New Rock Technologies, Inc.

MX Voice-Fax Gateway Series

User Manual

MX8
MX60
MX120

http://www.newrocktech.com
Email: gs@newrocktech.com

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1 Overview

1.1 Product Introduction

MX Series intelligent VoIP Gateways (**MX Gateways**) are designed to bridge the traditional telecom terminal device into IP networks through SIP or MGCP protocols. The main applications include:

For carriers and value-added service providers to provide telephone, fax and voice-band data services to subscribers using IP access methods such as FTTB, HFC, and ADSL;

Used to bridge the traditional telecom terminal equipments, such as PBXs, to the IP core networks of carriers;

Connected with PBX of enterprises to provide IP-based voice private network solutions for institutions, enterprises and schools;

Used as remote acces equipments for IP-PBXs in call center deployment.

MX Series includes MX8, MX60and MX120 subseries. Their features are similar with the main differences as follows:

Table 1-1 MX series gateway hardware specifications

Model	Voice ports	Chassis	Installation	CPU	RAM	Flash	Power
MX8	2-8	Plastic Casing	Desktop	MPC852T	64MB	16MB	5-9 VDC
MX60	16-48	19'' wide and 1U High	Rack	AT91SAM9G20B	64MB	16MB	100-240 VAC
MX120	48-96	19" Wide and 2U High	Rack	MPC8247	128MB	16MB	100-240 VAC, -48 VDC (Optional)

Hardware of MX series gateways use high-performance CPUs, ensure that each product of the series can achieve full-capacity concurrent calls with high speech quality.

MX gateways software adopts the stable and reliable embedded Linux operating system (OS), thereby implementing scores of business phone functions, including, call forwarding, call transfer, call hold, teleconference, caller identification, Do Not Disturb, ringback tone, hunt group simultaneous ring, distinctive ring, one phone with two numbers, and fax. In addition, MX gateways are featured with FXO port second stage dialing with voice prompt, routing table with a maximum of 500 entries, pone digit manipulation, and PSTN failover upon power-off or network disconnection.

MX gateways support local and remote management operations through Web GUI or Telnet and also support SNMPv2-based and TR069/TR104/TR106-based centralized management scheme. Maintenance tasks such as modifying configuration, upgrading software, collecting statistical data, downloading logs, and fault alarms can be performed.

1.2 Functions and Features

- Connect analog telephone, PBX, facsimile machine and POS machine to the IP core network, or PSTN;
- Work with service platform to provide various telephone supplementary services;
- Support protocols: SIP, MGCP;
- Flexible configuration of Phone/Line interfaces;
- Support G.711, G.729, G.723.1, GSM, iLBC;
- Support echo cancellation;
- Up to 500 routing rules can be stored in gateways;
- Intercom;
- Support digitmap;
- Support call progress tones for various countries and regions;
- Support Line second stage dialing or voice prompt;
- Support PSTN failover through FXO ports;
- Security strategy: IP filter, encryption
- Support routing table;
- Support T.30/T.38 fax mode;
- Support polarity inverse detection and busy tone detection;
- Support three-way calling;
- Compatible with unified communication solutions, such as CallManager, OCS and Asterisk;
- Support SNMPv2 and TR069/TR104/TR106
- Support Web GUI-based management , Telnet, automatic software upgrades, and configuration downloading
- Support high availability, implementing a cloud of SIP servers working in primary-standby or load balancing mode
- Support auto provisioning
- Support security settings such as white list

1.3 Equipment Structure

1.3.1 MX8

MX8 is the product with smallest capacity in MX Gateway Series. Designed with small plastic structure for desktop placement, MX8 can provide up to 8 analog line interfaces. MX8 supports the following types of configuration:

Table 1-2 Common Configuration Combination of MX8

Models	Number of FXS Ports	Number of FXO Ports
MX8-2S/2	2	2
MX8-6S/2	6	2

Models	Number of FXS Ports	Number of FXO Ports
MX8-4S	4	0
MX8-8S	8	0
MX8-4FXO	0	4
MX8-8FXO	0	8
MX8-4S/4	4	4

Figure 1-1 MX8 Front Panel

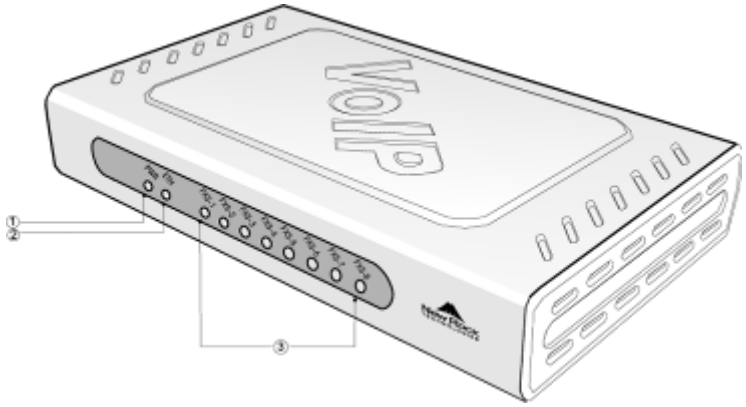


Table 1-3 Description of MX8 Front Panel

#	Description
①	Power indicator (PWR), Light-on indicates that it has been powered.
②	Steady on indicates valid Ethernet link, flashing indicates Ethernet activities (receiving and/or transmitting)
③	Analog subscriber line (FXS) or analog trunk (FXO) interface indicator, Light-on indicates that it is in use.

Figure 1-2 MX8 Back Panel

Table 1-4 Description of MX8 Back Panel

#	Description
①	Power interface, 5-9 VDC input

#	Description
②	10/100 M Ethernet Interface, RJ45
③	Analog subscriber line (FXS) or analog trunk (FXO) interface

Table 1-5 Configuration Description of Analog Line Interfaces for All MX8 Models

MX8 Models	RJ11 Interface Configuration							
	1	2	3	4	5	6	7	8
MX8-2S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	NA	NA	NA	NA
MX8-6S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6
MX8-4S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	NA	NA	NA	NA
MX8-8S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6	Subscriber Line 7	Subscriber Line 8
MX8-4FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	NA	NA	NA	NA
MX8-8FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	Trunk Line 5	Trunk Line 6	Trunk Line 7	Trunk Line 8
MX8-4S/4	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4

1.3.2 MX60

Designed with a 1U high and 19'' wide compact chassis, MX60 is suitable for installation in a standard cabinet. MX60 has a built-in power module with the rating voltage of 100-240 V AC or -48 V DC (DC is optional). The interface card of MX60 uses a RJ-45 socket and is connected to the distribution panel in equipment room using CAT-5 cables supplied with the unit. MX60 offers up to 48 interfaces of FXS/FXO. MX60 supports the following types of configuration.

Table 1-6 Configuration combination of MX60

Models	Number of FXS Ports	Number of FXO Ports
MX60-16S	16	0
MX60-32S	32	0
MX60-48S	48	0
MX60-16FXO	0	16
MX60-32FXO	0	32
MX60-48FXO	0	48
MX60-8S/8	8	8
MX60-24S/8	24	8
MX60-40S/8	40	8
MX60-16S/16	16	16
MX60-32S/16	32	16
MX60-24S/24	24	24

Figure 1-3 MX60 Front Panel

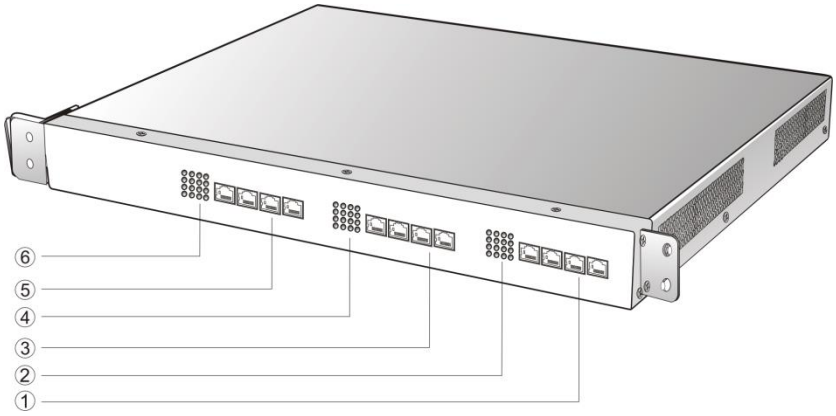


Table 1-7 Description of MX60 Front Panel

#	Description
①③⑤	Three interface slots; each can correspond with four RJ45 sockets; each RJ45socket can correspond with four pairs of analog lines. Note: Numbers of interface slots vary from different configuration.
②④⑥	Matrix of 4 x 4 LED status indicators on interface card.

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-8 Pin Specifications for MX60 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-4 Schematic Diagram of MX60 Subscriber Line Connection

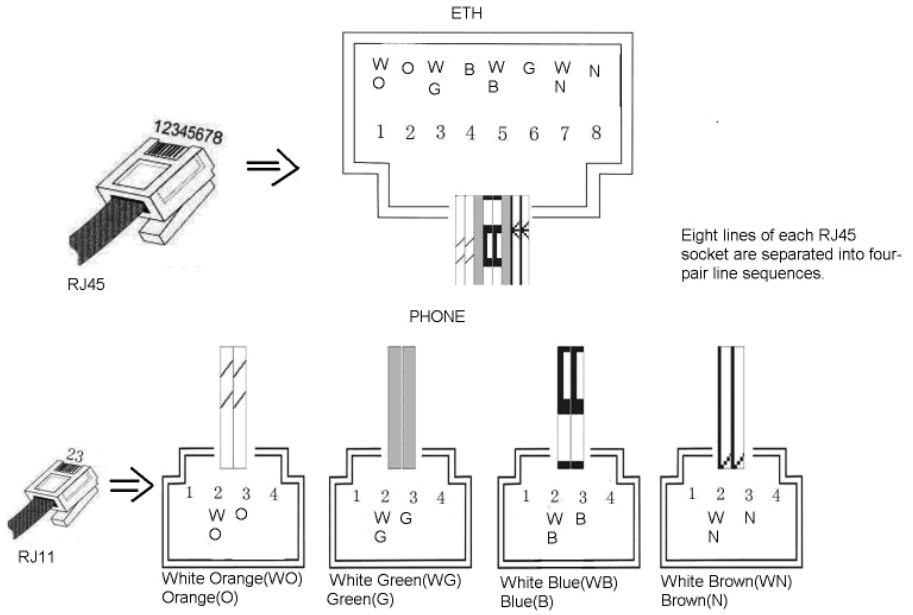


Figure 1-5 MX60 Back Panel-AC

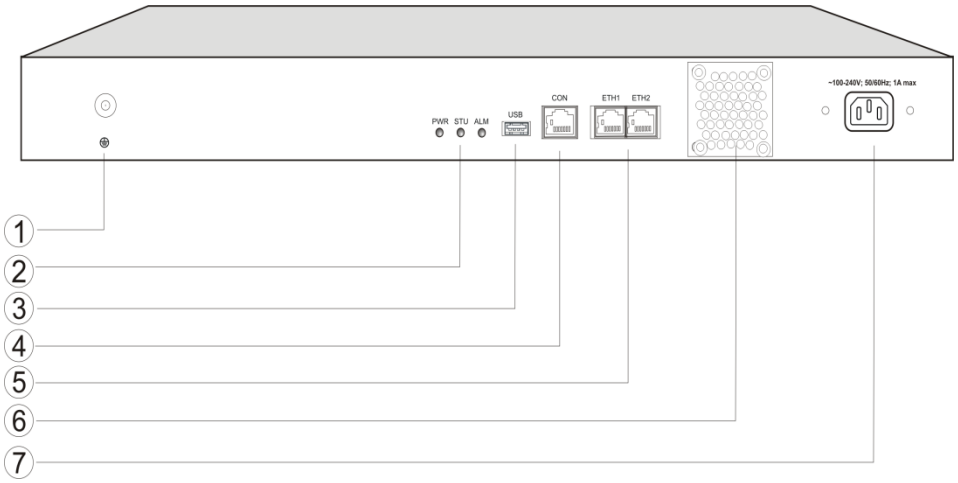


Figure 1-6 MX60 Back Panel-DC

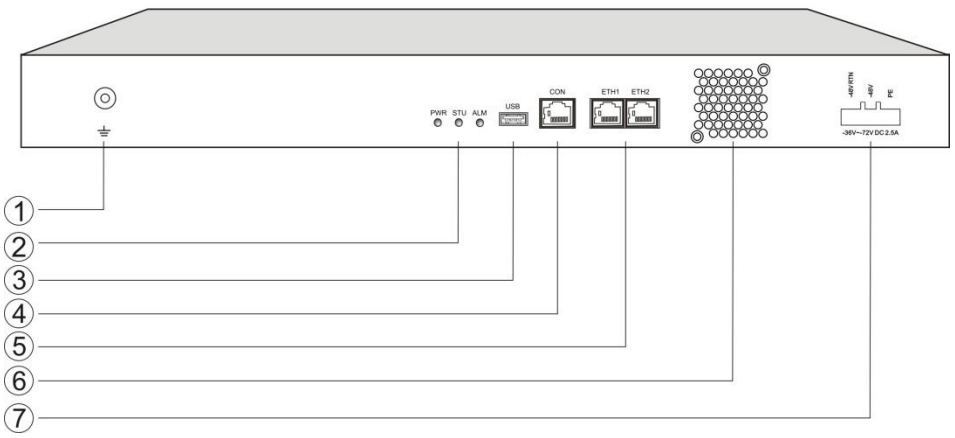


Table 1-9 Description of MX60 Back Panel

#	Description
①	Ground Pole
②	Indicator, see Table 1-10for description
③	USB Interface, reserved for future use
④	Configuration interface (CON), Ethernet lines used for local management and debugging
⑤	Two Ethernet interfaces: one IP address
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input or -48 V DC input.

