call. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the remote IP address intended to call is 192.168.0.111 and the port is 5060. Set **Index** to **63**, **Source Port Group** to **1**, fill in **Description** with **test**, configure **Destination IP** to **192.168.0.111**, **Destination Port** to **5060**, and keep the default values of other configuration items.

4. Pick up the station and dial the number set in Step1 to ring the remote IP phone. If you have set a particular number in Step 1, only this number you can dial; if you have set a string of 'x's, how many 'x's there are, how many random numbers you can dial.

   Example: Pick up the station and dial 123. Then the IP phone with the IP address 192.168.0.111 and the port 5060 will ring.

Situation 3: Call from an IP phone to a station (IP →Tel)

1. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add FXS ports which are connected with stations to it. Refer to Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the FXS port which is connected with a station is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

2. Go to 'Route Settings → IP→Tel' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to IP→Tel for detailed instructions. Fill in 'Source IP' with the IP address which initiates the call and select the port group created in Step1 as 'Destination Port Group'. You may use the default values of other configuration items and required not to leave 'Description' empty.

   Example: Provided the IP address of the IP phone which initiates the call is 192.168.0.111. Set **Index** to **63**, **Destination Port Group** to **1**, fill in **Description** with **test**, configure **Source**

IP to **192.168.0.111**, and keep the default values of other configuration items.

3. Pick up the IP phone and call the IP address and port of the SMG analog gateway to ring the station.

   Example: Provided the IP address of the SMG analog gateway is 192.168.0.101 and the port is 5060, use the IP phone to call the IP address 192.168.0.101 and the station connected with Port1 will ring.

Step 9: Enable the auto dial feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the auto dial feature and set the parameters 'Auto Dial Number' and 'Wait Time before Auto Dial'. If there is no dialing operation in a time period (i.e. Wait Time before Auto Dial) after pickup, the port will automatically call the preset number (i.e. Auto Dial Number). Refer to FXS for detailed instructions.

Step 10: Enable the DND (do not disturb) feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the DND feature. Then, the FXS port will reject all incoming calls. Refer to FXS for detailed instructions.

Step 11: Enable the call waiting feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the call waiting feature. Then the corresponding FXS port while in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. During the time in the waiting state, it will always hear the ringback tone from the FXS port. Refer to FXS for detailed instructions.

Step 12: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation

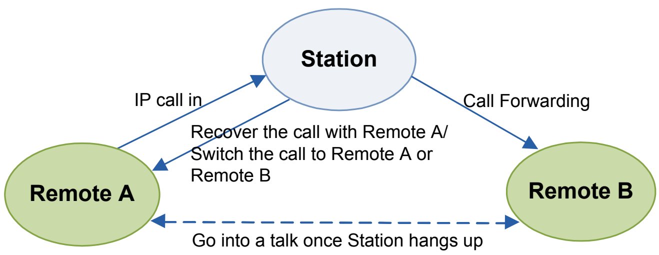


Figure 2-2 Call Forward via Hook-flash

As shown above, Remote A initiates and establishes a call with Station. Then by a hook-flash operation, that is, a rapid clap on the hook or pressing the 'flash' button on the phone set, Station can forward the call to Remote B.

Once a flash is generated, Station will go into the dialing state (the FXS port sends it dialing tones) before it dials the forwarding number.

If the dialing succeeds, the FXS port will send ringback tones to Station. Provided Remote B picks up the call, at this time Station can:

a) Directly talk with Remote B;

b) Perform another hook-flash operation to switch the call to either Remote A or Remote B.

c) Hang up to make Remote A and Remote B go into a direct talk with each other.

If the dialing fails, the FXS port will send busy tones to Station. At this time Station can:

a) Hang up to go back to the ringing state; then pick up the call again to recover the talk with Remote A.

b) Perform the hook-flash operation again without hanging up the call to recover the talk with Remote A.

Once Station recovers the call with Remote A, it can forward the call again by a new hook-flash operation.

Situation 2: Automatic call forward

Go to the port setting interface to enable the automatic call forward feature and fill in a forward number. According to what you set, the SMG analog gateway can automatically forward the incoming calls on three conditions: unconditional, busy, no reply. Note that this feature is applicable only to a single port, but not to a port group consisting of more than one port. Refer to FXS for detailed instructions.

Special Instructions:

- The chassis of the SMG-D analog gateway must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.
- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

Chapter 3 WEB Configuration

3.1 System Login

Type the IP address into the browser and enter the login interface. See Figure 3-1.

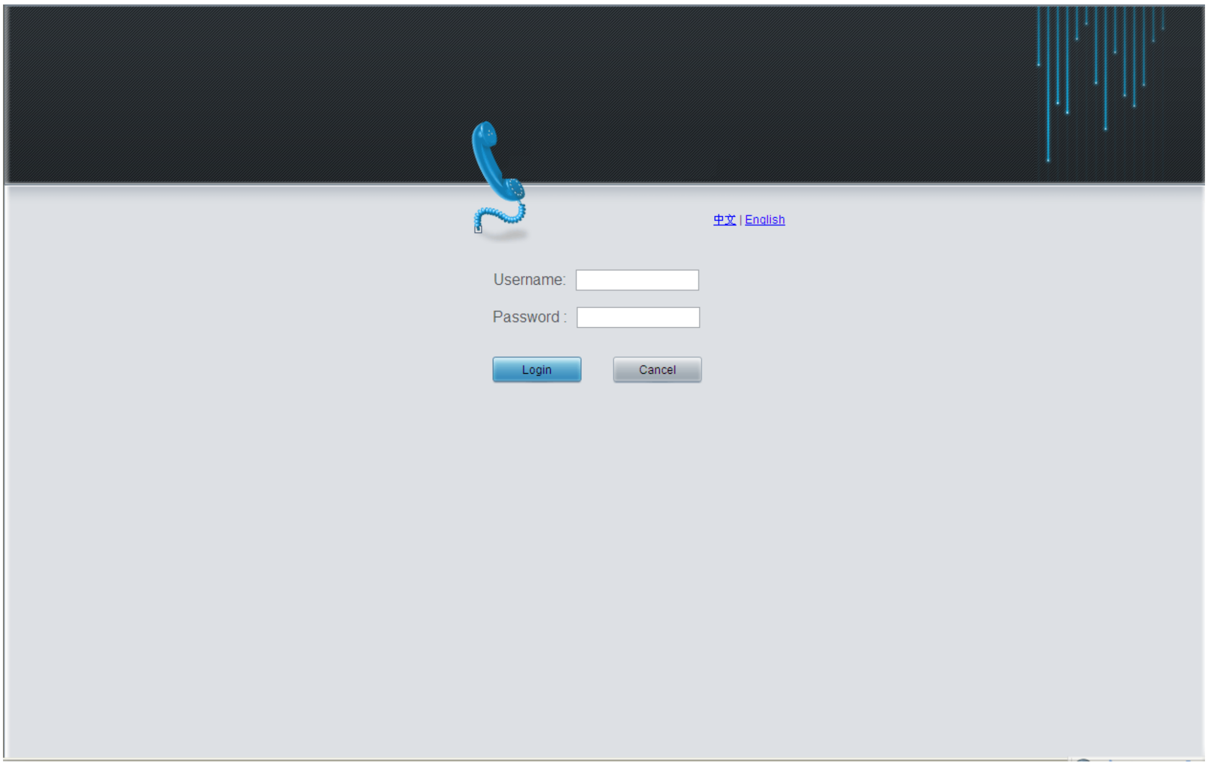


Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both ‘admin’. You can change the username and the password via ‘System Tools → Change Password’ on the WEB interface. For detailed instructions, refer to Change Password.

3.2 Operation Info

Operation Info includes four parts: ***System Info***, ***Channel State***, ***Call Count*** and ***SIP Message Count***, showing the current running status of the gateway.

3.2.1 System Info

On the system info interface, you can click ***Refresh*** to obtain the latest system information. The table below explains the items on the interface.

Item	Description
MAC Address	MAC address of LAN.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN.
DNS Server	DNS server address of LAN.
Receive Packets	The amount of receive packets after the gateway's startup, including three options: All, Error and Drop.
Transmit Packets	The amount of transmit packets after the gateway's startup, including three options: All, Error and Drop.
Current Speed	Show the current speed of data receiving and transmitting.
Work Mode	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.
Runtime	Time of the gateway keeping running normally after startup, which will be automatically updated.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Serial Num	Unique serial number of an SMG-D analog gateway.
U-boot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway.
Firmware	Current version of the firmware on the gateway.
Product Type	The type of current analog gateway.

3.2.2 Channel State

The channel state interface shows the channel type, the voltage and the channel state for each channel on the gateway. The table below explains the items on the interface.

Item	Description
Channel	Channel number on the device.
Type	Type of the channel on the device. If this item shows ---, it means this channel is unavailable, that is, the corresponding module to this channel is not inserted or damaged.
Number	The number corresponding to the port.
Voltage	Line voltage on the channel, calculated by volt (V).
State	Displays the channel state in real time. You can move the mouse onto the channel state icon for detailed state information.

State	Icon	Description
Idle		The channel is available.
Off-hook		The channel picks up the call.
Wait Answer		The channel receives the ringback tone and is waiting for the called party to pick up the phone.
Ringing		The channel is in the ringing state.
Talking		The channel is in a conversation.
Dialing		The channel is dialing.
Pending		The channel is in the pending state.
Internal State		Internal state of the channel.
Unusable		The channel is unavailable.
High Frequency		The upper limit to the frequency of outgoing calls on the FXO channel

Forbid Outgoing Call	Shows if this feature is enabled or disabled.
Direction	Displays the direction of the call on channel.
CallerID	Displays the CallerID of the call on channel.
CalleeID	Displays the CalleeID of the call on channel.
Reg Status	Displays the registration status of the port.
Polarity Reversal Count	The counts of the polarity reversal detected by the FXO port.

3.2.3 Call Count

The call count Interface shows the detailed information about all the calls counted from the startup of the gateway service to the latest open or refresh of this interface. You can click ***Refresh*** to obtain the current call count information. The table below explains the items on the interface.

Item	Description
Call Direction	A condition for call count, two options available: *IP→Tel* and *Tel→IP*.
Total Calls	Total number of calls in a specified call direction.
Successful Calls	Total number of successful calls in conversation.
Busy	Total number of calls which fail as the called party has been occupied and replies a busy message.
No Answer	Total number of calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up.
Call Forward	Total number of calls which have been forwarded.
Routing Failure	Total number of calls which fail because no routing rules are matched.
Dialing Failure	Total number of calls which fail as the called party number does not conform to the dialing rule or due to dialing timeout.
Caller Cancel	Number of calls which the caller canceled before the call was established.
No Resource	Number of calls which fail to establish because the gateway has no idle resources.
Unknown Failure	Total number of calls which fail due to unknown reasons.
Port	FXO port number.

Total Calls in Cycle	Number of FXO outbound calls during the specified period.
Total Call-ins	Total number of calls coming in from the FXO port.
Connected Call-ins	Total number of calls that are incoming from the FXO port and successfully connected.
Call-in Connection Rate	Percentage of connected incoming calls to total incoming calls.
Total Call-in Length	Total call duration for calls coming in from the FXO port.
Total Call-outs	Total number of calls going out from the FXO port.
Connected Call-outs	Total number of calls that are outgoing from the FXO port and successfully connected.
Call-out Connection Rate	Percentage of connected outgoing calls to total outgoing calls.
Total Call-out Length	Total call duration for calls going out from the FXO port.

3.2.4 SIP Message Count

The SIP Message Count interface is used to record the amount of the normal SIP messages that are sent/received or repeatedly sent/received during the period from the startup of the gateway service to the latest open or refresh of the interface. Click ***Refresh*** to refresh the count of SIP messages, or click ***Clear*** to clear the current count of SIP messages.

3.3 Quick Config

Go to the Quick Config interface. Follow the gateway Quick Configuration wizard and you can easily complete the settings on network, SIP and FXS/FXO. The gateway can work normally after configuration.

See the Quick Config-Network Settings interface. Refer to Network for detailed settings. After configuration, click ***Next*** to enter the SIP Settings interface.

See the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to SIP for detailed settings. You are required to fill with the information about the registrar if the gateway must be registered. After configuration, click ***Back*** to go back to the Network Settings interface; click ***Next*** to enter the FXS Settings interface.

See the FXS Settings interface. The configuration items on this interface are the same as those on the FXS interface. Refer to FXS for detailed settings. After configuration, click ***Back*** to go back to the SIP Settings interface; click ***Next*** to enter the FXO Settings interface.

See the FXO Settings Interface. The configuration items on this interface are the same as those on the FXO interface. Refer to FXO for detailed settings. After configuration, click ***Back*** to back to the FXS Settings interface; click ***Next*** to enter the Quick Config-Completion interface.

Click ***Back*** to go back to the FXS Settings interface; click ***Finish*** to finish the Quick Config wizard and now the gateway can work normally with basic configuration.

3.4 VoIP Settings

VoIP Settings includes six parts: ***SIP***, ***SIP Compatibility***, **SIP Station**, **SIP Server**, ***NAT Setting*** and ***Media***. ***SIP Settings*** is used to configure the general SIP parameters, ***SIP Compatibility*** is used to set which SIP servers and SIP messages will the gateway be compatible with, ***SIP Station*** is to set the basic information of the SIP station, ***SIP Server*** is to set the basic information of the SIP server, ***NAT Setting*** is used to configure the parameters for NAT, and ***Media Settings*** is to set the RTP port and the payload type.

3.4.1 SIP

On the SIP settings interface you can configure the general SIP parameters. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. The table below explains the items on the interface.

Item	Description
SIP Address	IP address of SIP signaling, using LAN 1 by default.
SIP Port	Monitoring port of SIP signaling. The value range of it must be greater than 1024 and less than 65535, with the default value of 5060.
Register Status	Registration status of the gateway. When ***Register Gateway*** is set to *No*, the value of this item is *Unregistered*; when ***Register Gateway*** is set to *Yes*, the value of this item is either *Failed* or *Registered*.
Register Gateway	Sets whether to register the gateway as a whole. The default value is *No*. Only when this configuration is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.
SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Authentication Username	Authentication username for registration.
Registrar IP Address	Address of the registry server for the gateway to register.
Registrar Port	Signaling port of the registry server.
Spare Registrar Server	Check the enable checkbox to enable the spare registrar server. By default, it is ***disabled***.
Spare Registrar IP Address	Address of the spare registry server for the gateway to register. The gateway will enable the spare registrar server if the master registrar server has no reply, or the master server is detected with no response in case the item **Detection Server Cycle** is enabled.
Spare Registrar Port	Signaling port of the spare registry server.
Register Interval Time	The registration interval between different ports, in milliseconds, with the value range of 0-1000.
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when ***Register Gateway*** is set to *Yes*. Range of value: 10~3600, calculated by s, with the default value of 600.
Multi-Registrar Server Mode	Tick the checkbox before to enable the multi-registrar server mode. By default, it is *disabled*.
SIP Transport Protocol	There are three modes *UDP*, *TCP* and *TLS* available for running the SIP protocol. The default value is *UDP*.
SRTP	Sets whether to enable SRTP for the gateway to call out.
Switch Signal Port if SIP Registration Failed	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new registration. It will continue until the registration succeeds. The default value is *disabled*.

IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. By default, this feature is *disabled*. Only when this feature is *enabled* will these items **Externally Bound Address**, **Externally Bound Port** and **Authentication Username** be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.

3.4.2 SIP Compatibility

On the SIP Compatibility interface you can configure the SIP parameters to determine which SIP servers and SIP messages will the gateway be compatible with. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

The table below explains the items on the interface.