Table 1-10 Meanings of MX60 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Off	No alarms
		Red Flash	New alarms occurred but not confirmed.
		Red Constant	System in the process of powerup and not in the normal operation mode
		Red	Alarms existed and all alarm information confirmed.

1.3.3 MX120

The device of MX120 can hold four interface cards which enable to flexibly configure FXS and FXO ports. And each card equips up to 24 ports. MX120 can provide up to 96 ports. It supports the following configurations:

Table 1-11 MX120 interface card

Type	FXS Ports	FXO Ports
24FXS	24	0
24FXO	0	24
8FXS/8	8	8
16FXS/8	16	8
12FXS/12	12	12

Table 1-12 Configuration Combination of MX120

Models	Number of FXS Ports	Number of FXO Ports
MX120-72S	72	0
MX120-96S	96	0
MX120-72FXO	0	72
MX120-96FXO	0	96

Models	Number of FXS Ports	Number of FXO Ports
MX120-64S/8	64	8
MX120-88S/8	88	8
MX120-60S/12	60	12
MX120-84S/12	84	12
MX120-56S/16	56	16
MX120-80S/16	80	16
MX120-52S/20	52	20
MX120-76S/20	76	20
MX120-48S/24	48	24
MX120-72S/24	72	24

Figure 1-7 MX120 Front Panel

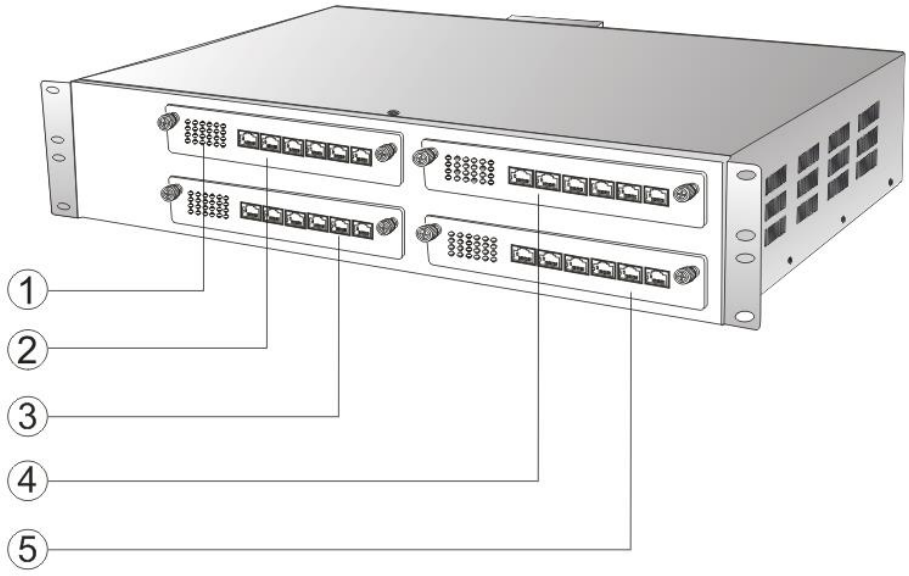


Table 1-13 Description of MX120 Front Panel

#	Description
①	Matrix of 6x4 LED status indicator on interface card
②③④⑤	Four interface slots; each can contain one 24-port interface card.

Note

Do not plug and remove the interface cards of MX120 when equipment is powered on.

Numbering definition of system interface slots: on the low-left side of chassis is #1 slot (marked with No.1 to 24), on the low-right side of chassis is #2 slot (marked with No.25 to 48), on the up-left side of chassis is #3 slot (marked with No.49 to 72), and on the up-right side of chassis is #4 slot (marked with No.73 to 96).

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation.

Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-14 Pin Specifications for MX120 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-8 Schematic Diagram of MX120 Subscriber Line Connection

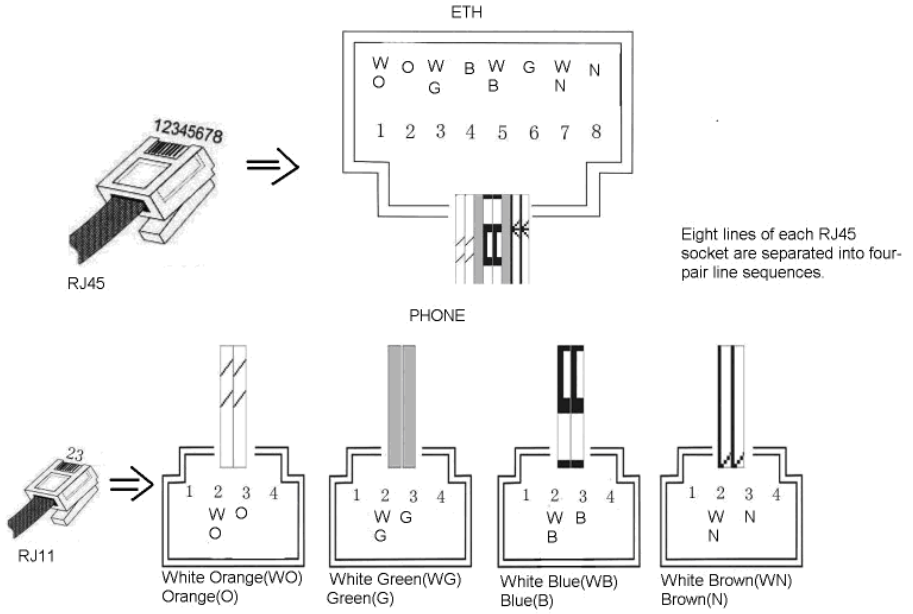


Table 1-15 Corresponding Relation Between MX120 RJ45 Socket and Line Number

RJ45 Socket No. (From Left to Right)	1	2	3	4	5	6
Line No. of This Card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

There is a 6 × 4 LED indicator matrixes on the left side of interface board. Each row of LED indicator matrixes matches four telephone lines on a RJ45. The first row on the left matches Line 1-4 respectively from top to bottom, the first row on the right matches Line 21-24 respectively from top to bottom, and the middle rows in the same manner.

LED indicators are used for multiple purposes as follows

Line status indication: this is the most common mode during normal use of equipment. In this mode, if a line is idle, the indicator corresponding to it goes off; if a line is in call or in use status (such as ringing, offhook) the indicator corresponding to it goes on.

Line type indication: this is the mode for cable wiring check when installing the equipment. This mode can be entered by disconnecting Ethernet cables (Both WAN and LAN ports must be disconnected) at installation stage. After entering this mode, steady on LED indicates that the corresponding line is

equipped as analog subscriber line type, flashing LED indicates that the corresponding line is equipped as analog foreigh exchange line type, off LED indicates that the corresponding line is not equpped or not ready for use.

System operation status indication: this is the mode for displaying information on system operation of equipment in specific conditions. Usually, this mode is entered when some prompts are required to give operator during equipment startup, diagnosis or operation. In this mode, LED flashes to display numbers, letters or other patterns in matrix. Please refer toTable 1-18.

Figure 1-9 MX120 Back Panel-AC

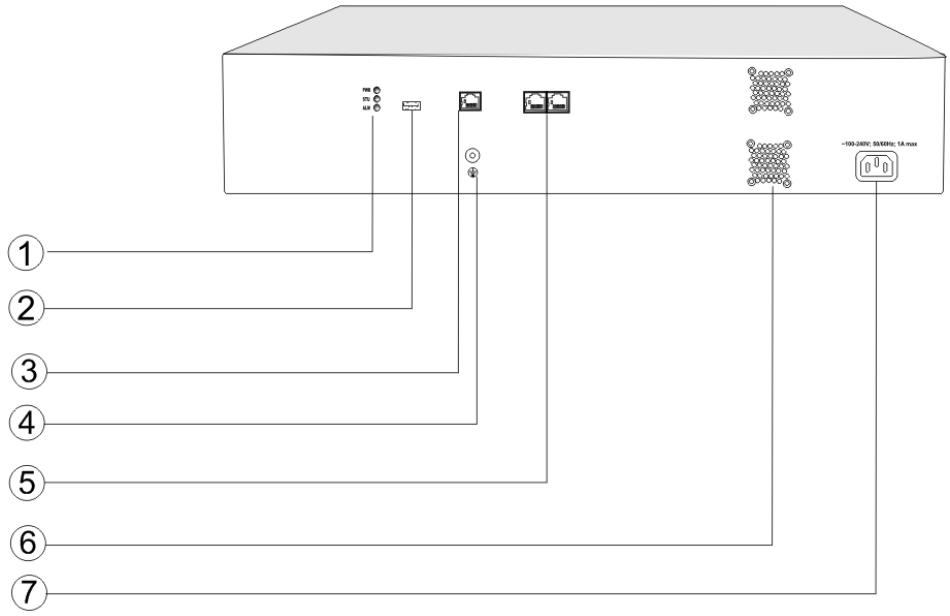


Figure 1-10 MX120 Back Panel-DC

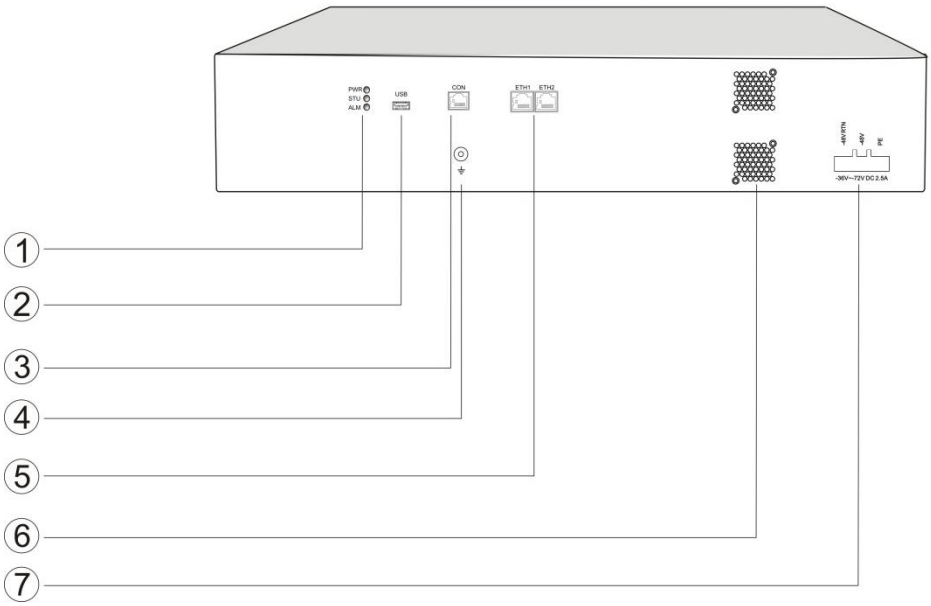


Table 1-16 MX120 Back Panel

#	Description
①	Indicator, see 0 for description.
②	USB interface, reserved for future use.
③	Configuration interface (CON), used for local management and debugging.
④	Ground Pole
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input.

Table 1-17 Meanings of MX120 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Steady red	Power-on is in progress; awaiting normal operation
		Red	Alarms existed and all alarm information confirmed

Table 1-18 MX120 system operation state

Glittery letter	Status meaning
C	The IP address of gateway conflicts with that of other equipment in LAN. Please settle this problem before the gateway can be operated normally.
D	Internal failures have been encountered during gateway start up procedure. Please contact your local distributor for further diagnosis.
P	The gateway is in progress of system software upgrade. Please guarantee stable power supply and do not conduct other operations during this period.
T	The application software of gateway has been exited. If it cannot be restored by rebooting the system, please contact your local distributor for further diagnosis.

2 Parameter Setting

2.1 Login

2.1.1 Obtaining Gateway IP Address

Differ from medium- and large-scale Gateways, MX8 Gateways start DHCP service by default, and automatically obtain an IP address on the LAN; users can use the factory-default gateway IP address if it is unable to be obtained (e.g. when connected directly with a computer).

MX60 and MX120 Gateways use a static IP address by default.

Table 2-1 Default IP Address of Gateway

Type	Default DHCP Service	Default IP Address	Default Subnet Mask
MX8	Enabled	192.168.2.218	255.255.0.0
MX60	Disabled	192.168.2.240	255.255.0.0
MX120	Disabled	192.168.2.240	255.255.0.0

DHCP Used in Network

Users can dial ## to obtain the current gateway IP address and version information of firmware using the telephone connected to the subscriber line (FXS ports) after the equipment is powered on.

If the gateways are only configured with FXO ports for analog trunks without FXS ports for subscriber lines (e.g. MX8-4FXO), users can dial into the gateway by connecting a PBX extension line or PSTN POTS line to a FXO port, and press ## to obtain the current gateway IP address and version information of firmware after receiving the second dial tone.

Note: The access mode of the FXO port must be configured as second stage dialing mode. For details, see section 2.5.4 Trunk Line Features.

Fixed IP Address Used

If the DHCP service on the network is not available or the gateway is directly connected with a computer, the gateways will use the factory-default IP address.

A user could fail to log in with the default IP address if the IP address of user's computer and the default gateway IP address are not at the same network segment. It is recommended that the IP address of user's computer is changed to be identical with the same network segment of the gateway. For example, if the gateway IP address is 192.168.2.218, it is recommended to set the computer's IP address to any address at the network segment of 192.168.2.XXX).

PPPoE Used

In **Basic** > **Network**, the gateways will automatically obtain the WAN address returned by the access network after the PPPoE service is started and the user name and password are set. Users can dial ## on

the gateways to receive the IP address and version of the firmware.

2.1.2 Logging On

Enter the gateway IP address in the browser address bar (eg. 192.168.2.218); you can enter the login interface for gateway configuration by entering a password on the login interface. Both Chinese and English Languages are provided for the Web interface.

Figure 2-1 Login Interface for MX Gateway Configuration

2.1.3 Permission of Gateway Administrator

Logon users are classified into **administrator** and **operator**. The default password is shown in Table 2-22. The password is shown in a cipher for safety.

Table 2-2 Default Passwords of Gateway

Type	Default Administrator Passwords (lowercase letters required)	Default Operator Password
MX8	mx8	operator
MX60	mx60	operator
MX120	mx120	operator

The administrator can browse and modify all configuration parameters, and modify login passwords.

The operator can browse and modify a subset of the configuration parameters.

The gateways allow multiple users to log in:

If both an administrator and operator have logged in, the administrator may modify the configuration, while the operator is limited to browsing;

When multiple users with the same level of permission log in, the first may modify, while the others may only browse.

Note

- The system will confirm timeout if users do not conduct any operation within 10 minutes after login. They are required to log in again for continuing operations.
- Upon completion of configuration, click the **Logout** button to return to the login page, so as not to affect the login permission of other users.
- To ensure system security, please choose **Tools** > **Change password** and change the password when you log in for the first time. For details, see section 2.9.1 Change Password.

2.2 Buttons Used on Gateway Management Interface

Submit buttons are at the bottom of the configuration screens.

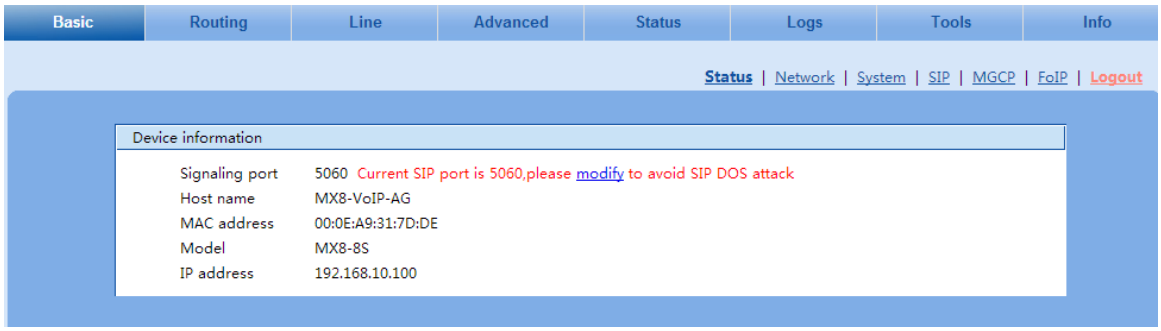
Submit Button: submit configuration information. Users click Submit button after completion of parameter configuration on a page. A success prompt will appear if configuration information is accepted by the system; if a The configuration takes effect after the system is restarted dialog box appears, it means that the parameters are valid only after a system restart; it is recommended that users press the Restart button on the Tool page to enable the configuration after changing all parameters to be modified.

2.3 Basic Configuration

2.3.1 Status

After login, open the **Basic** tab page to view device information. When the SIP port of the device is 5060, you are advised to modify it.

Figure 2-2 Status Interface



2.3.2 Network

After login, click **Basic** > **Network** tab to open the configuration interface.

Figure 2-3 Network Configuration Interface

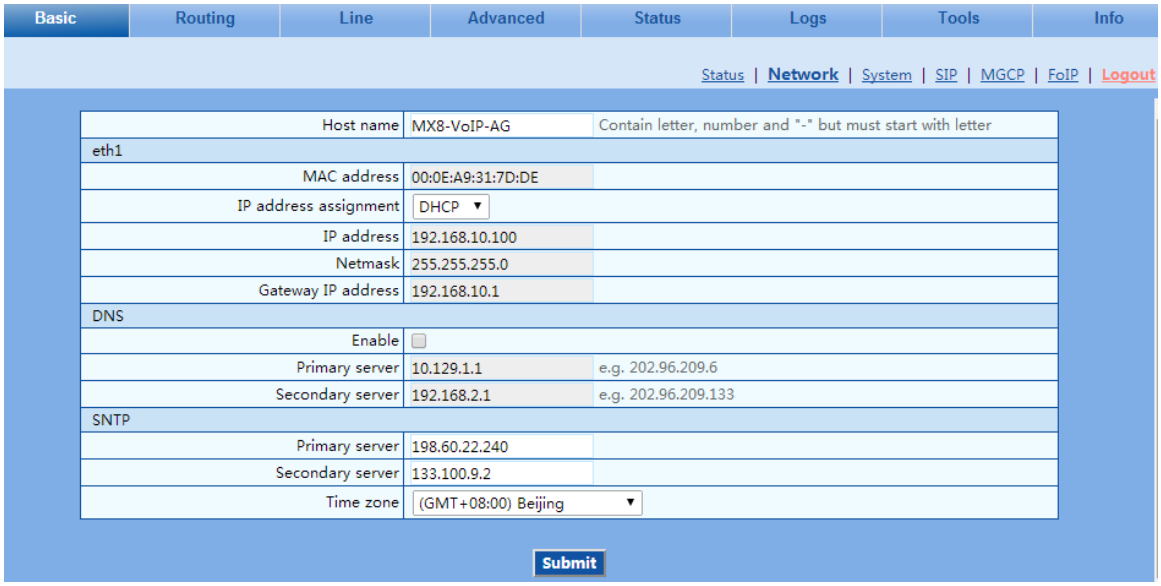


Table 2-3 Network Configuration Parameters

Name	Description
Host name	This is the equipment name of the gateway. The default values of MX8, MX60 and MX120 are MX8-VoIP-AG, MX60-VoIP-AG and MX120-VoIP-AG respectively. Users can set a different name for each gateway when multiple gateways are on the same subnet. A host name can be a maximum of 48 characters, either letters (A-Z or a-z), numbers (0-9) and minus sign (-). It may not include a null or space, and it must start with a letter.
ETH1	
MAC address	Display the MAC address of the gateway.
IP address assignment	Methods for obtaining an IP address • Fixed: static IP address is used; • DHCP: use the dynamic host configuration protocol (DHCP) to allocate IP addresses and other network parameters; • PPPoE: PPPoE service is used.
User name	Enter an authentication user name if PPPoE service is selected, and there is no default value.
Password	Enter an authentication password if PPPoE service is selected, and there is no default value.
IP address	If "Fixed" or "DHCP" is selected for the network type but an address fails to be obtained, the gateways will use the IP address filled in here. If the gateways obtain an IP address through DHCP, the system will display the current IP address automatically obtained from DHCP. This parameter must be set due to no default value.
Netmask	The subnet mask is used with an IP address. When the gateways uses a static IP address, this parameter must be entered; when an IP address is automatically obtained through DHCP, the system will display the subnet mask automatically obtained by DHCP. This parameter must be set due to no default value.
Gateway IP address	The IP address of LAN gateway. When the gateways obtain an IP address through DHCP, the system will display the LAN gateway address automatically obtained through DHCP. This parameter must be set due to no default value.
DNS	
Enable	Activate DNS service.
Primary Server	If DNS service is activated, the network IP address of the preferred DNS server must be entered, and there is no default value.
Secondary Server	If DNS service is activated, the network IP address of a standby DNS server can be entered here. It is optional and there is no default value.

Name	Description
SNTP	
Primary Server	Enter the IP address of preferred time server here. This parameter must be set due to no default value.
Secondary Server	Enter the IP address of standby time server here. This parameter must be set due to no default value.
Time Zone	Select a time zone, and the parameter values include: • (GMT-11:00) Midway Island • (GMT-10:00) Honolulu. Hawaii • (GMT-09:00) Anchorage, Alaska • (GMT-08:00) Tijuana • (GMT-06:00) Denver • (GMT-06:00) Mexico City • (GMT-05:00) Indianapolis • (GMT-04:00) Glace_Bay • (GMT-04:00) South Georgia • (GMT-03:30) Newfoundland • (GMT-03:00) Buenos Aires • (GMT-02:00) Cape_Verde • (GMT) London • (GMT+01:00) Amsterdam • (GMT+02:00) Cairo • (GMT+02:00) Israel • (GMT+02:00) Zimbabwe • (GMT+03:00) Moscow • (GMT+03:30) Teheran • (GMT+04:00) Muscat • (GMT+04:00) United Arab Emirates • (GMT+04:30) Kabul • (GMT+05:30) Calcutta • (GMT+05:00) Karachi • (GMT+06:00) Almaty • (GMT+07:00) Bangkok • (GMT+07:00) Indonesia • (GMT+08:00) Beijing • (GMT+08:00) Taipei • (GMT+08:00) Singapore • (GMT+08:00) Malaysia • (GMT+09:00) Tokyo • (GMT+10:00) Canberra • (GMT+10:00) Adelaide • (GMT+11:00) Magadan • (GMT+12:00) Auckland

2.3.3 System Configuration

After login, click **Basic** > **System** tab to open the configuration interface.

Figure 2-4 System Configuration Interface

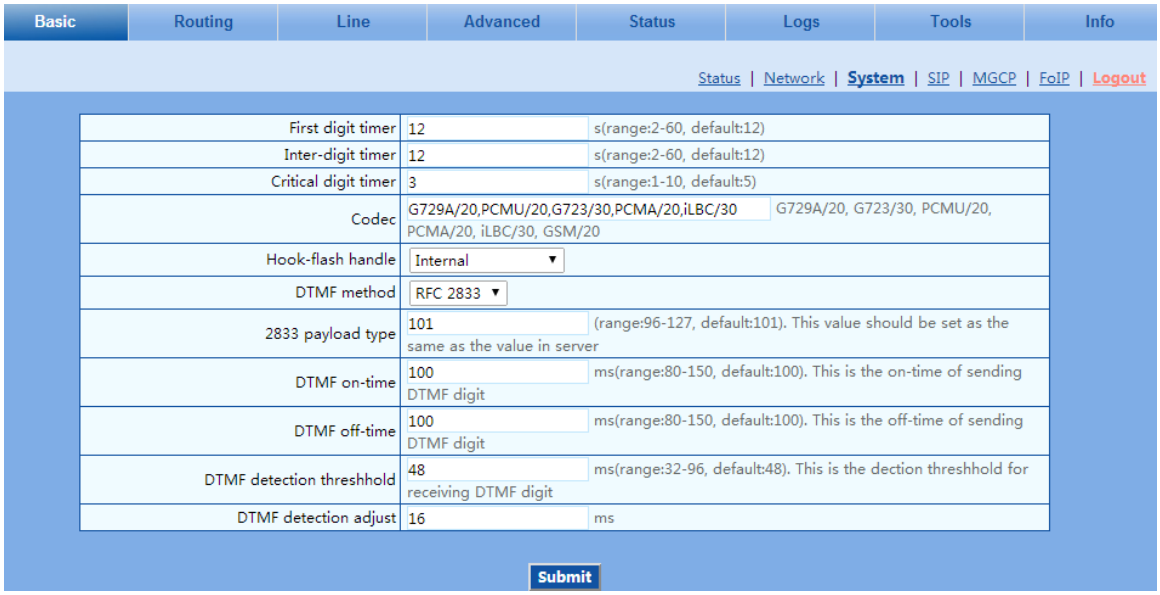


Table 2-4 System Configuration Parameters

Name	Description
First digit timer	If a subscriber hasn't dialed any number within a specified time by this parameter after offhook, the gateways will consider that the subscriber has given up the call and prompt to hang up in busy tone. Unit: second; Default value: 12 seconds.
Inter-digit timer	If a subscriber hasn't dialed the next number key from the time of dialing the last number key to the set time by this parameter, the gateways will consider that the subscriber has ended dial-up and call out the dialed number. Unit: second; Default value: 12 seconds.
Critical digit timer	This parameter is used with the "x.T" rule set in dialing rules. For example, there is "021.T" in the dialing rules table. When a subscriber has dialed 021 and hasn't dialed the next number within a set time by this parameter (eg. 5 seconds), the gateways will consider that the subscriber has ended dial-up and call out the dialed number 021. Input integers, not decimal fractions Unit: second; Default value: 5 seconds.
Codec	Codecs supported by MX include G729A/20, G723/30, PCMU/20, PCMA/20, iLBC/30 and GSM/20. This parameter must be set due to no default value. Several encoding methods can be configured in this item at the same time, separated with "," in the middle; the gateways will negotiate with the platform in the order from front to back when configuring the codec methods.
Hook-flash handle	The gateways provide the following processing modes after detecting hook flash from subscriber terminals: • Internal: the hook flash event will be handled internally; • Server(RFC 2833): transmitting the hook flash to platform with RFC 2833; • Server (SIP INFO): transmitting the flash-off to platform with SIP INFO.