Item	Description
Obtain CalleeID from	There are two optional ways to obtain the called party number: from "To" Field and from "Request" Field. The default value is "*Request" Field*.
Set CallerID Position	There are two options to set the position of the calling party number: "Displayname of From Field" and "Username of From Field". The default value is "*Username of From Field*".
Obtain CallerID from	There are two optional ways to obtain the calling party number: from "*Displayname of From Field*" and from "*Username of From Field*". The default value is "*Username of From Field*".
Use Source Address	Set whether to send the request message to the source address of the response message from the remote end. By default it is not enabled.
Use Contact Address	Sets whether to send the request message according to the content of Contact, with the default setting of *disabled*. As it is disabled, if the Contact field indicates an IP address within the LAN, the request message will be sent according to the source address; if the Contact field indicates an IP address belonging to the WAN, the request message will be sent according to this IP address.
Call Transfer Mode	There are two optional ways to deal with call transfer: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.
Internal Handle	Sets the internal handle mode for the call transfer, including two options: Match Port Number and Search Idle FXO Channel. The default value is *Match Port Number*.
Call Flash Mode	There are two optional ways to deal with call flash: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.
Hold Music Source	Sets the source of the hold music, with the default value of ***Remote***, This feature gets valid only when you choose the mode *Platform to Handle SIP Info*.
Two Stage Dialing for SIP Incoming Call	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is disabled.
Maximum Wait Answer	Sets the maximum time for the SIP channel to wait for the answer from the

Time	called party of the outgoing call it initiates. If the call is not answered within the specified time period, it will be canceled by the channel automatically. The default value is 60, calculated by s.
SIP Station Supported	Once this feature is enabled, a SIP terminal can be registered to the gateway and becomes a SIP station. By default this feature is disabled.
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is *Gateway*.
Maximum Wait RTP Time	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP packet is received within the specified time period, the channel will enter the pending state automatically and release the call. The default value is *15*, calculated by s.
Call Abnormal Hangup Detection	Sets the interval between checks of the remote end's abnormal hangup, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if this feature is necessary to be used.
Server Status Detection Cycle	The interval of sending a heartbeat packet to detect the master registrar server status, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 15s if this feature is necessary to be used.
Send Cue Tone	Sets whether to send a cue tone once the server gets disconnected, with the default setting of *disabled*.
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is *disabled*.
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Identifier	The identifier field of the VOS encryption, which is used to obtain the key of the SIP encryption.
Key	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is *disabled*.
Ignore ACK	Once this feature is enabled, the gateway is not necessary to wait for the ACK message after sending the 200OK message to establish a call. By default it is *disabled*.
User-defined SIP Code	Once this feature is enabled, you can define a SIP code for the corresponding SIP status, with the default value of *disabled*.
Use Iptables	Once this feature is enabled, only the calls from the SIP registration server, the source IP address of the route IP->TEL and these IP addressed set in Access Control interface are permitted.
Manage Refer	The way to process the Refer message. If you select Default, after the gateway receives the Refer message, it will forward the call to the destination IP address according to the normal process of the refer message. If you select Blind Call Transfer, the gateway will generate a flash signal over the corresponding FXO port and then dial out the call to forward it to the destination terminal via PSTN.
FXO Hangup Time	Sets how soon after finishing a dial will the FXO port hangs up the call, calculated by second, with the default value of 7.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway and becomes a SIP station. Enable the feature of '***SIP Station Supported***' on SIP Compatibility, and you will see the item SIP Station on the VoIP Settings menu. Click '***SIP Station***' to go into the SIP Station interface. By default, there is no available SIP station.

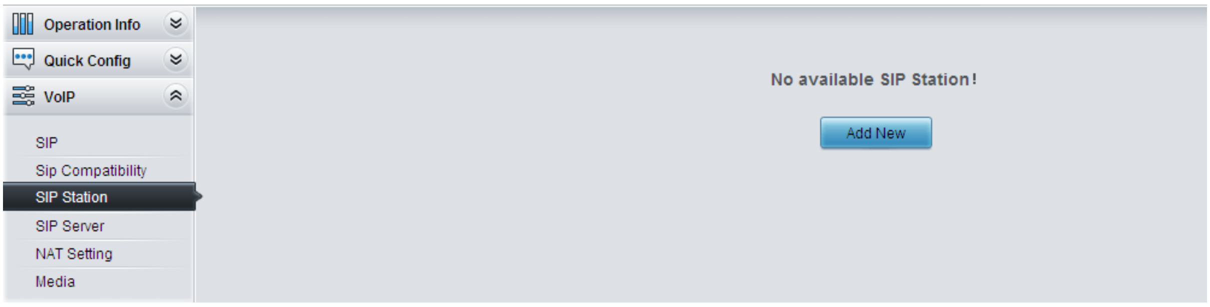


Figure 3-2 SIP Station Setting Interface

Click 

to add SIP stations manually. You can configure basic SIP station information on this interface. The bound port to a SIP station must be an FXO port and unique. The username must be the same as that used to register the SIP terminal to the gateway.

The table below explains the items on the interface:

Item	Description
Number	The logical number for a SIP station to register to the gateway.
Username	The username used to register a SIP station to the gateway.
Password	The password used to register a SIP station to the gateway.
Bound Port	The FXO port which is bound to the SIP station.
Description	It is user-defined, with the default value of *default*.
Batch Setting	Used to set multiple SIP stations at the same time.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings. See Figure 3-3 for the applied SIP station information.

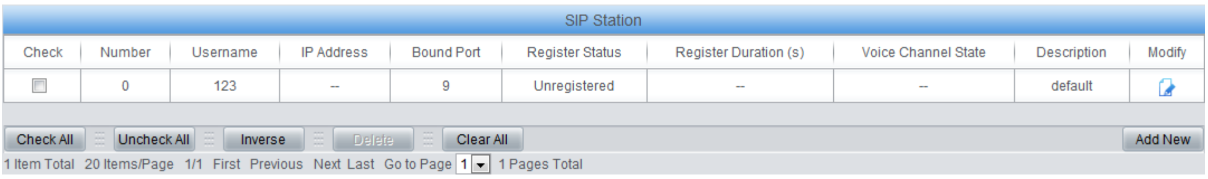


Figure 3-3 SIP Station Interface

Click ***Modify*** in the above figure to modify the configuration of the SIP station. The configuration items on this interface are the same as those on the ***Add New SIP Station*** interface.

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-3 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP stations at a time, click the ***Clear All*** button in Figure 3-3.

3.4.4 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '***Multi-Registrar Server Mode***' on the SIP interface (see SIP) and you will see the item SIP Server under the VoIP Settings menu. Click '***SIP Server***' to go into the SIP Server interface. By default, there is no available SIP server.

Click Add New to add SIP servers manually. You can configure basic SIP server information on this interface.

All the items except Index and Description are the same as those on the SIP interface (SIP).

Item	Description
Index	The index of each SIP server. The gateway supports up to 8 SIP servers.
Description	More information about each SIP server, with the default value of *default*.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Cancel*** to cancel the settings. See Figure 3-4 for the SIP server management interface.

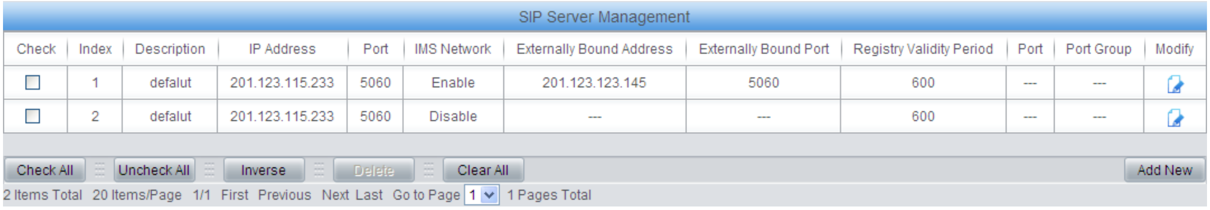


Figure 3-4 SIP Server Management

Click ***Modify*** in the above figure to modify the configuration of the SIP server. The configuration items on this interface are the same as those on the ***Add New SIP Server*** interface.

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-4 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP servers at a time, click the ***Clear All*** button in Figure 3-4.

3.4.5 NAT Setting

On the NAT setting interface you can configure the parameters for NAT. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

The table below explains the items shown on the interface.

Item	Description
Auto Nat	Sets whether to enable the Auto Nat feature. Three options are available: DisableAutoNat, Enable PMP and Enable UPNP, with the default value of *Auto Nat*.
Outer Network Address	The address of the outer network acquired automatically once the PMP or UPNP feature is enabled.
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled.
NAT Type	Detected NAT (Network Address Translation) type. The gateway will return the NAT type automatically in case ***STUN Server*** is enabled. It includes 9 types: unknown; no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric

	NAT with firewall; can't detect over (fail to send detect message) and fail to detect (No reply from the stun server).
STUN Server Address	Address of the server for STUN traversal.
Mapping Contact IP	The IP filled in here will be used in the Contact field of the SIP message.
Mapping SDP IP	The IP filled in here will be used in the SDP field of the SIP message.
Rport	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. The default value is *enabled*.
Learn NAT	When this feature is enabled, the gateway will parse the corresponding address and port in the message returned by Rport so as to use them for the following communication. By default, this feature is *disabled*. **Note:** This feature gets valid only when Rport is enabled.
Auto Detect NAT IP	When this feature is enabled, the gateway will parse the corresponding address and port in the message returned by Rport so as to use them for the following voice communication. By default, this feature is *disabled*. **Note:** This feature gets valid only when Rport and Learn NAT are enabled.
RTP Self-adaption	When this feature is enabled, the RTP reception address or port carried by the signaling message from the remote end, if not consistent with the actual state, will be updated to the actual RTP reception address or port. By default, this feature is *disabled*.

3.4.6 Media

On the media settings interface you can configure the RTP port and payload type depending on your requirements. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. The table below explains the items shown on the interface.

Item	Description
DTMF Transmit Mode	Sets the transmit mode for the IP channel to send DTMF signals. The optional values are *RFC2833*, *In-band* and *Signaling*, with the default value of *RFC2833*.
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.
RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 2000 and the upper limit of 60000 and the difference between larger than 480. The default value is 6000-10000.
JitterMode	Sets the mode for the Jitter buffer, with the default value of *Static Mode*.
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 20~280, calculated by ms, with the default value of 100. **Note:** This is only valid if the Jitter Mode is set to Static Mode.

<table>
<tr><td>Voice Gain Output from IP</td><td>Adjusts the gain of the voice output from IP. Range of value: -24~24, calculated by dB, with the default value of 0.</td></tr>
<tr><td>CODEC Priority</td><td>Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:
<table>
<tr><th>Sub-item</th><th>Description</th></tr>
<tr><td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr>
<tr><td>CODEC</td><td>Three optional CODECs are supported: G711A, G711U, G729A/B.</td></tr>
<tr><td>Packing Time</td><td>Time interval for packing an RTP packet, calculated by ms.</td></tr>
<tr><td>Bit Rate</td><td>The number of thousand bits (excluding the packet header) that are conveyed per second.</td></tr>
</table>
By default, all of the three CODECs are supported and ordered G711A, G711U, G729A/B by priority from high to low.
The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.
<table>
<tr><th>COEDC</th><th>Packing Time (ms)</th><th>Bit Rate (kbps)</th></tr>
<tr><td>G711A</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G711U</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G729A/B</td><td>10 / 20 / 30 / 40 / 60</td><td>8</td></tr>
</table>
</td></tr>
</table>

3.5 Advanced Settings

Advanced Settings includes fifteen parts: ***FXS***, ***FXO***, ***Tone Detector***, ***Tone Generator***, ***DTMF***, ***Ringing Scheme***, ***Fax***, ***Function Key***, ***Dialing Rule***, ***Dialing Timeout***, ***Cue Tone, Color Ring***, ***QoS***, ***Action URL*** and ***AMD***. ***FXS*** is used to configure the general properties of the FXS port; ***FXO*** is used to configure the general properties of the FXO port; ***Tone Detector*** is used to configure some properties of detected tones; ***Tone Detector*** is used to configure some properties of tones sent from gateway; ***DTMF*** is used to set the properties related to DTMF; ***Ringing Scheme*** is used to set the ringing scheme for the FXS port; ***Fax*** is used to configure multiple fax parameters; ***Function Key*** is used to set a cluster of combination keys for you to query a related number; ***Dialing Rule*** and ***Dialing Timeout*** are used to set the judging conditions for dialing; ***Cue Tone*** is used to set the gateway language for playing voice and the voice file used for the two-stage dialing; ***Color Ring*** is used to upload the color ring file which can be set as a ringback tone for an incoming call from IP to FXS port; ***QoS*** uses the differentiated services technology to increase the gateway's service quality. ***Action URL*** is used to designate the server path to report the on-hook or off-hook state of the FXS channel. ***AMD*** is used to detect if a call out from the FXO port is picked up by a man or not.

3.5.1 FXS

The table below explains the items shown on the FXS configuration interface.

Item	Description
Tone Energy	Sets the signal sending energy, with the value range of -35-15 in decibels, and the default value is -11.
Hook-flash Detection	Sets whether to enable the hook-flash detection feature or not, with the default setting of being disabled.

Minimum Time	Time length for judging a flash operation. Only a hook-flash operation which lasts a time more than the value of this configuration item will be regarded as a valid flash operation. Range of value: 80~ ***Maximum Time***, calculated by ms, with the default value of 80.
Maximum Time	Time length for judging a flash operation. Only a hook-flash operation which lasts a time less than the value of this configuration item will be regarded as a valid flash operation. Those lasting a time longer than the value of this configuration item will be regarded as hangup operations. Range of value: 80~2000, calculated by ms, with the default value of 700.
Minimum Time Length of On-hook Detection	The minimum time length for detecting whether the phone is on-hook or not. Range of value: 64~2000, calculated by ms, with the default value of 64. **Note:** This item is valid only when the item Hook-flash Detection is disabled.
Enable Press-key Call-forward	When this function is enabled, you can press a specified key to realize the hook-flash feature. It is disabled by default.
Call-forward Key	Set the specified key used for hook-flash. The default key is #.
Call-forward Method	Sets the way for Press-key call forward, Call Forward with Negotiation or Blind Transfer, and the former is default
CID Transmit Mode	The mode adopted by the FXS port to send the CallerID. The optional values are *FSK* and *DTMF*, with the default value of *FSK*.
Occasion to Send FSK CallerID	Sets when to send the CallerID, before rings or after the 1st Ring. The default value is *after 1st Ring*.
Send Polarity Reversal Signal	Once this feature is enabled, the gateway will send the polarity reversal signal to a corresponding FXS channel when it detects the called party pick-up behavior. By default, this feature is *disabled*.
Off-hook Dither Signal Duration	The minimum duration of the off-hook signal, calculated by milliseconds, must be an integer multiple of 16. The smaller the value is, the more sensitive it is. The default value is 64.
Handling of Call from Internal Station	Sets the handling mode for the calls from station to station, two options available: Internal Handling and Platform Handling, with the default value of Platform Handling.
Light Up Mode for Voice Message	Sets the light up mode for the voice message of the phone, There are two options: *Not Light Up* and *Light Up by FSK*, with the default value of *Not Light Up*.
Open Session In Advance	Sets whether to reply 183 for an incoming FXS call.
Report FXS Status	After the function is enabled, when the status of the FXS channel in the port group changes, the gateway will send the OPTION message carrying the change information to the destination IP address in the TEL->IP route of the port group in real time or to the IP address of the port's registration server if there is no corresponding route. By default this feature is disabled.

After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions.

3.5.2 FXO

The table below explains the particular configuration items for FXO.