Name	Description
DTMF method	Transmission modes of DTMF signal supported by the gateways include RFC 2833, Audio and SIP INFO. The default value is RFC 2833. • RFC 2833: separate DTMF signal from sessions and transmit it to the platform through RTP data package in the format of RFC2833; • Audio: DTMF signal is transmitted to the platform with sessions; • SIP INFO: separate DTMF signal from sessions and transmit it to the platform in the form of SIP INFO messages.
2833 payload type	Used with "RFC 2833" in the DTMF transmission modes. The default value of 2833 payload type is 101. The effective range available: 96 ~ 127. This parameter should match the setting of far-end device (eg. platform).
DTMF on-time	This parameter sets the on time (in ms) of DTMF signal sent from FXO port. The default value is 100 ms. The duration time range is 80 ~ 150 ms.
DTMF off-time	This parameter sets the off time (ms) of DTMF signal sent from FXO port. The default value is 100 ms. The interval time range is 80 ~ 150 ms.
DTMF detection threshold	Minimum duration time of effective DTMF signal. Its effective range is 32-96 ms. The default value is 48 ms .The greater the value is set, the more stringent the detection is.
DTMF detection adjust	Increase the value can prevent false detection of DTMF signal. The valid values are 16, 32, and 48 in million seconds.

Table 2-5 Codec Methods Supported by Gateways

Codec	Bit Rate (Kbit/s)	Time Intervals of RTP Package Sending (ms)
iLBC	13.3/15.2	20/30
GSM	13	20
G729A	8	10/20/30/40
G723	5.3/6.3	30/60
PCMU/PCMA	64	10/20/30/40

2.3.4 SIP Configuration

After login, click **Basic** > **SIP** tab to open the SIP configuration interface.

Figure 2-5 SIP Configuration Interface

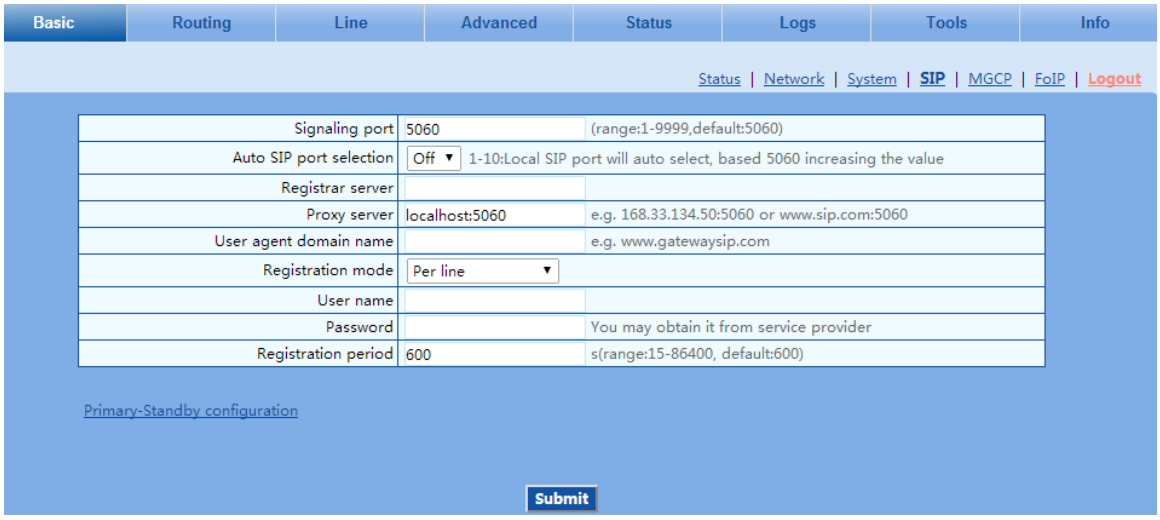


Table 2-6 SIP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving SIP messages, with its default value 5060. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.
Auto SIP port selection	If "n"(ranked from 1-10) is chosen, after the failure registration of signaling port's original configuration, the range of signaling port's change varies from "original signaling port, original signaling port +n". Register with the new signaling port value (signaling port +1) until it succeeds.
Register server	Configure the address and port number of the SIP registration server. The address and port number are separated by ":". It has no default value. The register server address can be an IP address or a domain name. When a domain name is used, you must activate DNS service and configure DNS server parameters on the network-configuration page. E.g. 168.33.134.51:5000 or www.sipproxy.com:5000.
Proxy server	Configure the IP address and port number of the SIP proxy server. The address and port number are separated by ":". There is no default value. The proxy server address can be set to an IP address or a domain name. When a domain name is used, you must activate DNS service and configure DNS server parameters on the network-configuration page. For example: **168.33.134.50:5060** or **www.sip.com: 5060**.
User agent domain name	This domain name will be used in INVITE messages. If it is not set here, the gateways will use the IP address or domain name of the proxy server as the user-agent domain name. It has no default value. It is recommended that subscribers not use LAN IP address to set the domain name parameter.
Authentication mode	The gateway supports three registration schemes: register per line, register per gateway and Line Reg/GW Auth. The default value is register by line. • Register by line: authentication and register per line; • Register by gateway: authentication and register per gateway; • Line Reg/GW Auth: register per line, but authentication per gateway.
User name	Configure the user name as part of the account for registration, and it has no default value. Note: If "Register by gateway" or "Line Reg/GW Auth", is selected, the user name must be entered here. If "register by line" is selected the user name should be set on "Line > Feature" page (Refer to "Feature").
Password	Password as part of account information is used for authentication by platform. It has no default value. It is formed with either numbers or characters, and case sensitive. Note: If "Register by gateway" or "Line Reg/GW Auth", is selected, the password must be entered here. If "register by line" is selected the password should be set on "Line > Feature" page (Refer to "Feature").
Registration period	Valid time of SIP re-registration in seconds. Its default value 600.

2.3.5 High Availability Configuration

After login, Choose **Basic** > **SIP** to open the configuration interface. For details, see section 3 Appendix: High Availability Configuration.

Figure 2-6 High availability configuration

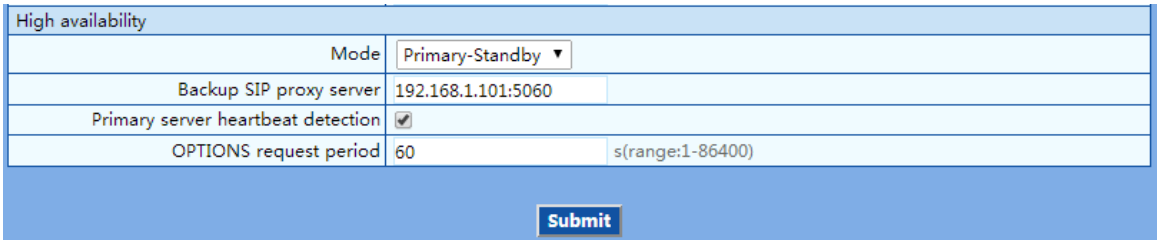


Table 2-7 Parameters

Parameter	Description
Mode	**High availability** can be configured as Primary-Standby, Active-Standby or Load Balancing mode.
Primary-Standby	
Backup SIP proxy server	Configure the address and port number of the backup SIP proxy server. When the primary SIP server faults, the gateway failovers from the primary server to the backup server automatically.
Primary server heartbeat detection	Select it to send OPTIONS request to the primary SIP server all the time. • If the gateway does not receive any response to OPTIONS request, it failovers to the backup server. • After failover to the backup server, the gateway will still send OPTIONS to the primary server. It switches back to the primary server once the response to the OPTIONS request is received.
OPTIONS request period	The interval between receiving the response (200) from the SIP server to the previous OPTIONS and sending the next OPTIONS.
Active-Standby	
SIP proxy server setting	A maximum of five servers can be added.
Fault condition	• No response to OPTIONS request • No response to REGISTER/INVITE request
OPTIONS request period	The interval between receiving the response (200) from the SIP server to the previous OPTIONS and sending the next OPTIONS.
OPTIONS request timeout	The period since the sending of the last OPTIONS with no response by the SIP server.
Active SIP server	This parameter displays the current SIP server address.
Switchover	If you click **Switchover**, the gateway performs switchover to the next available server in sequence based on the SIP server list. For details about how to configure SIP server, please see 3.3.2 “Configuring SIP Server”.
Load balancing	
SIP proxy server setting	A maximum of five SIP servers can be added.
OPTIONS request period	The interval between receiving the response (200) from the SIP server to the previous OPTIONS and sending the next OPTIONS.
OPTIONS request timeout	The period of time since the sending of the last OPTIONS with no response by the SIP server.
REGISTER request timeout	The period of time from the sending of the first REGISTER with no response by the previous SIP server to the sending of REGISTER to the next SIP server.
Active server list	This parameter displays all SIP server IP addresses on which the gateway is registered.

2.3.6 MGCP Configuration

The gateways use SIP protocol by default. When the gateways need to interface with MGCP protocol -based softswitch platform, set the relevant parameters here.

After login, click **Basic > MGCP** tab to open the configuration interface.

Figure 2-7 MGCP Configuration Interface

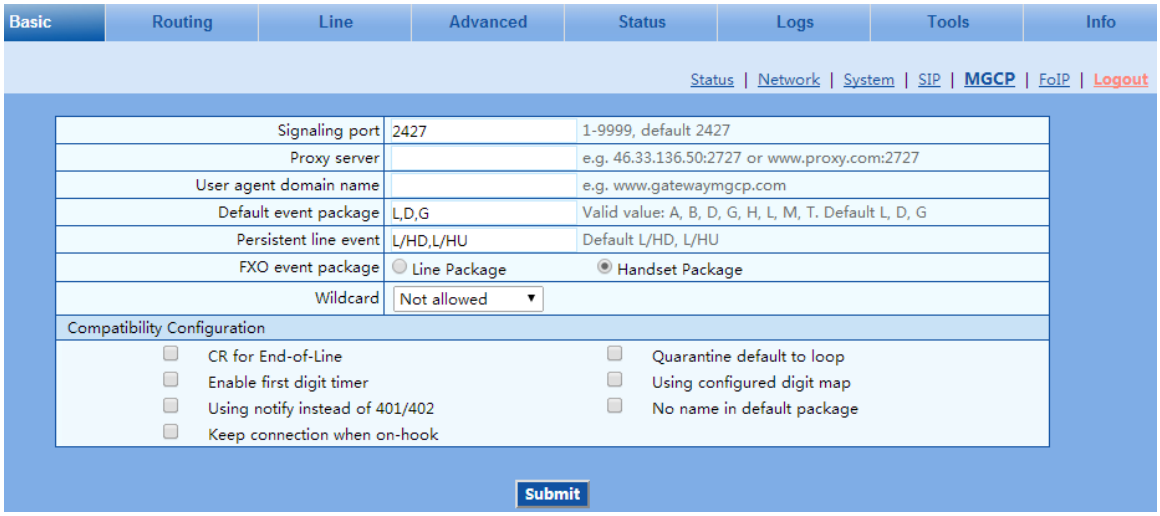


Table 2-8 MGCP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving MGCP messages, and the default value is 2427. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.
Proxy server	Configure the IP address and port number of MGCP proxy server, separated by ":", and it has no default value. The address can be set to an IP address or a domain name according to the subscribers' requirements. When a domain name is used, it is required to activate DNS service and configure DNS server on the page of configuring network parameters. Examples of complete and effective configuration: **46.33.136.50:2727** or **www.proxy.com: 2727**.
User agent domain name	The domain name associated with the call agent, and it has no default value. Example: www.gatewaymgcp.com.
Default event package	List all the types of default event packages supported by the HX4. Multiple package names are separated by",". The default value is L, D, G • L: Line Package • D: DTMF Package • G: Generic Media Package
Persistent line event	List the event types that the gateway can report, with multiple types separated by ",". When gateways process the events listed here, they will report to the call agent. Note: This parameter must be set since there is no default value. The factory setting is L/HD, L/HU: • L/HD: Offhook • L/HU: Onhook
FXO event package	• Handset Package • Line Package

Name	Description
Wildcard	Select whether a wildcard with prefix is allowed when a gateway registers to the proxy server. The default value is **not allowed**. • Partially allowed: gateways will use a wildcard with fixed prefix (e.g. aaln / *) when registering. For example, when configuring telephone numbers, if line 1 is set to **aaln/1**, line 2 is set to **aaln/2** and line 3 is set to **aaln/3**, the gateways will register to the call agent in **aaln/*** without the need of registering the lines individually. • Allowed: the gateways will use a wildcard in registering without prefix.
Compatibility Configuration	
CR for End-of-Line	Select whether CR is used as the end of line in the MGCP messages. Default not selected.
Quarantine default to loop	Select the Quarantine handle of gateways making a request to the outside, and default not selected. • Selected: quarantine using loop mode, the gateways will continually notify all events as requested after receiving a request.
Enable first digit timer	Select the processing mode when there is no timeout parameter in the outside request received by the gateways, and default not selected. • Selected: the gateways will report timeout in terms of its own timeout setting (the time interval set in non-dial timeout of configuration system parameters) when subscribers hasn't dialed up in time after offhook.
Using configured digit map	Select whether to activate the digit map configured by local gateway, and default value is not selected.
Using notify instead of 401/402	Set whether the gateways report "offhook events" to replace 401 messages in NTFY or report "onhook events" to replace 402 messages in NTFY when responding to messages sent by the proxy server. Default: not selected. • Selected: the gateways will use NTFY message to replace 401 and 402 messages.
No name in default package	Select if a package name is included when the gateways reply to the default package, and default not selected.
Keep connection when on-hook	Select if the gateways actively cancel connection disconnect when subscribers hook on, and default not selected.