Item	Description
Calling Party Detection Time	The maximum waiting time for the detection of the calling party number from FXO port. Range of value: 1~20, calculated by s, with the default value of *10*.
FXO Idle Valid Voltage	Set the threshold voltage value of the FXO port in idle state.
FXO Talk Valid Voltage	Set the threshold voltage value of the FXO port in talking state.
Silence Detection	Used to detect whether the line is silent or not according to the energy threshold and time threshold of silence. FXO will hang up the call automatically if these conditions are satisfied. The default setting is being disabled.
Energy Threshold of Silence	The energy threshold to judge whether the line is silent or not. The signal with the energy less than this set value will be determined to be silence. Range of value: -86~5, calculated by s, with the default value of -34. **Note:** This item will be valid only when Silence Detection is enabled.
Time Threshold of Silence	The time threshold to judge whether the line is silent or not, calculated by s, with the default value of 60. **Note:** This item will be valid only when Silence Detection is enabled.
Rapid Release	Once this feature is enabled, the FXO port will release the source rapidly and go to the idle state when a call from PSTN to soft-terminal via FXO port is rejected by the IP soft-terminal.
FSK Standard	Standard for sending FSK formatted CallerID, which varies in different countries and districts. The optional values are: ETSI (Europe), GR-30 (North America, China) and NIT (Japan), with the default value of *GR-30*.
Reception Interval of DTMF CallerID	The time interval between digits of the DTMF CallerID from FXO port, calculated by ms, with the default value of 250.
Delay for Two Stages Dialing	If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the two-stages dialing process,
Flash Time	Sets the time for generating a flash signal on the analog trunk. Range of value: 32~1000, calculated by ms, with the default value of 100.
Maximum Waiting Time of Dial Tone	Set the maximum waiting time for dial tone detection. The value range is 1~30, calculated by second, and the default value is 5.
FXO Pick-up Delay after INVITE Received at IP Side	Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.
Maximum Wait Answer Time	The maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.
Communication without Network	Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is *disabled*.

Communicate without Network Mode	Sets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of *Auto Search Idle Channel*. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.
Two Stages Dialing Mode	Sets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.
Avoid Being Detected as Flash Signal by PBX	Once this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default value is *enabled*.
Delay after Dial	Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 400~30000, calculated by ms, with the default value of 6000.
Delay to Send 200 OK to IP Side	Once this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is *disabled*.
Open Session In Advance	Once this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is *enabled*.
Priority Rule	Sets the priorities for number attribution and manipulation. The default setting is After Manipulation.
Remove Prefix 0 or Area Code	Sets whether to remove the prefix 0 or the area code from the call number when the CalleeID is a local number. By default it is disabled.
Add Prefix 0	Sets whether to add the prefix 0 to the call number when the CalleeID is not a local number. By default it is disabled.
Local Area Code	Sets the local area code.
High Frequency Call Limit	Sets whether to limit the call frequency for FXO call out. By default it is disabled.
Callout Count Limit	Set the limit on the number of FXO outbound calls. The default value is 0, which means no limit.
Callout Time LImit	Limit the length of the FXO outbound call. The length of the call is randomly set in the specified range, calculated by second. The default range is 0-5 seconds.
Call Type	Sets the call type for call restriction. The default setting is Any Call.
Maximum Call Times	Set the maximum number of allowed outgoing calls during the cycle. The default value is 0, which means no limit.
Cycle Time	Sets the period for call restriction in minutes. The default value is 60 minutes.
Call Type	Set the call type for call restriction. The default setting is Any Call.

After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions.

3.5.3 Tone Detector

On the Tone Parameters setting interface, at most ten pieces of tone parameters are allowed to set. By default, there are already three pieces of tone parameters on the gateway which you can modify or delete according to your actual requirement.

Click ***Modify*** to modify the tone parameter on the tone parameter modification interface.

The table below explains the items shown on the interface.

Item	Description
Index	The unique index of each group of tone detectors.
Tone	There are five options: ***Dial Tone***, ***Busy Tone***, ***Ringback Tone, F1, F2***.
The 1st Mid-frequency	The 1st center frequency. Range of value: 200~3500, calculated by Hz. The default value is 450.
The 2nd Mid-frequency	The 2nd center frequency. Range of value: 0 or 200~3500, calculated by Hz. The default value is 0.
Duration at ON State	The duration of tones at on state. The default setting: Dial Tone is 1500ms, Busy Tone is 350ms, Ringback Tone is 1000ms.
Duration at OFF State	The duration of tones at off state. The default setting: Dial Tone is 0ms, Busy Tone is 350ms, Ringback Tone is 4000ms.
Period Count	Sets the count of periods as the condition to determine a periodic tone. The default setting: Dial Tone is 0, Busy Tone is 2, Ringback Tone is 1.
Duration Error at ON/OFF State	Sets the duration error at ON/OFF state, calculated by ms, with the default value of 20.

To delete a piece of tone, check the checkbox before the corresponding index and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all tone at a time, click the ***Clear All*** button.

3.5.4 Tone Generator

By default, there are four tones on the Tone Generator Setting interface: Dial Tone—a single tone with 450HZ frequency, plays continuously; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. You can configure the tone generator manually.

3.5.5 DTMF

The DTMF configuration include two parts: DTMF Detector and DTMF Generator. The table below explains the items shown on the interface.

Item	Description
Energy Difference of High-freq minus Low-freq	Set the allowed difference of the high frequency energy in the DTMF signal over the low frequency energy. The value range is 0~24, in decibel, and the default value is 5.
Energy Difference of Low-freq minus High-freq	Set the allowed difference of the low frequency energy in the DTMF signal over the high frequency energy. The value range is 0~24, in decibel, and the default value is 9.
Minimum Duration at ON	Set the minimum duration at ON for the DTMF signal. Range of value: 10～2000, calculated by ms. The default value is 28.

Minimum Duration at OFF	Set the minimum duration at OFF for the DTMF signal. Range of value: 10～2000, calculated by ms. The default value is 36.
Ratio of DT Energy	Set the percentage of energy in the DTMF signal. The value range is 1-100 and the default value is 83.8.
Lowest Energy Threshold	Set the minimum energy threshold of the DTMF signal. Range of value: -40~9. The default value is -21.
DTMF Display via Channels Status	Once this feature is enabled, the received/sent DTMF will be displayed upon you putting the mouse on the icon of channel status. The default value is disabled.
ABCD Detection	Once this feature is enabled, the gateway can detect the DTMF digits A, B, C and D (Case-insensitive). The default value is disabled.
DTMF Energy Advance Set	When this function is enabled, different high frequency energy and low frequency energy can be configured for different DTMFs. By default the feature is disabled.
DTMF Energy	Energy of the DTMF signal sent by the FXS gateway. Range of value: -18~11, calculated by dB, with the default value of 0.
Duration at ON	Set the duration of the DTMF signal at ON state. Range of value: 0~16383, calculated by ms, with the default value of 100.
Duration at OFF	Set the duration of the DTMF signal at OFF state. Range of value: 0~16383, calculated by ms, with the default value of 32.

After configuration, click ***Save*** to save your settings into the gateway. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. Click ***Reset*** to restore the configurations.

3.5.6 Ringing Scheme

On the Ringing Scheme Configuration interface, the gateway can execute different ringing schemes according to the CallerID or Alert-Info..

The table below explains the items shown on the interface.

Item	Description
CallerID	The gateway will match the CallerID set in this item to that of the incoming call. If they are matched, the current ringing scheme will be executed; otherwise, the default ringing scheme (1 sec on and 4 sec off) will work. The rule to fill in the CallerID is the same as that of Dialing Rule. Multiple CallerIDs are supported; they should be separated by ","
Alert-Info Value	The gateway will match the alert-info value set in this item to that of the incoming call. If they are matched, the current ringing scheme will be executed; otherwise, the default ringing scheme (1 sec on and 4 sec off) will work..
Ringing Scheme	The ringing scheme can be "1,X,Y" or "2,X,Y,M,N", in which, the number 1 or 2 denotes one group or two groups; X, M denote the duration at on state while Y, N denote the duration at off state. Note: The duration at ON or OFF cannot be greater than 12000ms, the total duration at ON and OFF cannot be greater than 16000ms, and N - the last duration at OFF cannot be less than 1800ms if the item "Occasion to Send FSK CallerID" is set to After the first ring.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations.

3.5.7 Fax

The table below explains the items shown on the fax configuration interface.

Item	Description
Fax Mode	The real-time IP fax mode. The optional values are *T.38* and *Disable*, and the default value is *Disable* which means to disable T.38.

Under the T.38 mode, users can configure the general fax parameters via this interface. After configuration, click ***Save*** to save your settings into the gateway. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. Click ***Reset*** to restore the configurations. The table below explains the configuration items on the interface.

Item	Description
T38 Fax Port	The port for T.38 faxing, set to *Use New Voice Port* by default.
T38 Version	Version of T.38 which is defined by ITU-T, including 1, 2, 3, 4.
T38 Negotiation	The Negotiation mode of T.38, providing two options: *I*nitiate Negotiation as Fax Sender and Initiate Negotiation as Fax Receiver. The default value is *Initiate Negotiation as Fax Receiver*.
Maximum Fax Rate	Sets the maximum faxing rate for both receiving and transmitting, with the default value of 9600, calculated by bps.
Fax Train Mode	Sets the train mode for T.38 fax, with the default value of *transferredTCF*.
Error Correction Mode	Sets the error correction mode for T.38 fax. The optional values are *t38UDPRedundancy* (Redundancy Error Correction) and *t38UDPFEC* (Forward Error Correction), with the default value of *t38UDPRedundancy*.
T.30 ECM	Sets whether to enable T.30 ECM.

3.5.8 Function Key

On the Function Key Configuration interface you can set a cluster of combination keys to query a related number.

Click “Enable” to enable the corresponding function key. The gateway will use the default function keys when the mode is set to default; and it will allow you to set new function keys when the mode is set to user-defined. Click ***Save*** to save your settings into the gateway.

Note: Phone Test is used just to see if the phone can work normally. It requires you to hang up the phone after dialing the corresponding combination keys. Then the gateway will ring the phone. At that time, pick up the phone and you can hear the voice prompt played by the gateway (e.g. ‘Test successful.’)

When the **Blind Transfer** feature is enabled, set a corresponding function key in the box behind. After you clap the hook switch rapidly, dial the set function key for **Blind Transfer** and then the called party number. Hang up the call once hearing the howler tone and the subsequent call procedure will go out of your control.

To perform a multi-party conferencing, you should first go to ‘Advanced’ ->‘FXS’ to enable the Hook-flash Detection feature. Then refer to the following example to enter the conference:

1. 8001 dials 8002 and 8002 answers;

2. 8001 claps the hook switch and hears the dialing tone;
3. 8001 dials 8003 and 8003 answers;
4. 8001 dials the corresponding function key *070* to enter the third-party conference.

3.5.9 Dialing Rule

Considering efficiency, it is not acceptable that the gateway reports to the PBX or relevant devices every time it receives a number. Instead, we hope that the gateway can automatically judge the received number to see if it meets the set rule, if it is complete and if it is qualified to make outgoing calls. Therefore, a whole dialing plan, which consists of multiple dialing rules specifying the auto judging conditions, is required. Each dialing rule has a priority, which is used to restrict the sequence and avoid conflict.

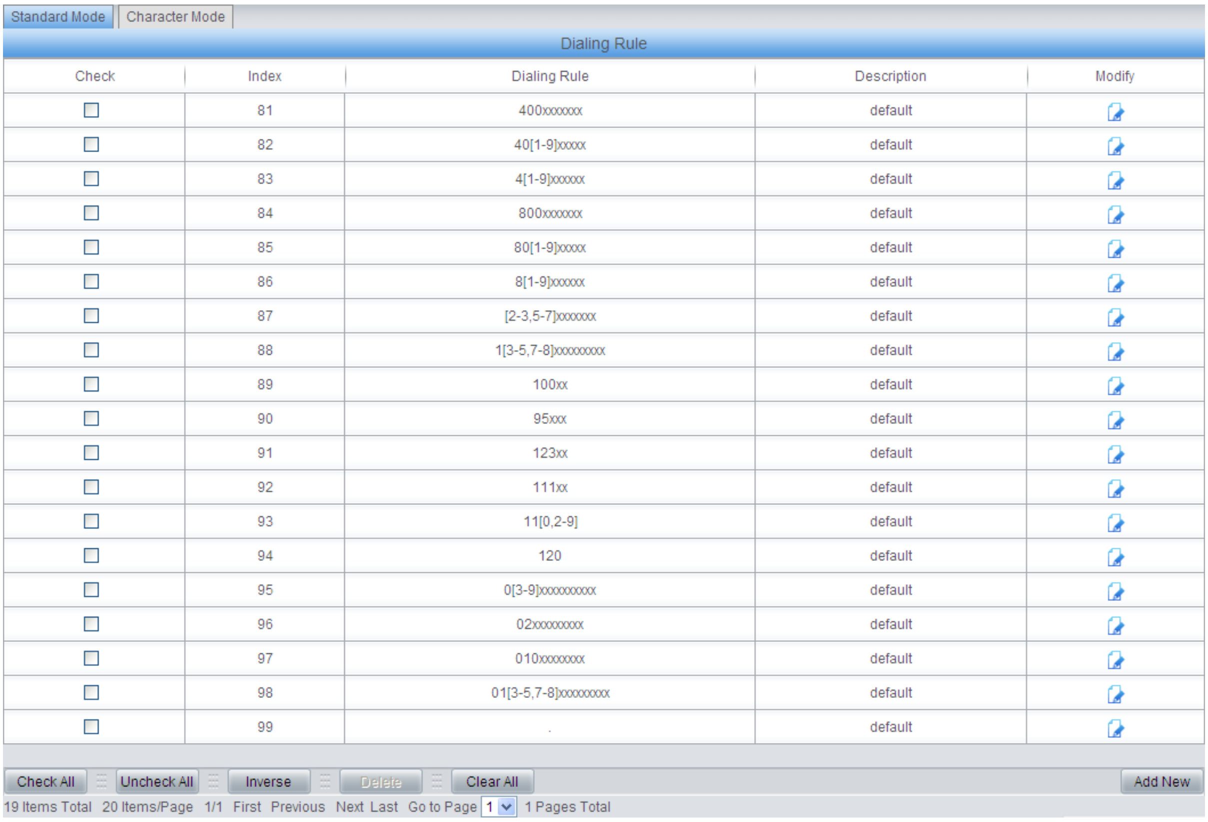


Figure 3-5 Dialing Rule Configuration Interface (Standard)

See Figure 3-5 for the Dialing Rule Configuration interface under the standard mode. The list in the above figure shows the dialing rules with their priorities and description, which can be added by the ***Add New*** button on the bottom right corner.

The table below explains the items on the dialing rule adding interface.

Item	Description
Index	The unique index of each dialing rule, which denotes its priority. A dialing rule with a smaller index value has a higher priority and will be checked earlier while matching.
Description	Remarks for the dialing rule. It can be any information, but can not be left empty.
Dialing Rule	Up to 100 dialing rules can be configured in the gateway, and the maximum length of each dialing rule is 127 characters. See below for the meaning of each character in the dialing rule. The gateway will do instant matching for your dialing number based

on the dialing rule and regard your dialing as finished upon receiving ‘#’ or dialing timeout.

Character	Description
“0”~”9”	Digits 0～9.
“A”~”D”	Letters A～D.
“x”	A random number. A string of ‘x’s represents several random numbers. For example, ‘xxx’ denotes 3 random numbers.
“.”	‘.’ indicates a random amount (including zero) of characters after it.
“[ ]”	‘[ ]’ is used to define the range for a number. Values within it only can be digits ‘0~9’, punctuations ‘-‘ and ‘,’. For example, [1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.
“-”	’-’ is used only in ‘[ ]’ between two numbers to indicates any number between these two numbers.
“,”	’,’ is used to separate numbers or number ranges, representing alternatives.
“*”	Only represents symbol ”*”.
“#”	Only set it at the beginning of the string, representing symbol “#”.

There are 19 dialing rules already configured on the gateway for easy use. See below for detailed information.