2.3.7 FoIP

After login, click the label of **Basic > FoIP** to open this interface.

Figure 2-8 Fax configuration interface

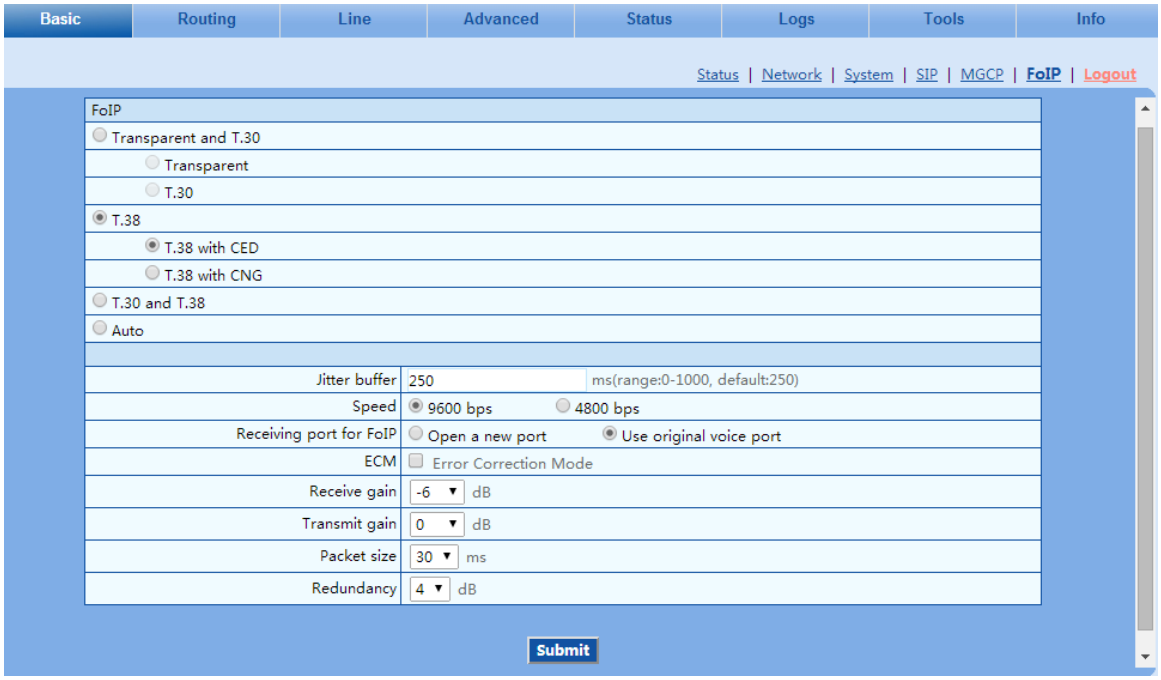


Table 2-9 Fax configuration parameters

Name	Description
Transparent and T.30	• Transparent: Transparent: in this mode, the voice coding mode must be set to G.711; otherwise the facsimile service fails. • T.30: in this mode, the media coding is negotiated through re-invite.
T.38	• T.38 with CED • T.38 with CNG
T.30 and T.38	Both T.30 and T.38 Note: When received fax signaling carries both T.30 and T.38 media, preferentially use T.38.
Auto	Note: When received fax signaling carries both T.30 and T.38 media, preferentially use T.30.
Jitter buffer	Set the extent of T.38 jitter buffer, and the default is 250. The valid range is 0~1000 in milliseconds.
Speed	Set the maximum fax transmission rate: • 4800: negotiates the transmission rate first in accordance with the V.27. The maximum value is 4800 bps. • 9600 (default value): negotiates the transmission rate first in accordance with the V.29. The maximum value is 9600 bps. • 14400: negotiates the transmission rate first in accordance with the V.17. The maximum value is 14400 bps. You need to modify it only when the negotiation peer requires. Otherwise, keep it the default value.
Receiving port for FoIP	Set whether to open a new port when the gateway is switching to T.38 mode, and by default, original voice port will be used. Open a new port: use the new RTP port. Use the port of the original voice call: use the original RTP port that created on call set.
ECM	Determine whether to use corrective mode of fax. By default, it is not selected.
Receive gain	Set the receiving gain of T.38 fax, with the default of 6dB.

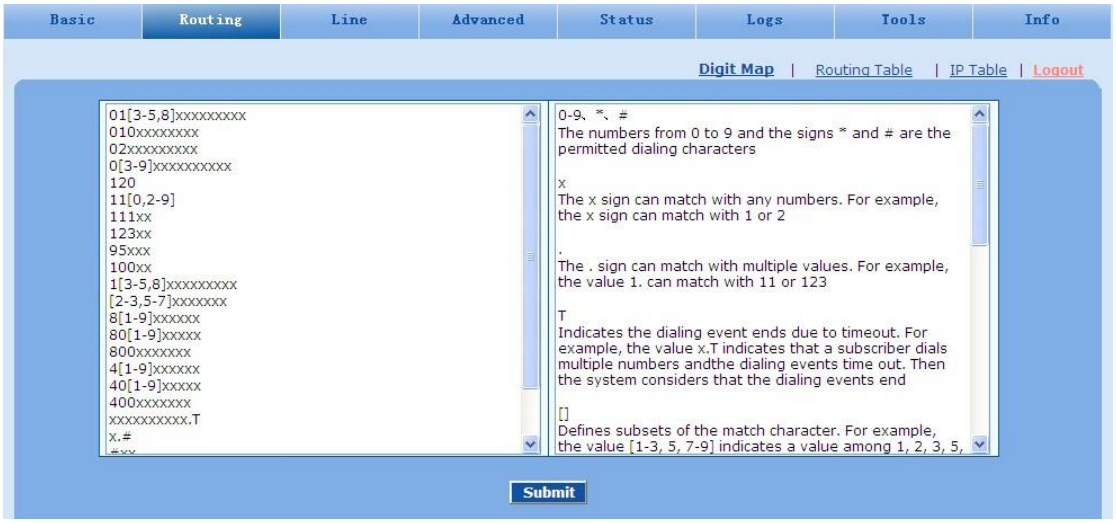
Name	Description
Transmit gain	Set the transmission gain of T.38 fax, with the default of 0dB.
Packet size	Set the packet size of T.38. 30 milliseconds is the default value.
Redundancy	Set the number of the redundant frames in T.38 date package, default is 4.

2.4 Routing

2.4.1 Digit Map

After login, click **Routing > Digit Map** tab to open the dialing rules interface.

Figure 2-9 Configuration Interface for Dialing



Dialing rules are used to effectively judge if the received number sequence is completed, for the purpose of terminating receiving numbers and sending received numbers. The proper use of dialing rules can help to reduce the connection time of telephone calls.

The maximum number of rules that can be stored in gateways is 250. Each rule can hold up to 32 numbers and 38 characters. The total length of dialing rules table (the total length of all dialing rules) can be up to 2280 bytes.

The following provides descriptions of typical rules:

Table 2-10 Description of Dialing

Digit map	Description
x	Represents any number between 0-9.
.	Represents more than one digit between 0-9.
##	## is a special dial string for users to receive gateway IP address and version number of firmware by default.
x.T	The gateways will detect any length of telephone number starting with any number between 0-9. The gateways will send the detected number when it has exceeded the dialing-end time set in system parameter configuration and hasn't received a new number.

Digit map	Description
x.#	Any length of telephone number starting with any number between 0-9. If subscribers press # key after dial-up, the gateways will immediately terminate receiving digits and send all the numbers before # key.
*xx	Terminate after receiving * and any two-digit number. *xx is primarily used to activate function keys for supplementary services, such as CRBT, Call Transfer, Do not Disturb, etc.
#xx	Terminate after receiving # and any two-digit number. #xx is primarily used to stop function keys for supplementary services, such as CRBT, Call Transfer, Do not Disturb, etc.
[2-8]xxxxxx	A 7-digit number starting with of any number between 2- 8, used to terminate the dialing.
02xxxxxxxxx	An 11-digit number starting with 02, used to terminate the long-distance dial string starting with 02.
013xxxxxxxxx	A 12-digit number starting with 013, used to terminate long-distance dial strings
13xxxxxxxxx	An 11-digit number starting with 13, used to terminate long-distance dial strings.
11x	A 3-digit number starting with 11, used to terminate the dial string of emergency calls.
9xxxx	A 5-digit number starting with 9, used to end special service calls.
17911 (e.g.)	Send away when the set number, like 17911, is received.

Dial rules by default as follows:

01[3-5, 8] xxxxxxxxx

010xxxxxxxx

02xxxxxxxxx

0[3-9] xxxxxxxxxx

120

11[0, 2-9]

111xx

123xx

95xxx

100xx

1[3-5, 8] xxxxxxxxx

[2-3, 5-7] xxxxxxx

8[1-9] xxxxxx

80[1-9] xxxxx

800xxxxxxx

4[1-9] xxxxxx

40[1-9] xxxxx

400xxxxxxx

xxxxxxxxxx.T

x.T

x.#

#xx

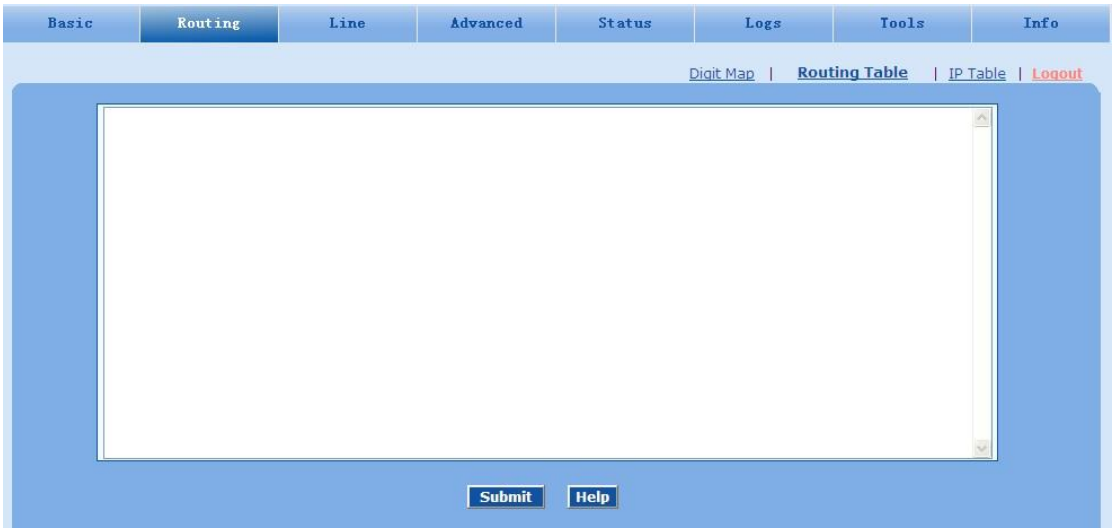
*xx

##

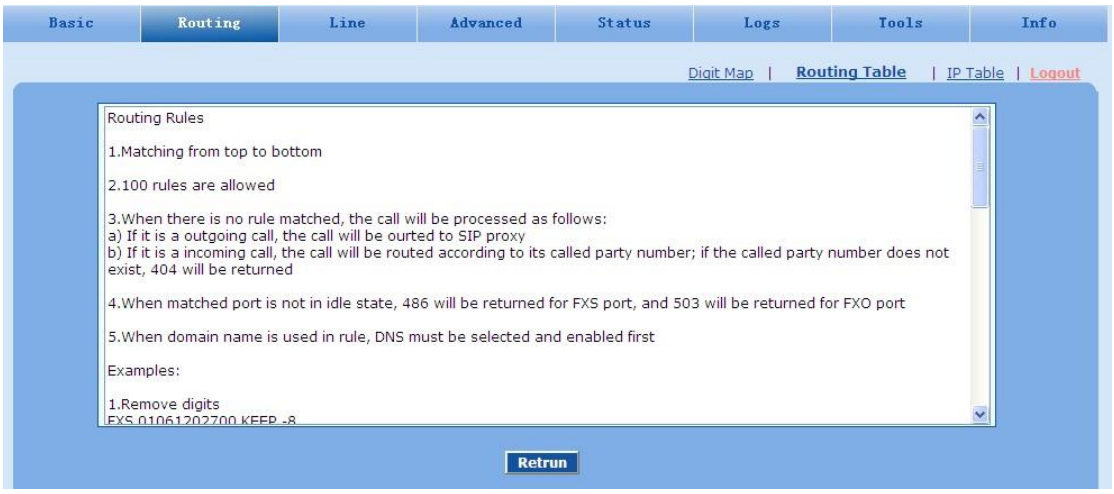
2.4.2 Routing Table

After login, click **Routing** > **Routing Table** tab to open the configuration interface.

Figure 2-10 Configuration Interface for Routing Table



Click **Help** to open the illustrative interface for routing configuration.



The routing table with 500 rules in capacity provides two functions including digit transformation and call routing assignment. Here are the general rules applied by gateways when executing the routing table.

Note

- Rules must be filled out without any blank at the beginning of each line; otherwise the data can't be

validated even if the system prompts successful submittal.

- The routing table is empty by default. The gateways will point a call to the SIP proxy server when there is no matched rule for the call.

The format of number transformation is

Source Number Replacement Method

For example: **FXS 021 REMOVE 3** means remove the prefix 021 of the called number for calls from the FXS port, where **FXS** is source, **021** is number, and **REMOVE 3** indicates the method of number transformation.

The format of routing rules is

Source Number ROUTE Routing Destination

For example: **IP 800[0-1] ROUTE FXO 1-2** means route calls from IP with called number between 8000~8001 to FXO port in a sequential selecting order of 1, 2. Namely, FXO Port 2 is selected when FXO Port 1 is busy and so on.

Detailed definitions of source and number, number transformation methods and routing destination are shown below.

Table 2-11 Routing Table Format

Name	Description
Source	There are three types of source: IP, FXS (Phone/fax) and FXO (Line). Among them, IP source can be any IP address and is denoted by **IP**; **IP [xxx.xxx.xxx.xxx]** is used to denote a specific IP address; **IP [xxx.xxx.xxx.xxx: port]** is used to denote specific IP address with port number. FXS and FXO ports can be any port, represented with **FXS** or **FXO**; special lines can be represented with FXS or FXO plus the port number, e.g. **FXS1, FXO2** or **FXS [1-2]**, etc.
Number	It could be a calling party number with the form of CPN + number, such as CPN6034340633 or a called party number with the form of number. The number may be denoted with digit 0-9,"*",".","#"," x ", etc., and uses the same regular expression as that of dialing rules. Here are examples of the form of number: • Designate a specific number: eg.**114**, or **61202700** • Designate a number matching a prefix: such as **61xxxxxx**. Note: The matching effect of **61xxxxxx** is different from that of **61x** or **61**. Number matching follows the principle of minimum priority matching • Specify a number scope. For example, **268[0-1, 3-9]** specifies any 4-digit number starting with 268 and followed by a digit between 0-1or 3-9 Note: Number matching follows the principle of minimum matching. For example: **x** matches any number with at least one digit; **xx** matches any number with at least two-digit; **12x** matches any number with at least 3-digit starting with 12.

Table 2-12 Number Transformations

Processing Mode	Description and Example
KEEP	Keep number. A positive number behind KEEP means to keep several digits in front of the number; a negative number means to keep several digits at the end of the number. Example: FXS 02161202700 KEEP -8 Keep the last 8 digits of the called number 02161202700 for calls from FXS. The transformed called number is 61202700.
REMOVE	Remove number. A positive number following REMOVE means to remove the first several digits of the number; a negative number means to remove the latter several digits of the number. For example: FXS 021 REMOVE 3 Remove 021 of the called number beginning with 021 for calls from FXS.
ADD	Add prefix or suffix to number. A positive number behind ADD is the prefix; a negative number is suffix. Example 1: FXS1 CPNX ADD 021 FXS2 CPNX ADD 010 Add 021 in front of calling numbers for calls from FXS port 1; add 010 in front of calling numbers for calls from FXS port 2. Note: CPNX denotes to any calling party number. Example 2: FXS CPN6120 ADD -8888 Add 8888 at the end of the calling number starting with 6120 for calls from an FXS (Phone/fax) port.
REPLACE	Number replacement. The replaced number follows REPLACE. Example: FXS CPN88 REPLACE 2682000 Replace the calling number beginning with 88 for calls from FXS port with 2682000.
REPLACE	Another use of REPLACE is to replace the specific number based on another number associated with the call. For example, replace the calling number according to the called number. Examples: FXS 12345 REPLACE CPN-1/8621 FXS CPN13 REPLACE CDPN0/0 For calls from FXS ports with called party number of 1234, remove one digit at the end of the calling number and add 8621; for calls from FXS ports with calling party number starting with 13, add 0 at the beginning of the called number.
END or ROUTE	End-of-number transformation. From top to bottom, number transformation will be stopped when END or ROUTE is encountered; the gateways will route the call to the default routing upon detecting END, or route the call to the designed routing after detecting ROUTE. Example 1: FXS 12345 ADD -8001 FXS 12345 REMOVE 4 FXS 12345 END Add suffix 8001 to the called number starting with 12345 for calls from FXS ports, then remove four digits in front of the number to end number transformation yielding 58001. Example 2: IP [222.34.55.1] CPNX. REPLACE 2680000 IP [222.34.55.1] CPNX. ROUTE FXS 2 For calls from IP address 222.34.55.1, calling party number is replaced by 2680000, and then the call is routed to FXS port 2 with the new calling party number.

Processing Mode	Description and Example
CODEC	Designate the use of a codec, such as PCMU/20/16, where PCMU denotes G.711, /20 denotes RTP packet interval of 20 milliseconds, and /16 denotes echo cancellation with 16 milliseconds window. PCMU/20/0 should be used if echo cancellation is not required to activate. Example: IP 6120 CODEC PCMU/20/16 PCMU/20/16 codec will be applied to calls from IP with called party number starting with 6120.
RELAY	Insert prefix of called party number when calling out. The inserted prefix number follows behind RELAY. Example: IP 010 RELAY 17909 For calls from IP with called party number starting with 010, digit stream 17909 will be outpulsed before the original called party number is sent out.

Table 2-13 Routing Destination

Destination	Description and Example
ROUTE NONE	Calling barring (also known as “blacklist”) . Example: IP CPN[1,3-5] ROUTE NONE Bar all calls from IP, of which the calling numbers start with 1, 3, 4, and 5.
ROUTE FXS	Route a call to FXS ports. Example 1: IP 800[0-3] ROUTE FXS 1-2 Select a port in sequential order. Note: 800[0-3] denotes to the UDP ports ranging from 8000 to 8003. Example 2: IP 800[0-3] ROUTE FXS 1 Point this call to FXS port 1. Example 3: IP 800[0-3] ROUTE FXS 1-2/R Select a port in round robin order Example 4: IP 800[0-3] ROUTE FXS 1-2/G Select all idle ports and provide ringing.
ROUTE FXO	Route a call to FXO port. Example 1: IP x ROUTE FXO 1-2 Select a port in sequential order. Example 2: IP 800[0-1] ROUTE FXO 1-2/R Select a port in round robin order.
ROUTE IP	Route a call to the SIP proxy server Example: FXS 021 ROUTE IP 228.167.22.34:5060 228.167.22.34:5060 is the IP address of the platform.

2.4.3 Application Examples of Routing Table

Some typical functions that can be realized by the routing table are provided in this section (Take

MX8-4S/4 gateway as an example):

1) One Phone with Two Numbers
2) Hunt Group
3) Outbound Call Barring
4) FXO Port Hunting for Outbound Call

One Phone with Double Numbers

A hand set connected to the HX4 can be configured with two numbers through One Phone with Double Numbers. For example, port FXS1 is set with PSTN number 61202701 and extension number 1001 for internal calling

Routing Setting

FXS 1001 ROUTE IP 127.0.0.1:5060

IP 1001 ROUTE FXS 1

Description:

1) Send a call with a called number starting with 1001 from FXS port to port 5060 of gateway's local IP;
2) Send a call with a called number starting with 1001 and from any IP to the FXS port 1.

Configuration number of FXS1 itself is 61202701, so the call of this number is not required to write specialized routing.

Hunt Group

A hunt group can be associated with a set of FXO ports, and an inbound call from IP or FXS ports can be routed to a hunt group.

Routing Setting:

Send an inbound call from the IP trunk or an FXO line in a sequential way to the phone set on the 1st or 2nd FXS port.

FXO x ROUTE IP 127.0.0.1:5060

IP x ROUTE FXS 1-2

Description:

1) Send all calls from the FXO port to port 5060 of gateway's local IP;
2) Send all inbound calls from any IP address to the 1st or 2nd FXS port in sequence. Namely, the first FXS port is selected firstly when it is available otherwise the 2nd port is selected.

Outbound Call Barring

Restrict users to from dialing certain telephone numbers, such as an international call. Examples are as follows:

Routing Setting	Description
FXS[1] 0 ROUTE NONE	A calling starting with 0 is barred from dialing using the phone set at FXS1 port.
FXS[1-2] 00 ROUTE NONE	A calling starting with 00 is barred from dialing at 1-2 FXS ports.

FXS CPN2 ROUTE NONE	The telephone whose calling number starts with 2 at a FXS port is barred to call out.

FXO-Port Hunting for Outbound Calls

Routing Setting:

FXS x ROUTE FXO 1-2

Description:

Send all calls from FXS ports to FXO ports in sequential order.

2.4.4 IP Table

After login, click **Routing** > **IP Table** tab to open the configuration interface.

Figure 2-11 Configuration Interface for IP Table

This table is designed to ensure the safe use of gateways. Administrators can add the authorized IP addresses to this table, and the gateways will only process the information from authorized IP addresses. If the IP table is empty, the gateways will not perform IP address-based message filtering.

For example, the gateway will only process the messages from 202.96.209.133 after adding 202.96.209.133 to its IP table.

If the gateway is deployed in a public network, you are advised to set IP filtering to prevent call theft.

2.5 Line Configuration

2.5.1 FXS Phone Number

After login, click **Line > FXS phone number** tab to open the configuration interface.

Figure 2-12 Configuration Interface for FXS phone number

Table 2-14 Configuration Parameters of FXS phone number

Name	Description
FXS 1st line No.	This number is used for the batch setup of consecutive number of subscriber line. Click **Batch** after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn't fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the subscriber line n (FXS port). This should be manually performed if Batch mode is not used.

2.5.2 FXO Phone Number

After login, click **Line > FXO phone number** tab to open the configuration interface.

Figure 2-13 Configuration Interface for FXO phone number

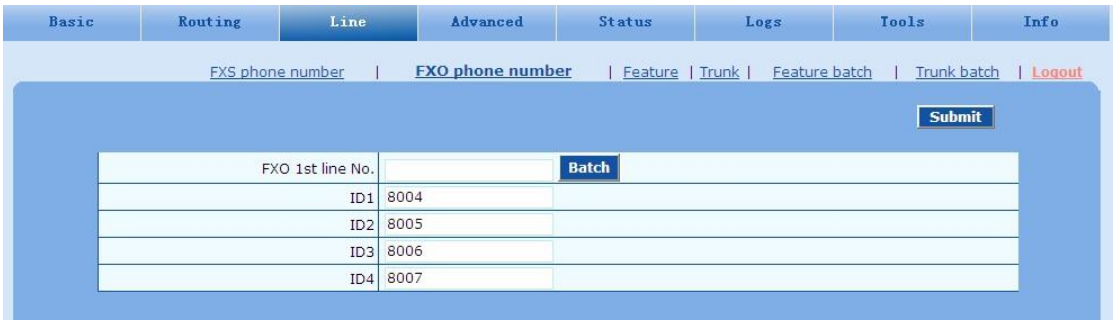


Table 2-15 Configuration Parameters of FXO phone number

Name	Description
FXO 1st line No.	This number is used for the fast setup of consecutive number of trunk line. Click **Batch** after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn't fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the trunk n (FXO port). This should be manually performed if Batch mode is not used.

2.5.3 Subscriber Line Features

This page is only used for configuring gateways with subscriber lines (FXS port).

After login, click **Line > Feature** tab to open the configuration interface.