Priority	Dialing Rule	Description
99	.	Any number in any length.
98	01[3-5,7-8]xxxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
97	010xxxxxxxx	Any 11-digit number starting with 010
96	02xxxxxxxxx	Any 11-digit number starting with 02
95	0[3-9]xxxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
94	120	Number 120。
93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
92	111xx	Any 5-digit number starting with 111
91	123xx	Any 5-digit number starting with 123
90	95xxx	Any 5-digit number starting with 95
89	100□x	Any 5-digit number starting with 100
88	1[3-5,7-8]xxxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7
86	8[1-9]xxxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89

	85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805,.806, 807, 808 or 809
	84	800xxxxxxx	Any 10-digit number starting with 800
	83	4[1-9]xxxxxx	Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
	82	40[1-9]xxxxx	Any 8-digit number starting with 401, 402, 403, 404, 405, 406, 407, 408 or 409
	81	400xxxxxxx	Any 10-digit number starting with 400

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-5 to modify the dialing rules. The configuration items on the dialing rule modification interface are the same as those on the ***Add New Dialing Rule*** interface.

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-5 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all dialing rules at a time, click the ***Clear All*** button in Figure 3-5.

Under the Character mode, you can edit the dialing rule list to add a new one or modify an old one. The exact meaning of each rule element is described on the page.

3.5.10 Dialing Timeout

The table below explains the items shown on the dialing timeout info interface.

Item	Description
Inter Digit Timeout	Sets the largest interval between two digits of a dialing number. Range of value: 1~30, calculated by s, with the default value of 6. In case your dialing rules do not include "**.**", the call will fail if there is no digit dialed or no dialing rule matched during this interval; in case your dialing rules include "**.**", the gateway will wait until this interval ends and match to the dialing rule "**.**" if there is no digit dialed or no other dialing rule matched during this interval.
Off-hook Waiting Keypress Timeout	Sets the maximum time to wait for keypress after the FXS port picks up the phone. The value range is 1~30, in seconds, and the default value is 6.
Description	More information about the configuration item ***Inter Digit Timeout***, such as the reason for adopting the current value.

Click ***Modify*** on the interface to modify the dialing timeout info. The configuration items on the dialing timeout info modification interface are the same as those on the ***Dialing Timeout Info Interface***.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings.

3.5.11 Cue Tone

The table below explains the items on the Cue Tone interface.

Item	Description
Upload a file of cue	Uploads a user-defined cue tone file to the gateway.

tone	

Click ***Save*** to save the above settings into the gateway.

3.5.12 Color Ring

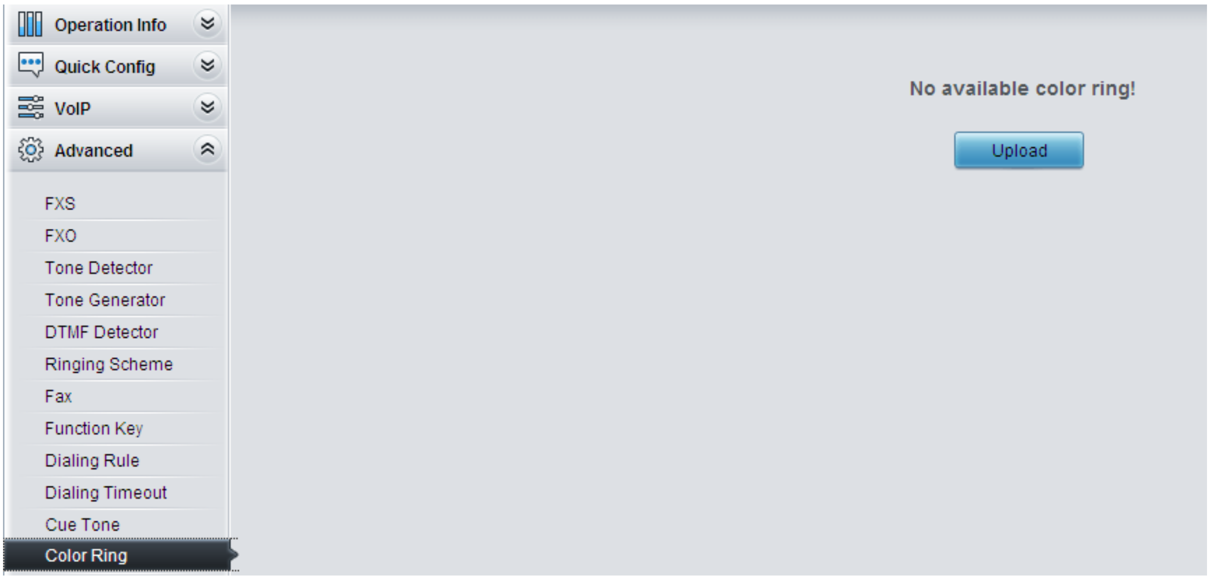


Figure 3-6 Coloring Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-6. Click ***Upload*** to upload a new color ring manually. The table below explains the items on the Color Ring Upload interface.

Item	Description
Index	The unique index of each color ring to be uploaded.
Description	It is user-defined, with the default value of *default*.
Color Ring	The file of the color Ring to be uploaded.

After configuration, click ***Upload*** to upload the color ring file to the gateway or click ***Return*** to cancel the upload.

Click ***Modify*** to modify the configuration of the color ring. The configuration items on the color ring modification interface are the same as those on the ***Color Ring Upload*** interface.

To delete a color ring, check the checkbox before the corresponding index and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all color rings at a time, click the ***Clear All*** button.

3.5.13 QoS

Via the Differentiated Services setting interface, the gateway can meet various application requirements under a limited bandwidth and ensure neither delay nor discard for important services so as to improve its quality of services.

The table below explains the items shown on the interface.

Item	Description
QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.

Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 46.
Control Premium QoS	Sets the priority of the control premium for QoS. A control premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 26.

3.5.14 Action URL

The Action URL interface is used to designate the server patch to report the on-hook or off-hook state of the FXS channel. You are allowed to designate two different server paths. After setting, the state will be reported to the designated server once any of the FXS channel hangs up or picks up a call. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

3.5.15 CDR Query

See the table below for the configuration items on the CDR interface.

Item	Description
Starting Date, Ending Date	Sets the starting and ending dates for CDR query.
Port	Sets the port on which CDR query will proceed.
Call Direction	Sets the call direction for CDR query.
CallerID, CalleeID	Sets the CallerID/CalleeID for CDR query.
Call Duration	Sets the minimum/maximum call duration for CDR query. Only the CDRs within the set call duration will be inquired.
Keyword	Sets the keyword to be filtered in querying. Only the CDRs containing the keyword will be inquired.

Click ***Query*** to query the CDR information based on the set conditions.

Note: This page will appear only when the CDR feature is enabled and saved to local (set in System Param).

3.5.16 AMD

The AMD Configuration interface is used to set the parameters for judging whether the phone is picked up by a man or not. See the table below for details.

Item	Description
AMD Detection for Outgoing Call	Sets whether to enable the AMD detection while making an outgoing call, with the default value of *Disabled*.
Line Silence Overtime after Dial Tone	Judges if the line silence after dial tone lasts overtime or not, calculated by ms, with the default value of *30000*.
Silence Overtime after Tone or Color Ring Being Detected	Judges if the silence after tone or color ring lasts overtime or not, calculated by ms, with the default value of *15000*.

Overtime for a Complete AMD Detecting Process	Judges the whole AMD detecting process overtime or not, calculated by ms, with the default value of *70000*.
Upper Limit of Detected Continuous Tones	Judges if the tone detected time is overtime or not.
Shortest Voice Duration at ON State	Sets the shortest duration when the voice goes into the High voltage state, calculated by ms, with the default value of *80*.
Shortest Voice Duration at OFF State	Sets the shortest duration when the voice goes into the low voltage state, calculated by ms, with the default value of *400*.
Maximum Greeting Duration at OFF State	Sets the longest duration of the greetings at the OFF state after a call is picked up by a man, calculated by ms, with the default value of *0*.
Shortest Silence Duration before Greeting	Sets the shortest silence duration before the phone is picked up by a man, calculated by ms, with the default value of *600*.
Shortest Greeting Duration	Sets the shortest greeting duration in case the phone is picked up by a man, calculated by ms, with the default value of *80*.
Maximum Greeting Duration	Sets the longest greeting duration in case the phone is picked up by a man, calculated by ms, with the default value of *1200*.
Shortest Silence Duration after Greeting	Sets the shortest silence duration after the phone is picked up by a man, calculated by ms, with the default value of *1200*.
Silence Energy Threshold	Sets an energy value that can judge the voice is silence or not, calculated by ms, with the default value of *500*.
Energy Difference Proportion of Tone	Sets the difference proportion of the high and low energies in the signal, with the default value of *30.*
Output AMD Debugging Info to Syslog	Sets whether to output the AMD debugging information to Syslog.

If this feature is enabled, the gateway will automatically activate the feature of 200OK Delay (setting global_delaysend200oktime to the default value of 15). Once a call is dialed out from the FXO port, the gateway will detect if the call is picked up by a man or not. If it is picked up by a man, the FXO port will go into the talk state immediately; otherwise, the FXO port will not go into the talk state until the set time of 200OK Delay is over. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

3.5.17 VPN

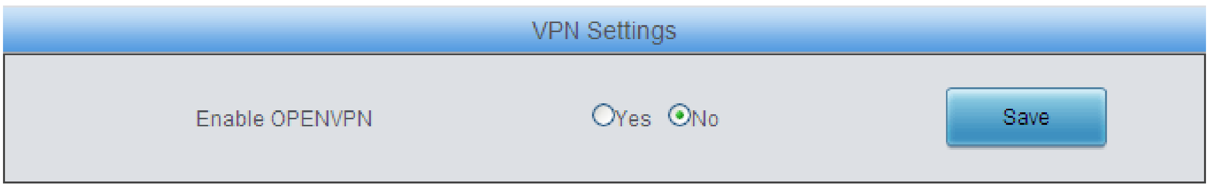


Figure 3-7 VPN Settings Interface

Thanks to the embedded VPN Client, the analog gateway can access the VPN network via OPENVPN directly, not requiring extra VPN client, which simplifies the network deployment. Meanwhile, the design of both SIP signaling messages and voice streams transporting via VPN avoids possible problems induced by the SIP protocol in passing through the firewall and NAT.

See Figure 3-9 for the VPN Settings interface. The table below gives the explanation to the items shown in the above figure.

Item	Description
Enable OPENVPN	Sets whether to enable the VPN feature, with the default value of *No*. If this feature is enabled, the gateway will work as a VPN client.

You are required to upload the VPN certificate after enabling the VPN feature. See Figure 3-10.

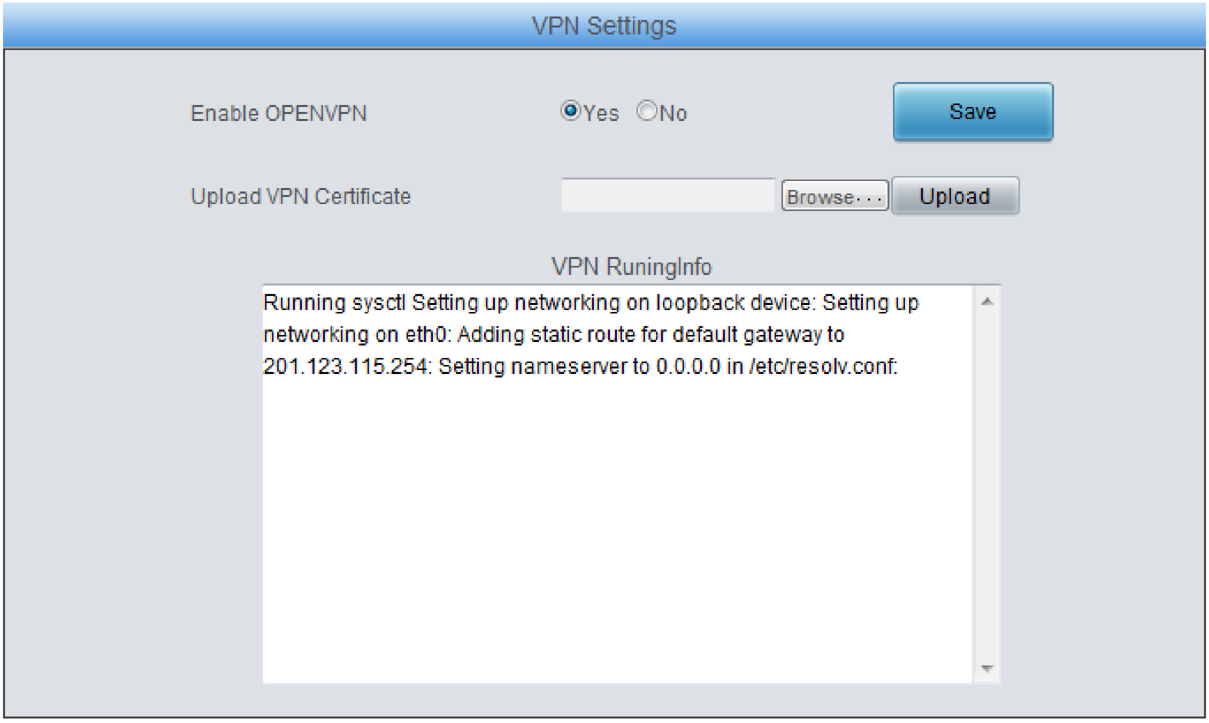


Figure 3-8 VPN Certificate Upload Interface

Note: Refer to Appendix C About VPN for how to make a VPN certificate.

3.5.18 Area Selection

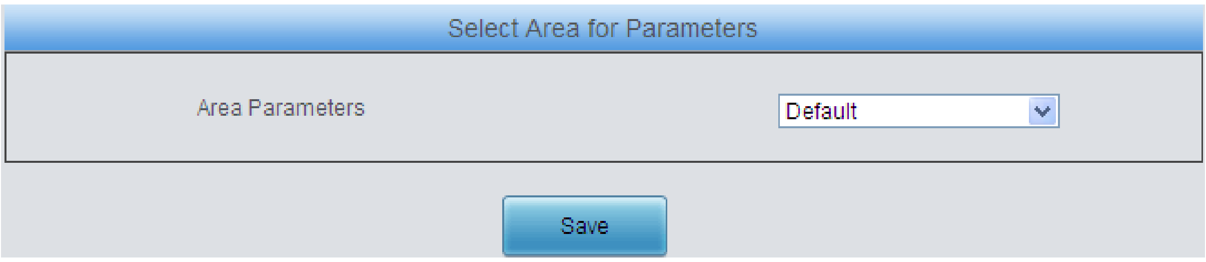


Figure 3-9 Area Selection Interface

See the table below for the configuration items on the Area Select interface.

Item	Description
Area Parameters	When Australia is selected, the gateway will automatically set such parameters as tone, ringing, feed, impedance to those applicable for Australia.

You are required to upload the VPN certificate after enabling the VPN feature. See Figure 3-10.