Figure 2-14 Configuration Interface for Subscriber Line Features

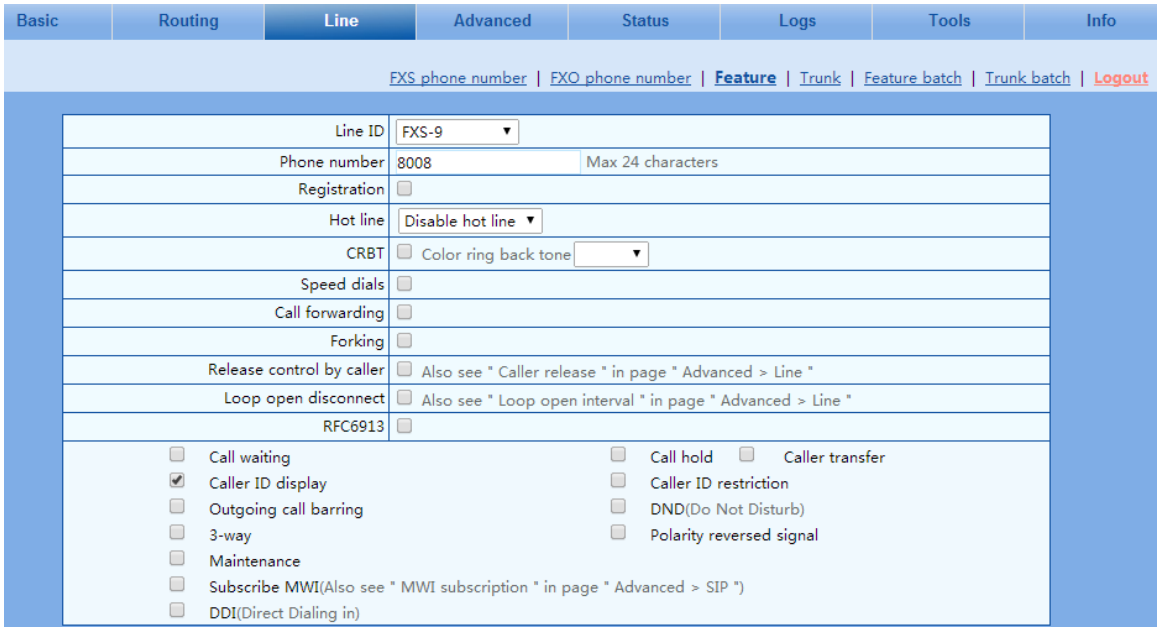


Table 2-16 Configuration Parameters of Phone Features

Name	Description
Phone number	Fill in the phone number associated with this port.
Display name	Fill in the name associated with this port.
Registration	Select if this line is required to register with a softswitch. This is selected as default.
Password	If **Registration** is selected, users must enter the authentication password for registering of this line here.
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, features are controled by the proxy server without the need for setting on the gateway.	
Hot line	Select if the gateway is required to automatically dial out the hotline number after offhook. By default, hot line is disabled. • **Disable** hot line**:** close this feature. • **Hot line:** automatically dial out the hotline number after offhook. • **Delay mode:** automatically dial out the hotline number when the offhook is timeout with a time delay of 5 seconds.
Hot line number	After the hotline function is activated on this line, the hotline number must be entered here.
CRBT(Color ring back tone)	Select it to activate CRBT (Color Ring Back Tone), and choose an audio file as ring back tone. There are two.dat files in the G.729 coding format (fring1.dat and fring2.dat) storage in MX for factory default. You can upload .wav files through the Web GUI, for details, see 2.6.10 Greeting.
Speed dials	Select if the Speed dials is activated on this line. By default, this is not selected.
Call forwarding	Select if Call forwarding is activated on this line. By default, it is not selected.
Forking	Select to activate Forking. Forking allows the gateway to initiate a call to another telephone terminal while ringing on this line terminal. Either terminal may answer, terminating ringing on the other terminal.
Release control by caller	Select if the call release is controlled by the caller. By default, this is not selected. Note: Also see **Caller release** on page **Advanced > Line**. • **Selected:** the gateway will immediately release the call upon caller hanging up; the gateway will not release the call after the called party hanging up as long as the caller is still off-hook until timeout (60 seconds by default); • **Unselected:** the gateway will immediately release the call upon either party hanging up the call.

Name	Description
Loop open disconnect	Select it only if the trunk of the PBX supports loop open signaling, in which the PBX takes the loop open as the indication of disconnection. Note: Loop open interval can be configured on the **Advanced > Line** page.
RFC6913	If this item is selected, the Fax over IP label carried in INVITE is supported.
Call waiting	Select if Call waiting is activated on this line. By default this is not selected.
Call hold	Select it to enable Call Hold on this line. By default this is not selected. Note: If this function is enabled, the gateways will automatically activate Call Transfer.
Caller transfer	Select if Caller Transfer is activated on this line. By default, this is not selected. When A calls B, B picks up the call and A transfers the call to C. Note: The call hold must be activated before caller transfer.
Caller ID display	Set whether the number of this telephone is sent to the called party. This feature requires the support of softswtich. By default this is not selected.
Caller ID restriction	Set whether the number of this telephone is sent to the called party with support from platform. By default this is not selected.
Outgoing call barring	Select if outgoing calls are barred on this line. By default, this is not selected.
DND(Do not disturb)	Select if **Do Not Disturb** is enabled on this line. By default, this is not selected.
DDI (Direct Dialing in)	Set whether DDI (Direct Dialing In) is activated, By default this is not selected. Different from FXS, DDI is only used for incoming calls, and the gateways will not send dial tone after off-hook (calling in) on user side. Note: Reverse polarity signal must be activated on the gateways when DDI is used.
Polarity reversal	Select if reverse polarity signal is activated on this line. By default, this is not selected. Note: The gateways will provide reverse polarity signal when the phone is connected after this feature is activated.
Maintenance	Select if the line is set to maintenance status, in which the FXS port no longer supplies current to the phone. By default, this is not selected.
Subscribe MWI	Select if voice mail service is activated, and by default this is not selected. (Also see **MWI Re-subscription timer** on page **Advanced > SIP**.)
3-way	Select if 3-way service is activated, and by default this is not selected.

2.5.4 Trunk Line Features

This page is only used for configuring gateways with trunks (FXO port).

After login, click **Line > Trunk** tab to open the configuration interface.

Figure 2-15 Configuration Interface for Trunk Line Features

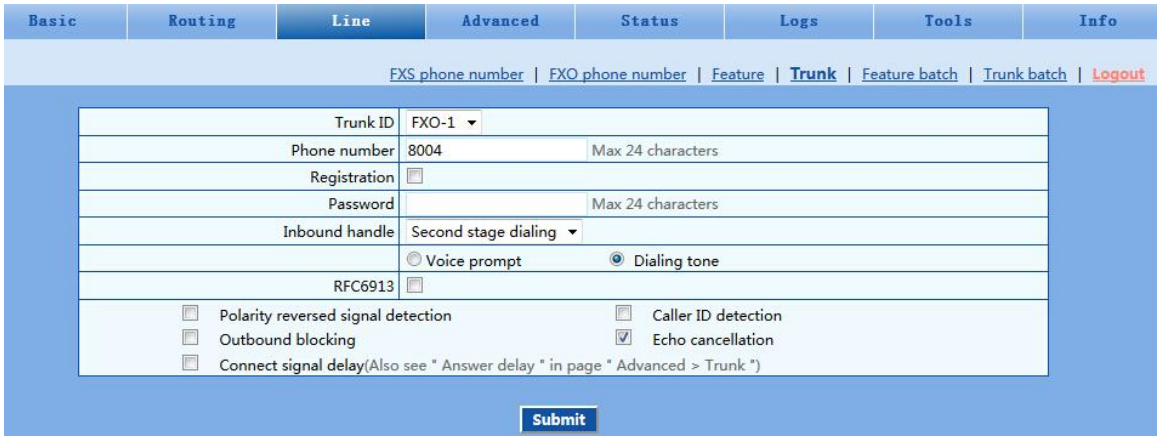


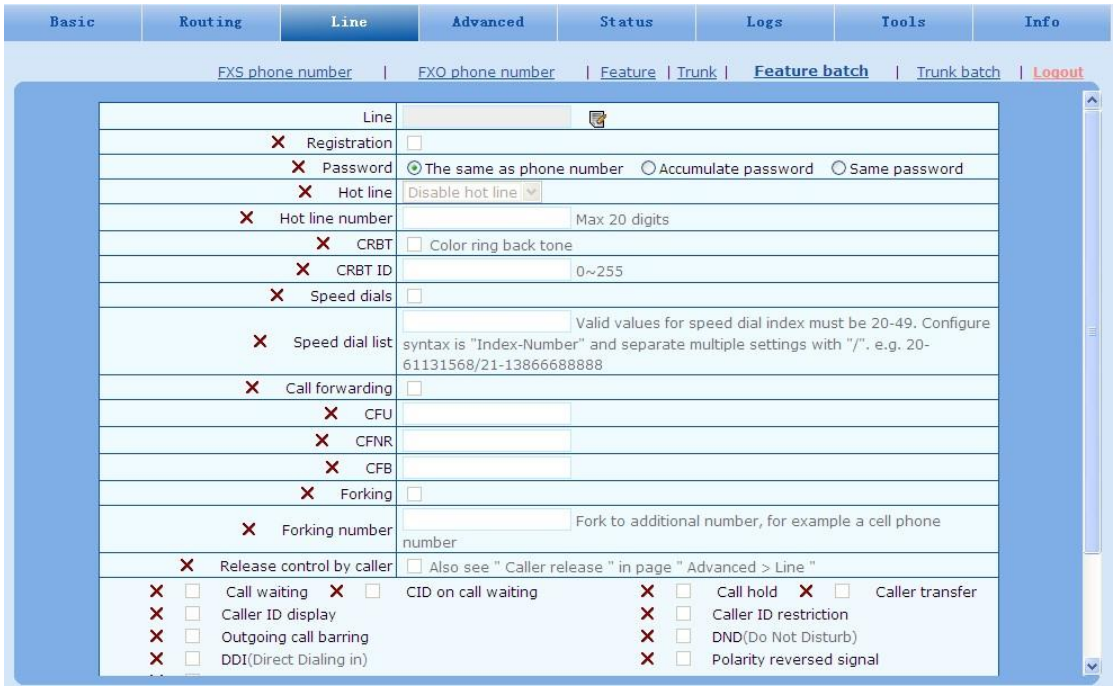
Table 2-17 Configuration Parameters of Trunk Line Features

Name	Description
Trunk ID	Select a trunk line required to configure. “FXO-n” corresponds to the “Line > FXO phone Number ID n”. Copy the configuration of “FXO-n” for selected line to “FXO-n+1”~“FXO- m” by clicking “Batch”, where n indicates the current selected trunk number and m indicates the total number of trunks.
Phone number	Display phone number associated with the trunk set in **Line > FXO phone Number**
Registration	Select if this trunk registers with the SIP registration server. By default, this is not selected.
Password	If **Registration** is selected, the authentication password for register of this line must be entered here.
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, the control of all call services is provided by the proxy server without the need of these setting.	
Inbound handle	The gateways provide three scenarios for handling incoming calls on the FXO trunk: • **Binding:** when a telephone call comes to the FXO port, the gateways will route the call to a FXS port according to the DID number bound with the port. Note: Setting a number to be bound is required or this setting is invalid. • **Second-stage dialing:** when a telephone call comes to the Line port, the gateways will provide the second dial tone and route the call according to the extension number entered. Note: Dialing tone or voice prompt file can be changed by user. • **Direct:** the gateways will route the incoming call on FXO port n to FXS port n.
RFC6913	If this item is selected, the Fax over IP label carried in INVITE is supported per RFC6913
Polarity reversal detection	If a PSTN line supports reverse polarity, make the selection here. By default, this is not selected.
Caller ID detection	Select if the detection function of caller ID for this FXO port is enabled. By default, this is selected.
Outbound blocking	Select if this FXO port bars outgoing call service to the PSTN. By default, this is not selected.
Echo cancellation	Select if echo cancellation is enabled for this FXO (Line).By default, this is selected.
Connect signal delay	After making an outgoing call from a FXO port, the gateway will send a 200 OK message to the platform with a delay if this parameter is selected. If unselected, the system sends a 200 OK message to the platform after off hook on the FXO port. Also see **Answer delay** on page Advanced > Trunk.

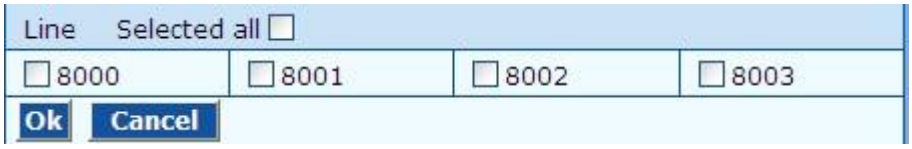
2.5.5 Feature Batch

After login, click **Line > Feature Batch** to open this interface.

Figure 2-16 Feature batch configuration interface



Step1 Click , the following interface is shown. Choose batch configured features and click “ok”.