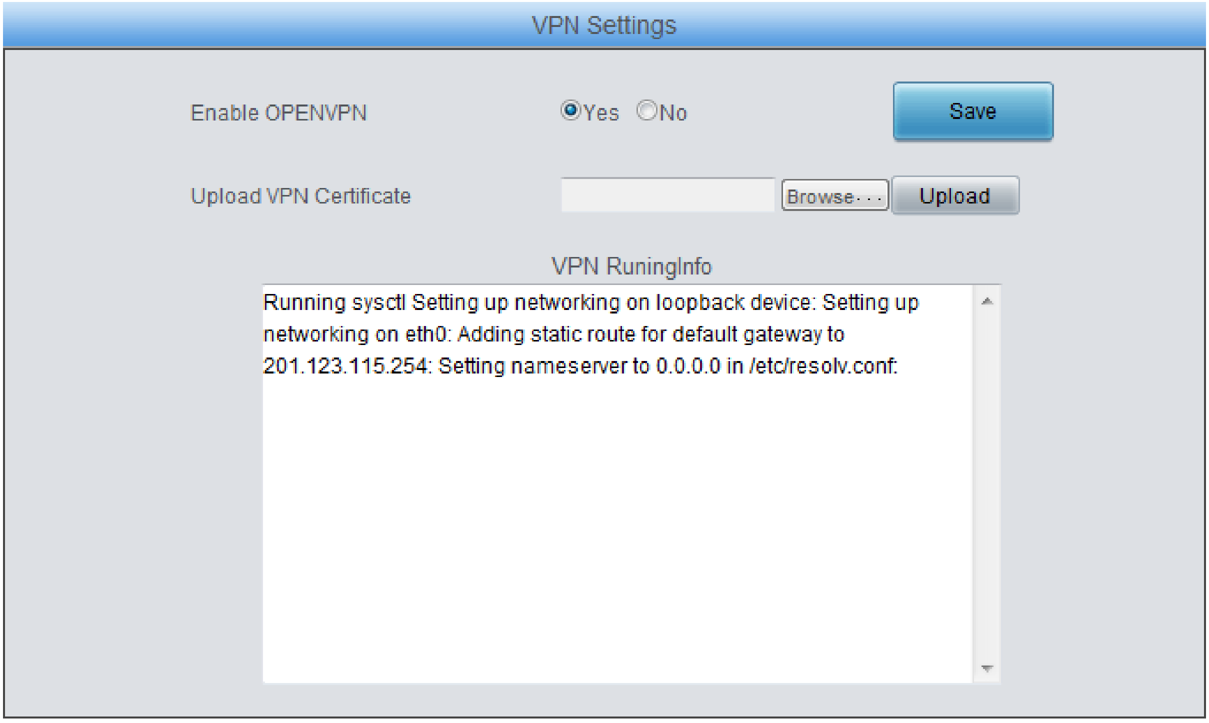


Figure 3-10 VPN Certificate Upload Interface

Note: Refer to Appendix C About VPN for how to make a VPN certificate.

3.6 User Management

Clicking the button Add New, you can configure the added user in the pop-up window. The configuration information includes username, password, user authorities and check channel status.

When the user authority is set to read-only, the corresponding user's FXS/FXO port setting interface has no modification button, and the FXS/FXO port settings cannot be changed. When the user authority is set to Read and Write, the FXS/FXO port settings can be changed.

The block Check Channel Status is used to configure the range of ports that the corresponding user can view or modify.

3.7 Port Settings

Port Settings includes six parts: ***FXS***, ***FXO***, ***FXO Port Timer***, ***FXO List Timer***, ***Port Group*** and ***Advanced FXO Settings***.

3.7.1 FXS

The list on the FXS settings interface shows the feature and properties of each FXS port. Click ***Modify*** to modify the properties of the corresponding port.

The table below explains the configuration items on the FXS modification interface.

Item	Description
Port	Serial number of the FXS port on the device.
Type	Type of the port on the device (FXS). This item is not configurable.

Register Port	Sets whether to register the port to the SIP server. When this item is set to *No*, the item ***Reg Status*** on the FXS settings interface shows *Unregistered*; when this item is set to *Yes*, the item ***Reg Status*** shows *Failed* or *Registered*.
SIP Account	When the port initiates a call to SIP, this item corresponds to the username of SIP. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 8 is 8008.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port. To register a port to the SIP server, both items ***SIP Account*** and ***Password*** must be filled in.
Display Name Preferred	In case this feature is enabled and the port group or the whole gateway is registered, if the display name set by the port are different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the sent SIP message will be the display name set by the port group; if the whole gateway is registered, the displayname in the sent SIP message will be the displayname of the gateway.
Server Index	The index of the SIP server which will be quoted by the current FXS port.
Auto Dial Number, Wait Time before Auto Dial	The FXS port will dial the ***Auto Dial Number*** if there is no dialing operation after pickup within a designated time period (i.e. ***Wait Time before Auto Dial***).
Input Gain, Output Gain	Adjusts the gain of the voice input to/ output from the FXS port. Range of value: -6~6, calculated by dB, with the default value of 0.
Echo Canceller	The echo cancellation feature for a call conversation over the FXS channel. By default, this feature is enabled and the effect can reach 128ms.
Forbid Outgoing Call	If this feature is enabled, the FXS port will be forbidden to call out. The default setting is *disabled*.
CID	CallerID. If this feature is enabled, the FXS port will send the CallerID of the incoming IP call together with the ringing tone to the corresponding station. The default setting is *enabled*. CallerID displays digits only and will filter out any other characters if exist.
Call Waiting	If this feature is enabled, the FXS port in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. The default setting is *disabled*.
DND	Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 message to reject all incoming calls. The default setting is *disabled*.
Call Forward	The automatic call forward feature for the FXS port. Once this feature is enabled, the FXS port will forward incoming IP calls according to ***FWD Type***. Note: To enable this feature, do not put the FXS port into a port group with other ports. The default setting is *disabled*.

<table>
<tr><td>FWD Type</td><td>Forward conditions for the FXS port to forward incoming IP calls. The optional values are:
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Unconditional</td><td>The FXS port will forward all incoming IP calls to the preset FWD Num immediately when it receives them.</td></tr>
<tr><td>Busy</td><td>The FXS port will forward incoming IP calls to the preset FWD Num if it is busy upon receiving them.</td></tr>
<tr><td>No Reply</td><td>The FXS port will forward incoming IP calls to the preset FWD Num if the corresponding station does not answer them in a designated time period (i.e. Time for No Reply Forward). Only when this forward condition is selected does the configuration item Time for No Reply Forward become valid.</td></tr>
</table>
This item is valid only when Call Forward is set to Enable.</td></tr>
<tr><td>FWD Num</td><td>The number to which the incoming IP call is forwarded. If the Call Forward feature is enabled, this item can not be left empty.</td></tr>
<tr><td>Color Ring</td><td>Sets whether to enable the color ring feature or not, with the default setting of being disabled.
Note: Only when there are available color rings will this item appear.</td></tr>
<tr><td>Color Ring Index</td><td>The index of the color ring which will be quoted by the current FXS port.</td></tr>
<tr><td>Talkback</td><td>With this feature enabled and a number bound, the port can talkback to its bound number. That is, they can start a call with each other as soon as picking up the phone. The default setting is disabled.
Note: This feature is only used in the case of channel registration.</td></tr>
<tr><td>Bound Number</td><td>Sets the bound number for talkback.</td></tr>
<tr><td>Ringing Parameter</td><td>Set the ringing parameters of the FXS module. The default value is RING_ABS120V_DEF.
Note: It is not necessary to change this value under normal circumstances. For modification, please contact our technical support.</td></tr>
<tr><td>Feed Voltage Parameter</td><td>Set the feed voltage parameters of the FXS module. The default value is DCFEED_48V_21MA_DEF.
Note: It is not necessary to change this value under normal circumstances. For modification, please contact our technical support.</td></tr>
<tr><td>Impedance Parameter</td><td>Set the impedance parameter of the FXS module. The default value is ZSYN_200_680_100_30_0.
Note: It is not necessary to change this value under normal circumstances. For modification, please contact our technical support.</td></tr>
</table>

After configuration, click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

Or you can click ***Batch*** to modify several pieces of FXS settings at the same time. The configuration items on the FXS batch modification interface are the same as those on the FXS modification interface.

Some configuration items on this interface are the same as those on the ***FXS Modification Interface***. The others are described in the table below.

Item	Description
Starting Port	The starting serial number of the FXS port on the device in the batch setting.
Ending Port	The ending serial number of the FXS port on the device in the batch setting.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Display Name	The starting displayname in the batch setting.
Starting Authentication Password	The starting authentication password in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including ***Increase*** and ***Decrease*** two options.
SIP Account Batch Step Size	Sets the increase or decrease step size of the SIP account in the batch setting.
Display Name Batch Rule	The rule for batch setting the display name, including ***Increase***, ***Decrease*** and ***All Same*** three options.
Display Name Batch Step Size	Sets the increase or decrease step size of the display name in the batch setting.
Authentication Password Batch Rule	The rule for batch setting the authentication password, including ***Increase***, ***Decrease*** and ***All Same*** three options.
Authentication Password Batch Step Size	Sets the increase or decrease step size of the authentication password in the batch setting.

After configuration, click ***Modify*** to save the settings into the gateway, or click ***Cancel*** to cancel the settings.

3.7.2 FXO

The list on the FXO Settings interface shows the feature and properties of each FXO port. Click ***Modify*** to modify the properties of the corresponding port.

The table below explains the configuration items on the FXO modification interface.

Item	Description
Port	Serial number of the FXO port on the device.
Type	Type of the port on the device (FXO). This item is not configurable.
Register Port	Sets whether to register the port to the SIP server. When this item is set to *No*, the item ***Reg Status*** on the FXO settings interface shows *Unregistered*; when this item is set to *Yes*, the item **Reg Status** shows *Failed* or *Registered*.
SIP Account	Registration account of an FXO port. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 32 is 8032.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port. To register a port to the SIP server, both items ***SIP Account*** and ***Password*** must be filled in.

<table>
<tr><td>Display Name Preferred</td><td>In case this feature is enabled and the port group or the whole gateway is registered, if the display names set by the port are different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the sent SIP message will be the display name set by the port group; if the whole gateway is registered, the displayname in the sent SIP message will be the displayname of the gateway.</td></tr>
<tr><td>Server Index</td><td>The index of the SIP server which will be quoted by the current FXO port.</td></tr>
<tr><td>Connection Method</td><td>FXO connection methods include:
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Static Binding</td><td>Bind the number which corresponds to an FXS port to an FXO port. The number will be listed in the Bound Number column. This helps to achieve the corresponding binding between an FXO port and an FXS port (two-way).</td></tr>
<tr><td>Two Stages Dialing Mode (default)</td><td>Under this mode, an incoming call from an FXO port will go into the IVR system. Then IVR will play a speech prompt “Please dial the extension number”. If you fail to input the correct target station number before IVR finishes the third repeat of the prompt, the FXO will hang up the call automatically; otherwise, the corresponding station will ring.</td></tr>
</table>
Note: Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.</td></tr>
<tr><td>Input Gain, Output Gain</td><td>Adjusts the gain of the voice input to/ output from the FXO port when it is offhook or onhook. Range of value for output: -24~13, and range for input: -24~24, calculated by dB, with the default value of 0.</td></tr>
<tr><td>Echo Canceller</td><td>The echo cancellation feature for a call conversation over the FXO channel. By default, this feature is enabled and the effect can reach 128ms.</td></tr>
<tr><td>Forbid Outgoing Call</td><td>If this feature is enabled, the FXO port will be forbidden to call out. The default setting is disabled.</td></tr>
<tr><td>Caller ID Detection</td><td>If this feature is enabled, the FXO port will detect the caller IDs from the incoming calls. The default setting is enabled.</td></tr>
<tr><td>Polarity Reversal Detection</td><td>Once this feature is enabled, only when the FXO port detects the polarity reversal signal will the corresponding channel go into the talking state. The default setting is disabled. Note: This feature and the Two Stages Dialing feature cannot be enabled at the same time.</td></tr>
</table>

After configuration, click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

Or you can click ***Batch*** to modify several pieces of FXO settings at the same time. The configuration items on the FXO Batch Modification interface are the same as those on the FXO Modification interface.

Some configuration items on this interface are the same as those on the ***FXO Modification Interface***. The others are described in the table below.

Item	Description

Starting Port	The starting serial number of the FXO port on the device in the batch setting.
Ending Port	The ending serial number of the FXO port on the device in the batch setting.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Display Name	The starting displayname in the batch setting.
Starting Authentication Password	The starting authentication password in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including ***Increase*** and ***Decrease*** two options.
SIP Account Batch Step Size	Sets the increase or decrease step size of the SIP account in the batch setting.
Display Name Batch Rule	The rule for batch setting the display name, including ***Increase***, ***Decrease*** and ***All Same*** three options.
Display Name Batch Step Size	Sets the increase or decrease step size of the display name in the batch setting.
Authentication Password Batch Rule	The rule for batch setting the authentication password, including ***Increase***, ***Decrease*** and ***All Same*** three options.
Authentication Password Batch Step Size	Sets the increase or decrease step size of the authentication password in the batch setting.
Batch Rule of Bound Number	The rule for batch setting the bound number, including ***Increase***, ***Decrease*** and ***Use the same number*** three options.
Batch Step Size of Bound Number	Sets the increase or decrease step size of the bound number in the batch setting.
Whitelist of FXO Out Calls	Fill in the rule to match FXO outbound whitelisted number. Only those numbers matching this rule will be allowed by the gateway to call out from the FXO port. The default setting is null which means this feature is disabled..

After configuration, click ***Save*** to save the settings into the gateway, or click ***Cancel*** to cancel the settings.

3.7.3 FXO Port Timer

The FXO Port Timer interface displays such information as the max call time limit for a single call, the max call time limit for the total calls on each FXO port, as well as the timer clear cycle. Click Modify for each port to modify the timer settings.

The table below explains the configuration items on the interface:

Item	Description
Port	Serial number of the FXO port on the device.
Unit	Sets the timing unit for the call. The actual call time will be calculated as the integral multiple of the setting time. Take an example: supposed the setting time is 30s and the actual call time is 72s, thus, the gateway will consider the call time as 90s.
Time Limit on a Single Call	Sets whether to enable the time limit on a single call.
Max Call Time	Sets the maximum time length of a call.

Time Limit on Total Calls	Sets whether to enable the time limit on all calls at the port.
Timing Cycle	Sets the time count cycle for the port.
Clear	Sets the time node to clear the time count.
Set Spent Call Time	Sets the spent call time length of the port.
SIP Code Reply	Once the spent call time reaches the total time limit, the FXO port will not be able to make outgoing calls and the gateway will reply the designated SIP code to the IP side.
Time Limit per Day	Set the maximum length of calls per day for this port.
Apply to Other Ports	Sets whether to apply above settings to other ports or port groups.

Click ***Modify*** to save the settings into the gateway, click ***Return*** to cancel the settings.

3.7.4 FXO List Timer

The FXO List Timer interface displays the index information of the FXO port in timing. Click the ***Setting*** button on the top right corner to set the timer. Click the ***Add New*** button at the bottom to add the list timing rule.

The table below explains the configuration items on the interface:

Item	Description
Rule Index	The index of timing rule, used for the FXO port in list timing.
Set Spent Call Time	The length of the time already used in this rule.
Import Number	Import the matching numbers.
Number Matching Rule	There are two number matching modes: ***Prefix Matching*** and ***Whole Words only***.
Max Call Time	The maximum call time in this rule
Timing Cycle	The timing cycle in this rule
Clear	The time to clear the timer within the timing cycle in this rule

Click ***Save*** to save the settings into the gateway; click ***Reset*** to restore the configurations; click ***Return*** to cancel the settings.

3.7.5 Port Group

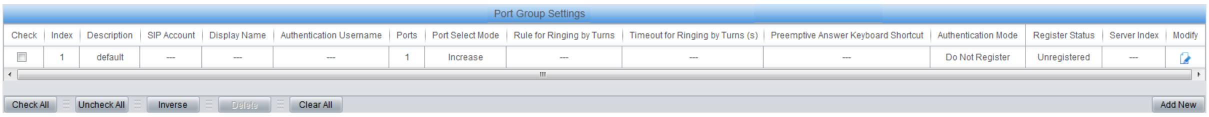


Figure 3-11 Port Group Settings Interface

See Figure 3-11 for the port group settings interface. A port group is a set containing single or multiple ports, used to specify such properties as ***Port Selection*** and ***Authentication Mode*** for all the ports in it. A new port group can be added by the ***Add New*** button on the bottom right corner of the above list. Note that a port which has been occupied by one port group cannot be chosen by others.

The table below explains the items on the interface.

Item	Description

<table>
<tr><td>Index</td><td>The unique index of each port group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to port groups.</td></tr>
<tr><td>Description</td><td>More information about each port group, with default value of default.</td></tr>
<tr><td>Register Port Group</td><td>To register the port group to the SIP server. Only when this configuration item is set to Yes can you see the configuration items SIP Account and Password.</td></tr>
<tr><td>SIP Account</td><td>When the port group initiates a call to SIP, this item corresponds to the username of SIP.</td></tr>
<tr><td>Display Name</td><td>Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.</td></tr>
<tr><td>Password</td><td>Registration password of the port group. To register the port group to the SIP server, both configuration items SIP Account and Password should be filled in.</td></tr>
<tr><td>Authentication Username</td><td>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.
Note: This item appears only when IMS Network or Multi-Registrar Server is enabled.</td></tr>
<tr><td>Server Index</td><td>The index of the sip server which will be quoted by the current FXS port.</td></tr>
<tr><td>Authentication Mode</td><td>Sets the way for SIP to make outgoing calls (Tel→IP) on the gateway.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Do Not Register (default)</td><td>SIP initiates a call in a point-to-point mode.</td></tr>
<tr><td>Register Gateway</td><td>SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to SIP for gateway registration.)</td></tr>
<tr><td>Register Port Group</td><td>SIP initiates a call with the registered SIP account and password of the port group.</td></tr>
<tr><td>Register Port</td><td>SIP initiates a call with the registered SIP account and password of the port.</td></tr>
</table></td></tr>
<tr><td>Register Status</td><td>Registration status of the port group. When Register Port Group is set to No, the value of this item is Unregistered; when Register Port Group is set to Yes, the value of this item may be Failed or Registered.</td></tr>
</table>

<table>
<tr><td>Port Select Mode</td><td>When the port group receives a call, it will choose a port based on the select mode set by this configuration item to ring or to connect. The optional values and their corresponding meanings are described in the table below.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Increase (default)</td><td>Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Decrease</td><td>Search for an idle port in the descending order of the port number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Decrease</td><td>Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Group Ringing</td><td>Ring all the idle FXS ports in this port group.</td></tr>
<tr><td>Ringing by Turns</td><td>Ring the ports in this port group according to the Rule for Ringing by Turns which can be user-defined. If there are more than one rule, they should be separated by comma. By default, the ringing will be carried out in the ascending order of the port number. Timeout for Ringing by Turns is used to set the overtime for ringing. Range of value: 15~60, calculated by s, with the default value of 20.</td></tr>
</table></td></tr>
<tr><td>Preemptive Answer Keyboard Shortcut</td><td>When a channel in a port group is ringing, another channel in the same port group can press the keyboard shortcut set by this item to transfer the call from the ringing channel to the current channel.
Note: This item will become invalid if the gateway works under the port select mode Group Ringing or Ringing by Turns.</td></tr>
<tr><td>Port Reused by Multiple Groups</td><td>Once this feature is enabled, a port can be added to different port groups.</td></tr>
<tr><td>Port</td><td>The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. Once the feature “Port Reused by Multiple Groups” is enabled, a port which has been occupied is still available for other port groups. All selected ports for a port group will be displayed in the Ports column in Figure 3-11. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.</td></tr>
</table>

After configuration, click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings. ***Check All*** means to select all available ports on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. ***Check All FXO Ports*** means to select all available FXO ports on the current page; ***Check All FXS Ports*** means to select all available FXS ports on the current page.

Click ***Modify*** at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. The configuration items on this interface are the same as those on the ***Add New Port Group*** interface.

To delete a port group, check the checkbox before the corresponding index in Figure 3-11 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all port groups at a time, click the ***Clear All*** button in Figure 3-11.

3.7.6 Advanced FXO Settings

The table below explains the configuration items on the Advanced FXO Settings interface.

Item	Description
Mailbox Account, Password	Sets the account and password of the mailbox.
Outgoing (SMTP), Port	Sets the server address and port for Email sending.
SSL	Sets whether to encrypt the sending/receiving mails via SSL.
Recipient	Sets the address of the recipient.
Subject	Sets the mail subject.
Content	Sets the mail content.
FXO Off-line Alarm	After selecting the ports, the gateway will send the alarm email when the selected ports are off-line.
Blacklist of FXO Out Calls	Fill in the rule to match FXO outbound blacklisted number. All the numbers matching this rule will be prohibited by the gateway to call out from the FXO port.
Sensitive Number Call Out Limit	Sets whether to forbid sensitive numbers (110, 119, 120, 122) to call out from the FXO port. By default it is checked.
International Call	When this function is enabled, the numbers starting with 00 can be called out from the FXO port. It is disabled by default.
Blacklist of FXO Incoming	Sets the blacklist of the FXO incoming calls.
Processing Mode	Sets the processing mode for the blacklist, including two options: Hang up after pick-up and Hang up after ringing. The default value is *Hang up after pick-up*.
Hang-up Delay	Sets the delay to hang up the call after the pick-up.

After configuration, click ***Save*** to save the settings into the gateway or click ***Reset*** to reset the settings.

3.8 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel and Tel→IP.

3.8.1 Routing Parameters

On the routing parameters configuration interface, you can set the routing rules for calls respectively on two directions IP→Tel and Tel→IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*. The gateway will send the option message to detect whether the TEL->IP routing is valid or not after setting the Route Detection Cycle. If the remote address doesn't respond this option message within the set cycle, this routing will be regarded as invalid and the outgoing calls won't be routed to this TEL->IP routing.

After configuration, click ***Save*** to save the above settings into the gateway.

3.8.2 IP to Tel

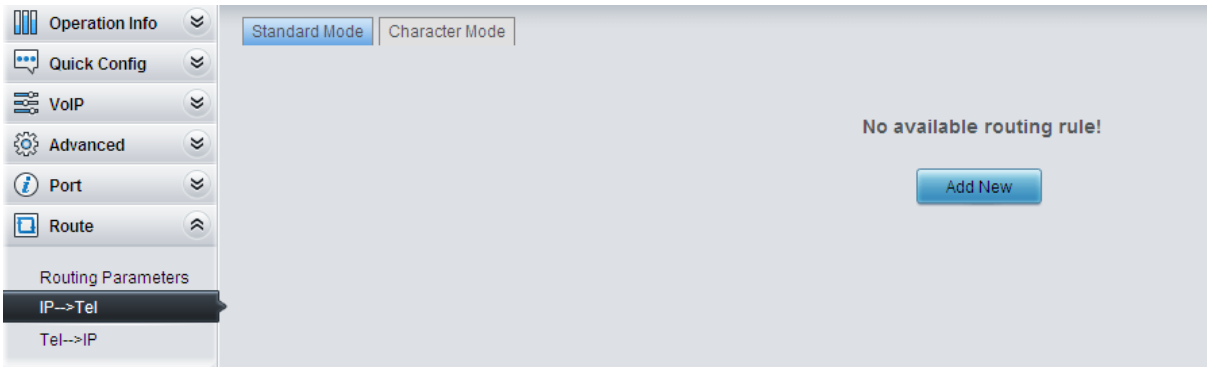


Figure 3-12 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-12 for the IP→Tel routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP→Tel routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. You may use the default values of all the configuration items herein.

The table below explains the items on the interface.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default*.
Source IP	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, 、"[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be characters '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source IP*** specify a routing rule for calls.

	Note: "[*]" represents TFM symbol *, while "*" represents any string.
Route by Number	When this feature is enabled, the gateway will route a call from IP to a corresponding port based on its number. And the number of the port which this call will be routed to can be set via the item ***SIP Account*** on the FXS or FXO Settings interface. In such case, the configuration item ***Call Destination*** goes invalid and shows *Route by Number* on the routing rule configuration interface. The default setting is *disabled*.
Call Destination	Port group to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-13 for the IP→Tel routing rule configuration interface after your configuration. There is a rule displayed with Index 63 and Call Destination 'Route by Number', having no restriction on Source IP, CallerID Prefix and CalleeID Prefix, which indicates the gateway will route a call from any IP address to a corresponding port based on its number.

Press the ***Add New*** button on the bottom right corner of the list to add a new routing rule.

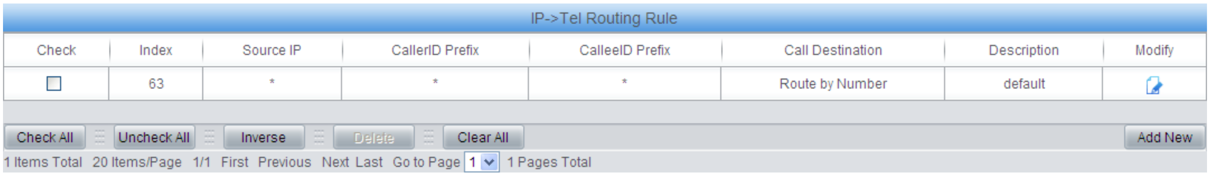


Figure 3-13 IP→Tel Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-13 to modify a routing rule. The configuration items on the IP→Tel routing rule modification interface are the same as those on the ***Add New Routing Rule (IP→Tel)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-13 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-13.

Under the Character mode, you can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.8.3 Tel to IP

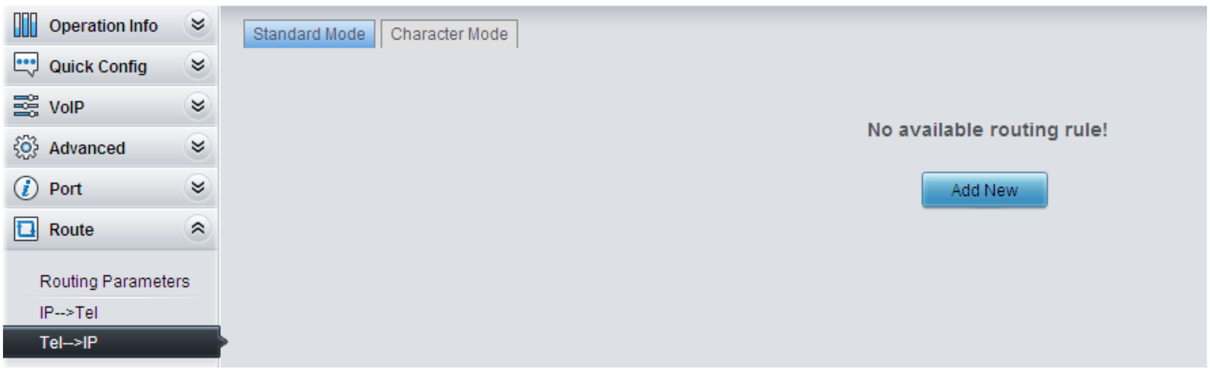


Figure 3-14 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-14 for the Tel→IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The Tel→IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. You may use the default values of all the configuration items herein except for ***Destination IP*** and ***Destination Port***.

The table below explains the items on the interface.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default.*
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or characters ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any characters between these two characters. ',' is used to separate characters or characters ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source Port Group (Call Initiator)*** specify a routing rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Destination IP, Destination Port	IP address and port number of the remote end to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-15 for the Tel→IP routing rule configuration interface after your configuration. There is a rule displayed with Index 63, Destination IP '192.168.1.101' and Destination Port '5060' (i.e. default IP address and port of the gateway), having no restriction on Call Initiator, CallerID Prefix and CalleeID Prefix, which indicates all the outgoing calls from Tel which conform to the dialing rule will be routed to the gateway.

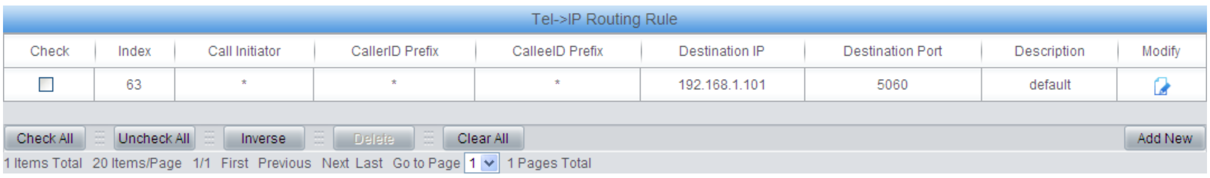


Figure 3-15 Tel→IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-15 to modify a routing rule. The configuration items on the Tel→IP routing rule modification interface are the same as those on the ***Add New Routing Rule (Tel→IP)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-15 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-15.

Under the Character mode, you can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.9 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-16.

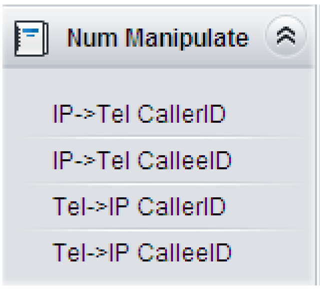


Figure 3-16 Number Manipulation

3.9.1 IP to Tel CallerID

On the IP→Tel CallerID manipulation interface under the Standard mode, a new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list. You may use the default values of all the configuration items on the IP→Tel CallerID manipulation rule adding interface.

The table below explains the items on the interface.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default*.
Call Initiator	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.

Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. The default value is 0.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. The default value is 0.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated. The default value is 20.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** to modify a number manipulation rule. The configuration items on the IP→Tel CallerID manipulation rule modification interface are the same as those on the ***Add IP→Tel CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

To delete a number manipulation rule, check the checkbox before the corresponding index and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button.

Under the Character mode, you can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.9.2 IP to Tel CalleeID

The number manipulation process for IP→Tel CalleeID is almost the same as that for IP→Tel CallerID; only the number to be manipulated changes from CallerID to CalleeID. The configuration items on IP→Tel CalleeID manipulation interface are the same as those on ***IP→Tel CallerID Manipulation Interface***.

3.9.3 Tel to IP CallerID

Under the Standard mode, a new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. You may use the default values of all the other configuration items on the Tel→IP CallerID manipulation rule adding interface.

The table below explains the items on the interface.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of

	default.
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. The default value is 0.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. The default value is 0.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated. The default value is 20.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** to modify a number manipulation rule. The configuration items on the Tel→IP CallerID manipulation rule modification interface are the same as those on the ***Add Tel→IP CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

To delete a number manipulation rule, check the checkbox before the corresponding index and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button.

Under the Character mode, you can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.9.4 Tel to IP CalleeID

The number manipulation process for Tel→IP CalleeID is almost the same as that for Tel→IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. The configuration items on Tel→IP CalleeID manipulation interface are the same as those on ***Tel→IP CallerID***

Manipulation Interface.

3.10 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check.

3.10.1 Management

The table below explains the items on the Management Parameters Setting interface.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also can set an IP blacklist to forbid all IPs within it to access the gateway.
SYSLOG	Sets whether to enable SYSLOG. It is required to fill in **SYSLOG Server Address** and **SYSLOG Level** in case SYSLOG is enabled. By default, ***SYSLOG*** is disabled.
Server Address	Sets the SYSLOG server address for log reception.
SYSLOG Level	Sets the SYSLOG level. There are three options: *ERROR*, *WARNING*, *INFO* and *DEBUG*. The default value is *INFO*.
Send CDR	Sets whether to enable the feature of sending CDR. It is required to fill in **Server Address** and **Server Port** in case Send CDR is enabled. By default, **Send CDR** is disabled.
Server Address	The address of the server to receive CDR.
Server Port	The port of the server to receive CDR.
NTP	Sets whether to enable the NTP time synchronization feature. It is required to fill in ***NTP Server Address***, ***Synchronizing Cycle*** and ***Time Zone*** in case NTP is enabled. By default, ***NTP*** is enabled.
NTP Server Address	Sets the Server address for NTP time synchronization. By default, the address is time.nist.gov
Synchronizing Cycle	Sets the cycle for NTP time synchronization, calculated by s, with the default value of 3600.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset ***Restart Time***. By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.
System Time	The system time. Check the checkbox before ***Modify*** and change the time in the edit box when NTP is disabled.
Time Zone	The time zone of the gateway.

3.10.2 Configuration File

The Configuration File interface includes two files: SMGConfig.ini and ShConfig.ini. You can check and modify the items in these configuration files through this interface. Configurations about the gateway server, such as route rules, number manipulation and so on, are included in SMGConfig.ini; configurations about the board are included in ShConfig.ini. You can modify these configurations on the interface directly, and then click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations.

3.10.3 Network

A gateway has two LANs which can be respectively configured with network type and IP address on the network settings interface. Network Type has three options: Static, DHCP and PPPoE. IPv4 and IPv6 address configurations are supported. If PPPoE is used, it is necessary to enter the username and the password of the network.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. After changing the IP address, you shall log in the gateway again using your new IP address.

3.10.4 Upgrade

On the upgrade interface you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package "*.tar.gz" (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification.) via ***Browse…*** and click ***Update***. Then the file uploading interface will appear. See Figure 3-17.

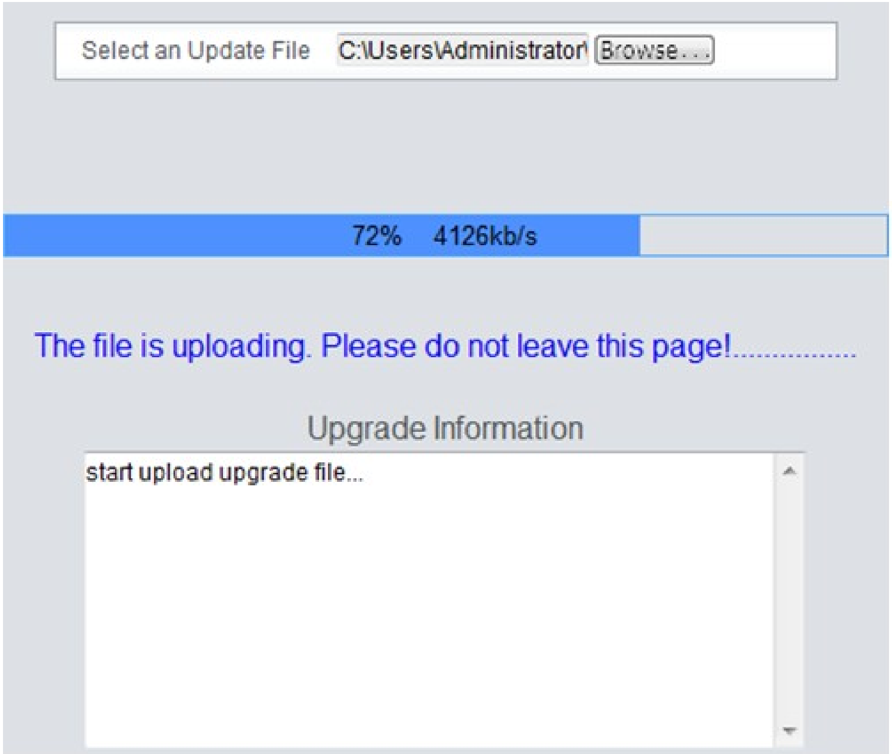


Figure 3-17 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-18 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

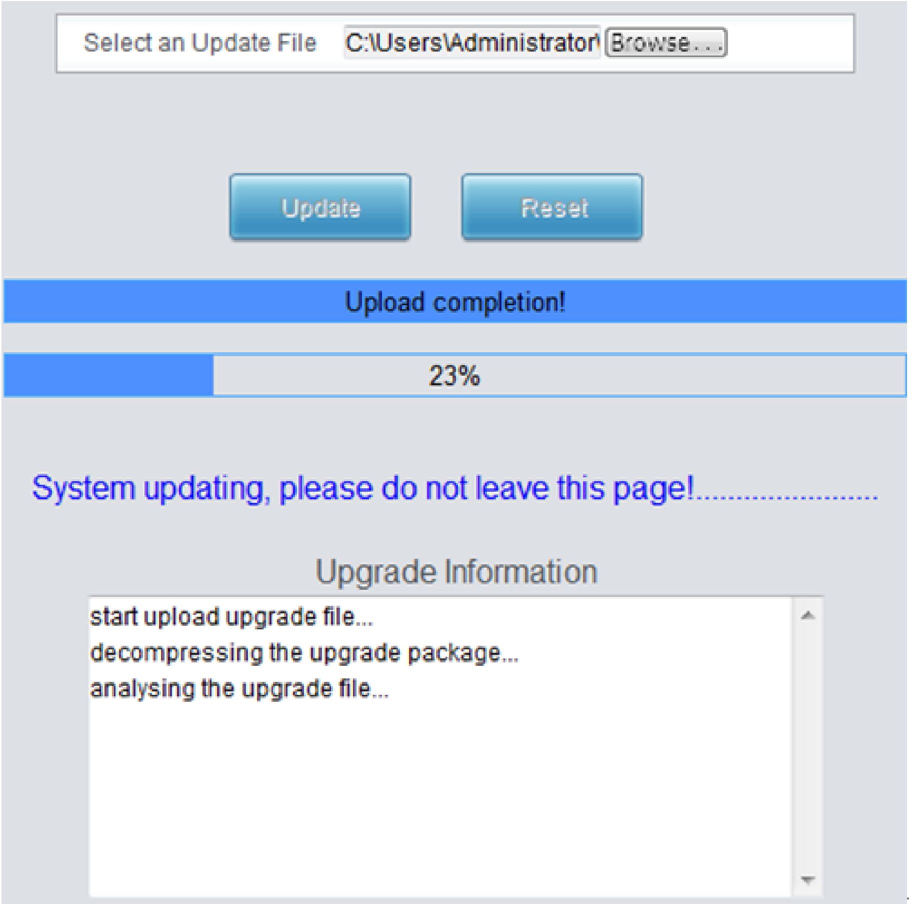


Figure 3-18 System Upgrading Interface

Note that clicking ***Reset*** can only delete the selected update file but not cancel the operation of ***Update***.

Note: Please contact our technicians if you need to downgrade the gateway to an old version. An improper operation may cause unexpected problems.

3.10.5 Signaling Capture

See the Signaling Capture interface. Packet capture contains Signaling Packet Capture and RTP Packet Capture. You can select either of them to start the capture according to your requirement. Click ***Start*** to start capturing packets. Click ***Stop*** to stop the capture and download the captured packets.

On the Debug & Record interface, you can select a channel and the recording mode to start the data recording. Click ***Start*** to start the corresponding recording. Click ***Stop*** to stop the recording and download the recorded file.

3.10.6 Call Log

On the Call Log interface, click the checkbox before ***Enable Call Log*** to enable the call log feature, including ***Call Log*** and ***SIP Log***. ***Call from IP Channel*** displays the call log information generated on all IP channels, and ***Call from Port*** displays the call log information generated on the port you select. All the SIP related information will be displayed in ***SIP Log***.

3.10.7 Operation Log

The Operation Log interface is used to check the operation records on WEB. Click ***Refresh*** to refresh the log; click ***Clear All*** to clear all the operation logs and click ***Download*** to download the logs.

Note: The sign <@#> here means the configuration item is unconfigured.

3.10.8 Backup & Upload

See the backup and upload interface. To back up the configuration file to your PC, just click ***Backup***. To upload a configuration file, select it via ***Browse…*** and click ***Upload***.

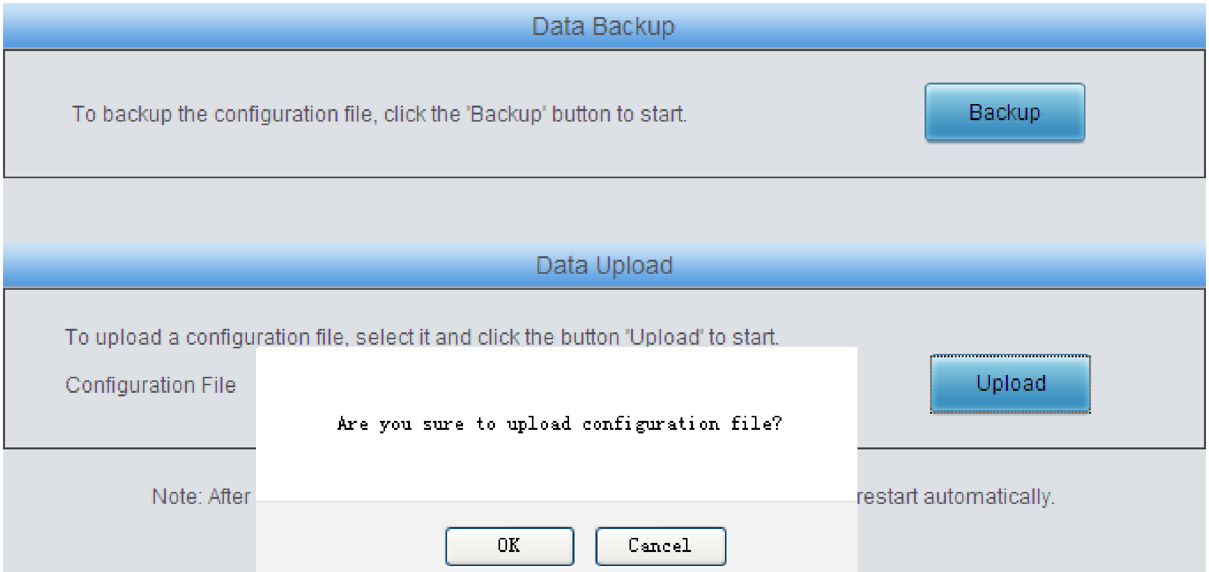


Figure 3-19 Backup & Upload & Prompt Interface

Click ***OK*** on the prompt box to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. The gateway will overwrite the current configurations with the uploaded data after restart. Click ***Cancel*** to cancel this upload directly.

3.10.9 Factory Reset

On the factory reset interface, click ***Reset*** to restore all configurations on the gateway to factory settings.

3.10.10 System Monitor

See the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled. Threshold to Judge Heartbeat Loss for Service is used to judge whether the gateway receives the heartbeat packets from the service during the set time, if not, it is considered that the gateway service has been disconnected. It is calculated by s, with the value range of 20~120s and the default value of 60s.

3.10.11 Certificate Management

The Certificate Management interface provides the service to create, download, and upload related certificates required by the TLS protocol.

To make a certificate, fill in the certificate related information on the page, including the country, state or province, city, company, department, host name (consistent with the gateway SIP address), email, then click Generate and the gateway will automatically generate the relevant certificate.

Note: All the above information must be written in English.

After the certificate is successfully generated, you can click Download to download the CA certificate required by the TLS protocol.

Click Upload to upload the relevant certificate required by the TLS protocol.

3.10.12 Call Test

See the Call Test interface. A call test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items on the interface.

Item	Description
Test Type	There are two types of call tests: ***PSTN Call out*** and ***IP Call out***.
Channel	The channel on which the call test will be performed.
CalledID	The called party number of the call from the PSTN channel.
Local Alias	The content of displayname in the from field of the invite message during the call out from the IP channel.
Local SIP Account	The content of username in the from field of the invite message during the call out from the IP channel.
Remote Alias	The content of displayname in the to field of the invite message during the call out from the IP channel.
Remote SIP Account	The content of username in the to field of the invite message during the call out from the IP channel.
Called IP Address	The called IP address of the call out from the IP channel.
Called Port	The called port of the call out from the IP channel.
DTMF	The DTMF digits sent by the IP channel after starting a call.
Add or Modify Invite Header Field	The field name and content added or modified in the message header during the call out from the IP channel.
Signaling Trace	Displays the call test process.

After configuration, click ***Start*** to execute the call test; click ***Stop*** to terminate it immediately; click ***Clear*** to clear the records of call tests.

3.10.13 Centralized Manage

Go to the Centralized Manage Setting interface. The gateway can register to a centralized management platform and accept the management of the platform. The table below explains the items on the interface.

Item	Description

Management Platform	Select a management platform for the gateway to register, including two options: DCMS and Others.
Server Address	The address of the server in which the management platform locates, It can be IP or a domain name, valid only when DCMS is selected. Note: To configure the domain name, the DNS should be already configured and the corresponding domain name must be analyzable.
Company Name	The name used to register the gateway to Synway DCMS, valid only when DCMS is selected.
Authorization Code	The authorization code is used for the connection verification. A device can connect to the DCMS successfully only after it passes the verification. Only valid when DCMS is selected.
Gateway Description	The description displayed on Synway DCMS after the gateway is registered to Synway DCMS, giving an easy identification of the gateway in device grouping. This item is valid only when DCMS is selected.
Enable Lock Feature Once Successfully Connected	Once this feature is enabled, you can lock the device according to the corresponding parameters. This item is valid only when DCMS is selected.
IP Address	Once this feature is enabled, you are required to fill in the authorization code while modifying the information related to the IP address in the Network interface. This item is valid only when DCMS is selected.
Registrar Server	Once this feature is enabled, you are required to fill in the authorization code while modifying the address and port of the registrar server in the SIP Settings interface. This item is valid only when DCMS is selected.
Working Status	The status of the connection between the gateway and the centralized management server. This item is valid only when DCMS is selected.
Centralized Management Protocol	Set the centralized management protocol. It only supports SNMP currently.
SNMP Version	Set the version of SNMP, three options available: V1, V2 and V3, with the default value of V2. This item is valid only when Others is selected.
Monitoring Port	Monitoring Port for SNMP on the gateway. This item is valid only when Others is selected.
Community String	Community string used for information acquisition.
Account	The account of SNMP, valid only when the SNMP version is set to V3.
Grade	The grade of SNMP, three options available: Neither authenticated nor encrypted, Authenticated but not encrypted and Authenticated and encrypted, with the default value of *Neither authenticated nor encrypted*. It is valid only when the SNMP version is set to V3.
Authentication Password	The authentication password required to enter when the item Grade is set to Authenticated but not encrypted or Authenticated and encrypted.
Encryption Password	The encryption password required to enter when the item Grade is set to Authenticated and encrypted.

3.10.14 Access Control

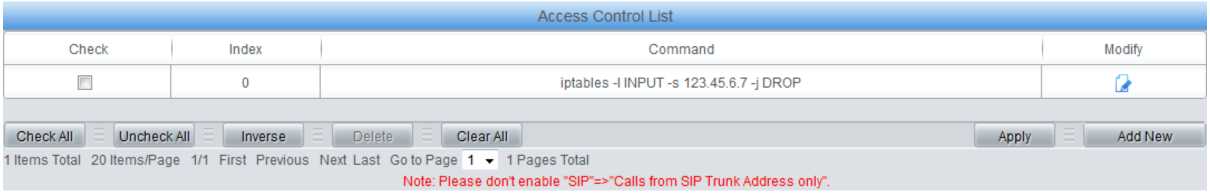


Figure 3-20 Access Control List Interface

See the Access Control List interface. Once you add a piece of command to ACL, the network flow will be restricted: only the particular devices are allowed to visit the gateway and only the data packages on the designated ports can be forwarded. Click ***Add New*** to add a new piece of command. See Figure 3-21.

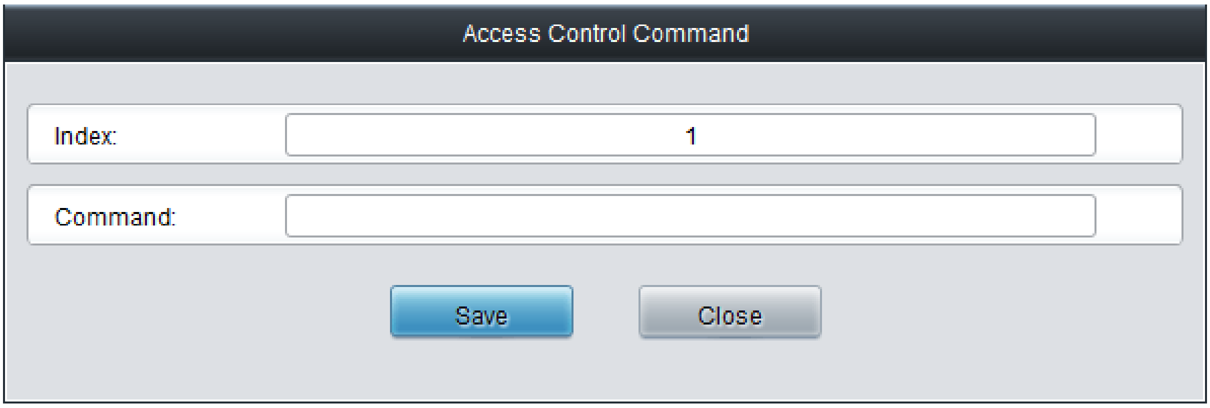


Figure 3-21 Add Access Control Command Interface

Input a piece of command into the Command item and click ***Save*** to save the settings to the gateway. Click ***Close*** to cancel your settings. After that, click ***Apply*** to make the new command valid.

Click ***Modify*** in Figure 3-20 to modify a command. The configuration items on the Access Control Command Modification interface are the same as those on the ***Add Access Control Command*** interface. Note that the item ***Index*** cannot be modified.

To delete an Access Control Command, check the checkbox before the corresponding index and click the ***Delete*** button, and then click the ***Apply*** button to make the deleted command invalid. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all access control commands at a time, click the ***Clear All*** button.

Note:

1. Currently, only the command iptables is supported by the gateway.

2. After you add, modify or delete a command manually, don't forget to click the ***Apply*** button to make your settings valid. However, in case the gateway restarts or the configuration is leading-in, the command will get valid automatically without the need for you to click the ***Apply*** button.

3.10.15 PING Test

On the Ping test interface, a Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items on the interface.

Item	Description

Destination Address	Destination IP address or domain name on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

After configuration, click ***Start*** to execute the Ping test; click ***End*** to terminate it immediately.

3.10.16 DNS Test

The DNS test is used to test whether the domain name can be successfully parsed by the gateway. Fill in the domain name that needs to be parsed and click the button Test, then the window will pop up and prompt the test result. When the parsing succeeds, the test result will contain the IP address acquired from the domain name.

3.10.17 TRACERT Test

On the Tracert test interface, a Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items on the interface.

Item	Description
Source IP Address	Source IP address where the Tracert test is initiated.
Destination Address	Destination IP address on which the Tracert test is executed.
Maximum Jumps	Maximum number of jumps between the gateway and the destination address which are returned by the Tracert test. Range of value: 1~255.
Info	The information returned during the Tracert test, helping you to learn the detailed information about the jumps between the gateway and the destination address.

After configuration, click ***Start*** to execute the Tracert test; click ***End*** to terminate it immediately.

3.10.18 Change Password

On the Password Changing interface you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click ***Save*** to apply the new username and password or click ***Reset*** to restore the configurations. After changing the username and password, you are required to log in again.

3.10.19 Restart

On the Service Restart part, click ***Restart*** to restart the service; on the System Restart part, click ***Restart*** to restart the whole gateway system. A dump file will be generated each time you restart the system. Click ***Download*** and you can download it to help troubleshoot issues.

Appendix A Technical Specifications

Dimensions

SMG1008D: 180×30×108mm^3

SMG1032D: 440×44×202mm^3

Weight

SMG1008D: 0.55kg

SMG1032D-32S: 2.7kg

SMG1032D-32O: 2.6kg

Power Consumption

SMG1032-32O: 5-10W

SMG1032-32S: 11-40W

Environment

Operating temperature: 0℃—45℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 4 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

FXS/FXO Port

Amount: 4/8/16/32

Type: RJ11

Impedance

Telephone line impedance: Compliant with the national standard impedance for three-component network

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: MINI USB Connector

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

Note: Follow the above settings to configure the serial port; or it may work abnormally.

Power Requirements

Input power:

SMG1004D, SMG1008D: 12V DC the direct current bigger than 3A

SMG1016D, SMG1032D: 100～240V AC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A	*64 kbps*
G.711U	*64 kbps*
G.729A/B	*8 kbps*

Sampling Rate

8kHz

Safety

Lightning resistance: Level 4

Meet YD-T 993-2006 lightning protection technical requirements and test methods for telecommunication terminal equipment

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the SMG-D gateway?

There are two ways to get the IP address:

1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101

2) Dial the corresponding function key through an FXS port to query the IP address. See Function Key for more details.

Q2. The SMG-D gateway only supports routing on two directions, i.e. Tel→IP and IP→Tel. What to do if I want to make a Tel→Tel call?

By default, you can make Tel→Tel calls without any routing configuration.

If you need to make Tel→Tel calls in a specific way, try via the routing of Tel→IP→IP→Tel. See below for detailed introductions.

Provided you are going to initiate a call from Port Group 1 to Port Group 2; the IP address and port number of your gateway are 192.168.1.101 and 5060 respectively.

a) Add a new routing rule on the Tel→IP routing rule configuration interface. Select a port group (e.g. **Port Group 1**) as 'Source Port Group' to initiate the call and fill in 'Destination IP' and 'Destination Port' with the gateway's IP address (e.g. **192.168.1.101**) and port number (e.g. **5060**). Then the call initiated from the station corresponding to Port Group 1 will be routed to the gateway.

b) Add a new routing rule on the IP→Tel routing rule configuration interface. Fill in 'Source IP' with the gateway's IP address (e.g. **192.168.1.101**) and select a port group (e.g. **Port Group 2**) as 'Destination Port Group' to be called. Then if the IP end of the gateway calls itself, the station corresponding to Port Group 2 will ring.

c) Finishing the above configurations, you can perform a Tel→Tel call from Port Group 1 to Port Group 2 simply by the way you make a Tel→IP call.

Q3. Does call forwarding involve routing and number manipulation?

Case 1: If the forwarding number is the number of the gateway port. There is no need to use routing and number manipulation rules. Because the gateway will find the corresponding number according to the forwarding number and make a call.

Case 2: If the forwarding number is not the number of the gateway port. It is required to use routing and number manipulation rules. A call forward procedure can be regarded as a Tel→IP call. It uses the routing rules and number manipulation rules in the same way as the Tel→IP call. A complete call forward is performed as follows:

a) An incoming IP call to the gateway rings the port which matches the IP→Tel routing and number manipulation rules and obtains a new CallerID.

b) Then the gateway uses the newly obtained CallerID and the call forward number, via the Tel→IP routing and number manipulation rules, to make another call from the port to a remote IP address.

Q4. In what cases can I conclude that the SMG-D gateway is abnormal and turn to Synway's technicians for help?

a) During runtime, the run indicator does not flash or the alarm indicator lights up or flashes, and such error still exists even after you restart the device or restore it to factory settings.

b) Voice problems occur during call conversation, such as that one party or both parties cannot hear the voice or the voice quality is unacceptable.

c) The port of the gateway is well connected, but the channel indicator never lights up after the gateway startup or the color it lights up does not comply with the actual state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers and abnormal dialing operations on the FXS port are probably caused by configuration errors. We suggest you refer to Chapter 3 WEB Configuration for further examination. If you still cannot figure out or solve your problems, please feel free to contact our technicians.

Q5. What to do if I cannot enter the WEB interface of the SMG-D gateway after login?

This problem may happen on some browsers. To settle it, follow the instructions here to configure your browser. Enter 'Tools > Internet Options >Security Tab', and add the current IP address of the gateway into 'Trusted Sites'. If you changes the IP address of the gateway, add your new IP address into the above settings too.

Q6. How many ports can be rung by turns according to the *Ringing by Turns* rule?

According to the 180s ringing timeout limit in RFC3261 protocol, the time used for ringing all ports by turns cannot exceed 180s. Therefore, based on the minimum timeout 15s for each port in the ringing queue, the maximum number of ports for ringing by turns is 12.

For example, if you set ***Timeout for Ringing by Turns*** to 20s, the maximum number of ports for ringing by turns should be 180s/20s=9; if you set ***Timeout for Ringing by Turns*** to 30s, the maximum number of ports for ringing by turns should be 180s/30s=6.

Q7. Is there any cell-phone APP can make calls to the SMG-D gateway?

Yes. Linphone is a soft SIP phone that is supported by multiple platforms, such as Linux, Windows, iOS, Android, etc. It must be registered to the SIP registrar server before dialing to other SIP devices or PSTN telephones,

Q8. Does the SMG-D gateway support fax?

Yes. Currently the SMG-D gateway supports two fax modes: T.38 and Pass-Through.

Q9. Which RTP codecs are supported by the SMG-D gateway?

At present, the supported RTP codecs are: G.711A, G.711u, G.729.

Q10. How to configure the feature Communication without Network for the SMG-D analog gateway?

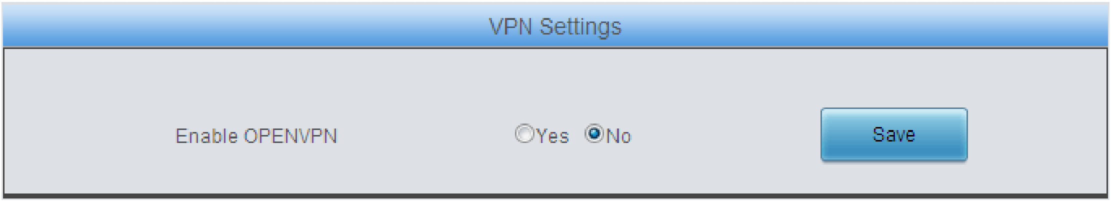
The feature **Communication without Network** is implemented via the WEB management over the analog gateway. It will automatically route a call to the FXO port in case of network failure or call timeout.

Refer to Q2 in this chapter for detailed information.

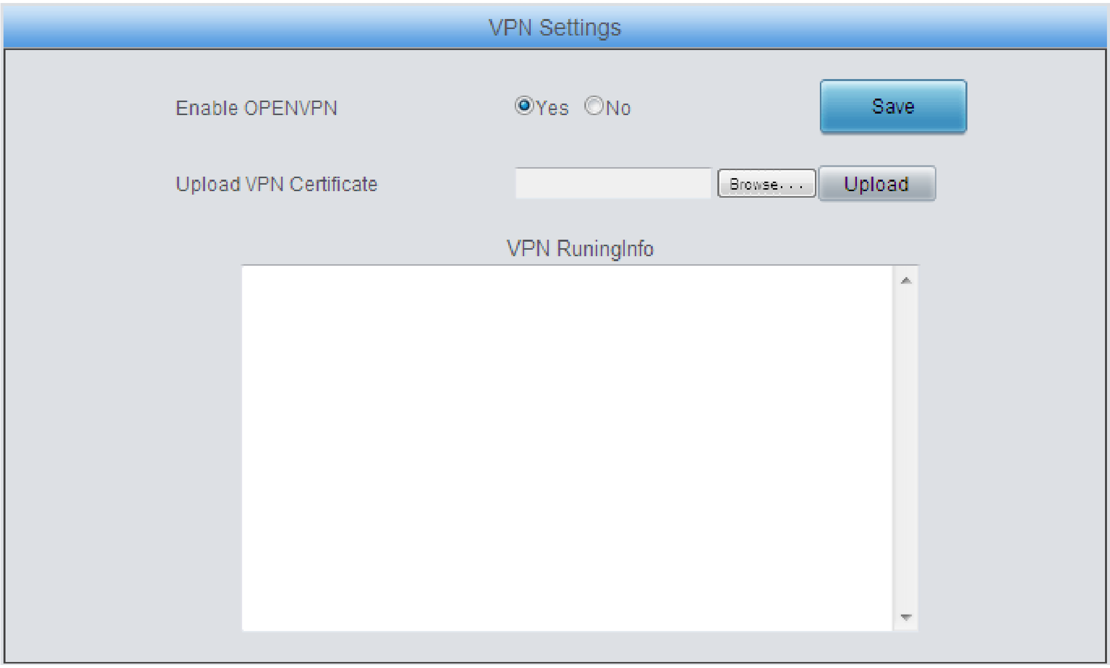
Appendix C About VPN

Part 1: Steps to Enable VPN Feature

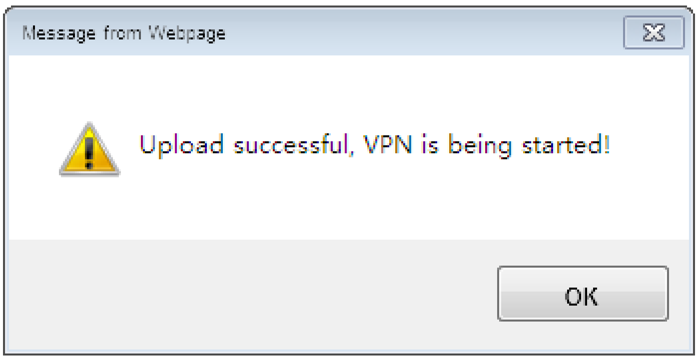
Find the VPN Settings interface under Advanced Settings on the web. This featured is disabled by default.



Step 1: Select Yes to enable this feature, click the ‘Save’ button and the following interface will appear.



Step 2: Select a certificate from the client, that is, a configuration file with the suffix of .conf, and then click the ‘Upload’ button. The following dialog will appear.



Step 3: Now you will get a virtual IP address which is allocated automatically by the VPN server. Note that each upload will lead to a new allocation of the IP address; however, restarting the gateway will not change the virtual IP address.

Then you may use the PING test under System Tool on the web to test if the client connects successfully with the server via IP, by which to check whether the VPN feature is successfully enabled or not.

Part 2: Steps to Make VPN Certificate

Step 1: Get the file of client.ovpn from the VPN server (under the 'sample-config' directory of the installation package) and rename it to "client.conf".

Step 2: Examine or add the following content into the file.

The file should contain the following content, in which the black part is fixed while the red part shall change according to the note.

client

dev tap (Note: Fill in tap or tun according to the VPN server's requirement.)

proto tcp (Note: Connect via TCP which should be consistent with that of the server.)

;cipher AES-128-CBC (Note: Select an encryption algorithm which should be consistent with that of the client. It is not necessary to add if there is no algorithm at the client.)

remote 192.168.143.235 1194 udp (Note: Fill in the IP address and the port number of the VPN server, and the protocol can be left empty.)

;remote-random (Note: If there are multiple servers configured, let the client connect at random.)

resolv-retry infinite (Note: Analyze the server's domain name)

nobind (Note: Not to bind any port to the client)

persist-tun

persist-key

mute-replay-warnings (Note: Set as a flag to warn about replayed data packages.)

ns-cert-type server

comp-lzo (Note: Use the lzo compression which is consistent with the server.)

verb 3

;tls-client

;tls-auth ta.key 1 (Note: It is used to enable the feature of TLS encryption, and should be consistent with that of the server.)

<ca>

-----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of ca.crt.

-----END CERTIFICATE-----

</ca>

<cert>

-----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of client.crt, that is, the content inbetween "-----BEGIN CERTIFICATE-----" and "-----ENDCERTIFICATE-----"

-----END CERTIFICATE-----

</cert>

<key>

-----BEGIN RSA PRIVATE KEY-----

Note: Fill in the key copied from the file of client.key

-----END RSA PRIVATE KEY-----

</key>

Note: The following key is not necessary to add if it is never encrypted at the server.

<tls-auth>

Note: Fill in the key copied from the file of ta.key

</tls-auth>

Make sure the three key files ca.crt, client.crt and client.key are of the newest versions.

Step 3: Save the file after your examination or supplement and upload it to the device. Note that the suffix of the file must be .conf.

Part 3: Attentions

a) After the VPN featured is opened at the server, use your PCs to connect as a test. If two PCs can PING through each other, it means the server works normally.

b) Make sure the server is OK and the configuration file is ready before opening the VPN feature. The system time of the analog gateway must be consistent with that of the server, or the connection may sometimes fails.

After enabling the VPN feature successfully, you can use the virtual IP of the gateway to make calls in both directions IP-->tel and tel-->IP.

Appendix D Technical/sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you.

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