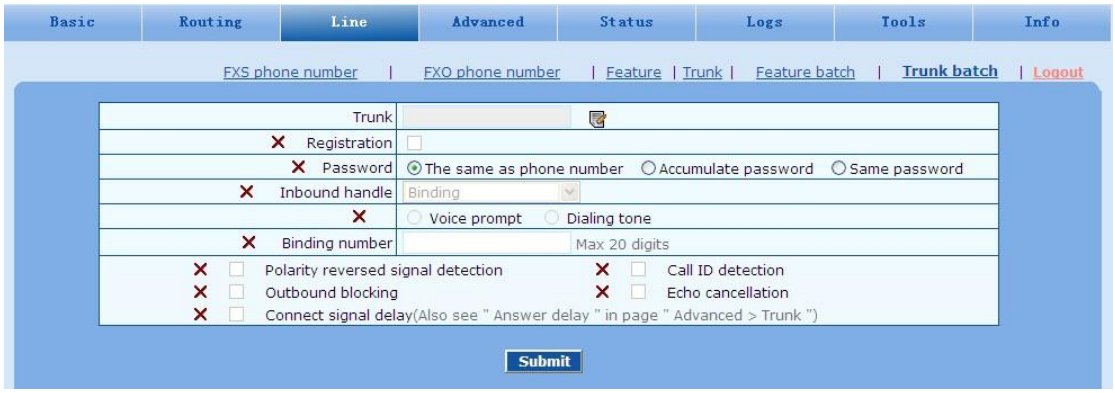


Step2 Click to activate this function to configure this parameter. For details of the parameter, see **Subscriber Line Features**.

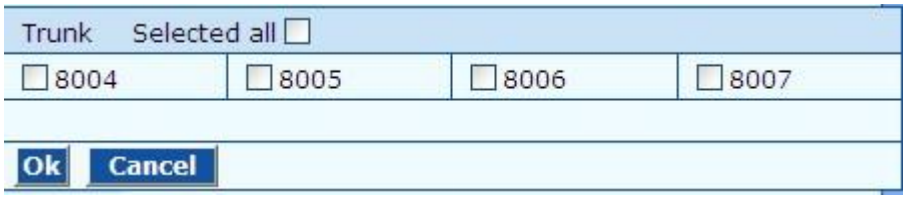
2.5.6 Trunk Batch

After login, click **Line > Trunk Batch** to open this interface.

Figure 2-17 Trunk Batch configuration interface



Step3 Click, the following interface is shown. Choose batch configured trunks and click **OK**.



Step4 Click ✕ to activate this function to configure this parameter. For details of the parameter, see **Trunk Line Features**.

2.6 Advanced Configuration

2.6.1 System

After login, click the label of **Advanced > System** to open this interface.

Figure 2-18 Interface of system advanced configuraiton

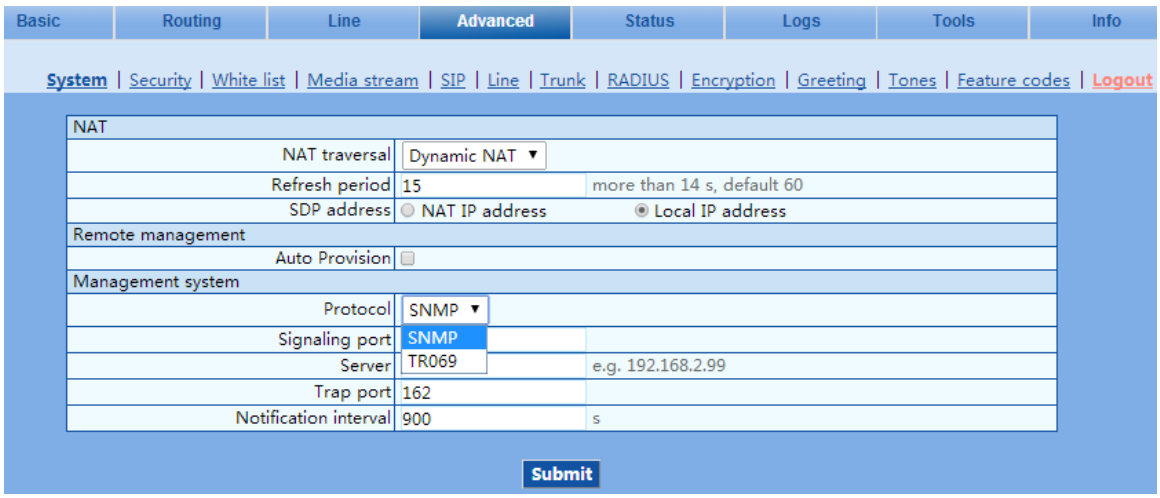


Table 2-18 Parameters of system advanced configuration

Title	Explanation
NAT traversal	Gateways support several mechanisms for NAT traversal. Usually, static NAT is used when a fixed public IP address is available. It's necessary to perform port mapping or DMZ function on router when choosing dynamic or static NAT.
Refresh period	The refresh time must be filled in here when choosing dynamic NAT or STUN traversal. Refresh time interval shall be determined by giving consideration to the NAT refresh time of the LAN router where the gateway is located. Gateway's NAT holding function and STUN function will carry out periodic operation according to this parameter. With seconds as its unit, default value of 60 seconds.
SDP Address	This parameter determines the IP address used in transmitted SDP. • **NAT IP Address:** apply NAT address into the transmitted SDP; • **Local IP Address:** apply the gateway's IP address into the transmitted SDP. Note: The parameter should come into effect only on condition that gateway successfully obtained NAT address.
Auto provision	Use auto provision function.

Title	Explanation
Protocol	SNMP: use the SNMP network management. TR069: use the TR069 network management.

Figure 2-19 Auto provisioning configuration interface

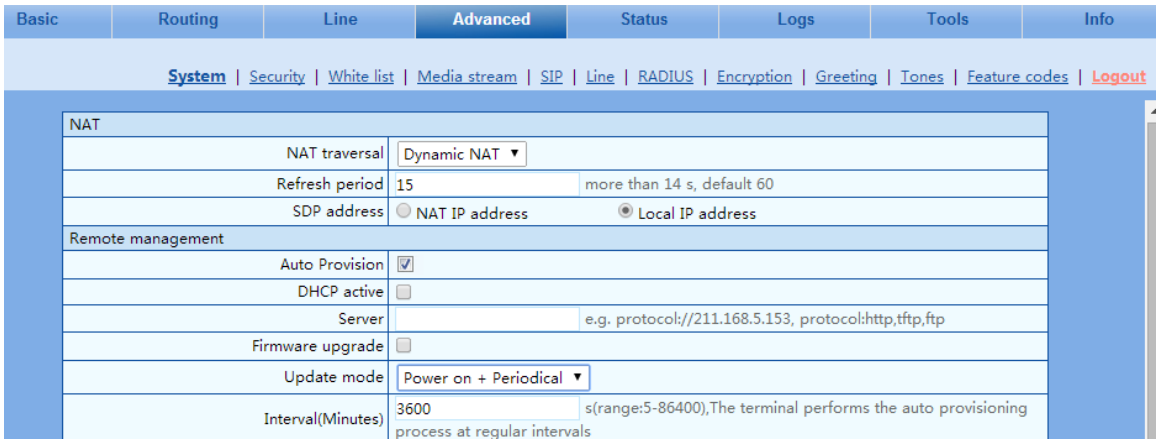


Table 2-19 Parameters for auto provisioning configuration

Parameter	Description
DHCP	ACS (Auto Provisioning Server) address is obtained by using OPTION66 of the DHCP.
Server	Manually configure the ACS address, which can be the TFTP, FTP, or HTTP server. • tftp://*ACS address* • ftp:// *ACS address* • http:// *ACS address*
Firmware upgrade	Supports firmware download and update using ACS.
Update mode	The following modes are available. • Power on: the gateway detects whether there are configurations and firmware to be updated when the device is powered on. • Power on + Periodical: when the device is powered on, the gateway first checks whether there are configurations and firmware to be updated, and then periodically performs checking based on the set times.
Interval(minutes)	When **Power on+Periodical** is set, this parameter specifies the interval for periodic automatic upgrades.

Figure 2-20 SNMP configuration interface

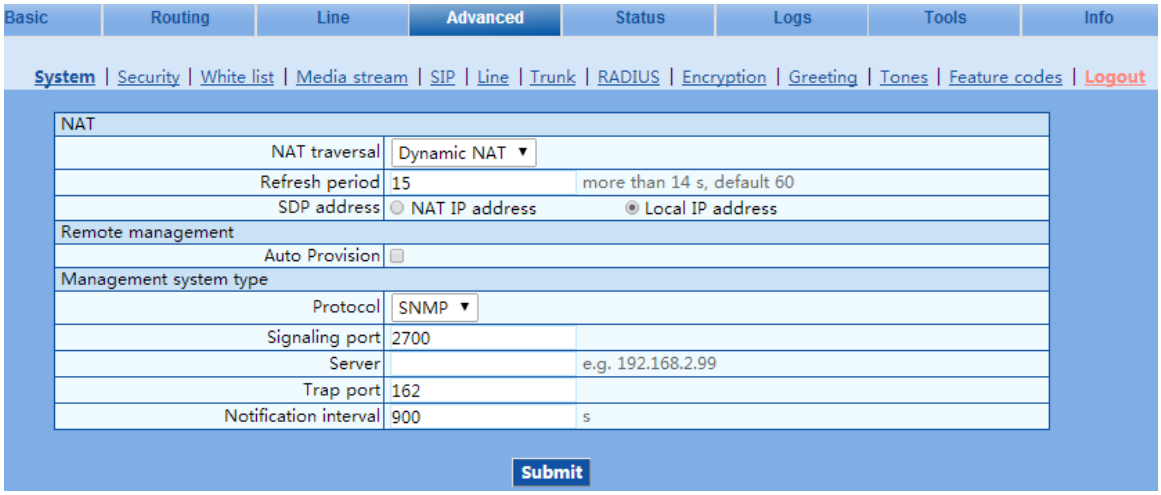


Table 2-20 Parameters for SNMP configuration

Parameter	Description
Signaling port	Enter the SNMP local port. The default value is 2700. If **SNMP** is selected, the following three parameters need to be specified.
Server	Enter the address of the SNMP server.
Trap port	Enter the port number of the SNMP server. The default value is 162.
Notification interval	The default value is 900 seconds.

Figure 2-21 TR069 configuration interface

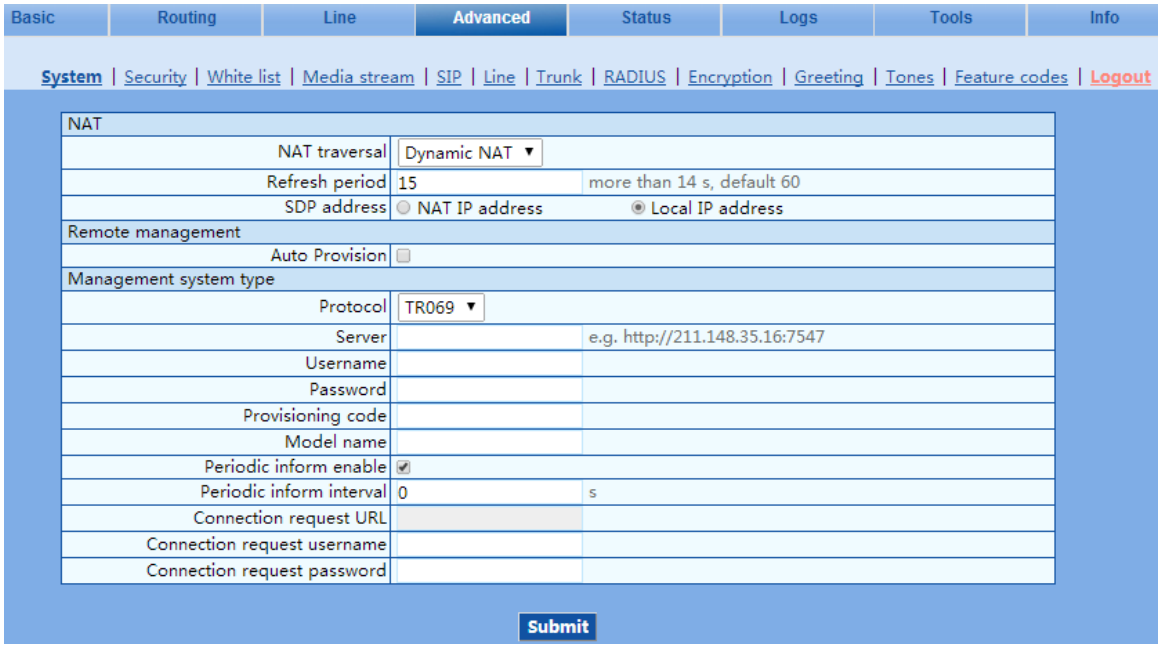


Table 2-21 Parameters for TR069 configuration

Parameter	Description
Server	Specify the URL of the ACS.

Parameter	Description
Username	Specify the user name to be used by the device to authenticate with the ACS.
Password	Specify the password to be used by the device to authenticate with the file server
Provisioning code	Information of the device vendor, which may be used to indicate the primary service provider and other provisioning information to the ACS. It can be numbers or English letters.
Model name	A brief description of the interface type or name. It is a string of characters.
Periodic inform enable	A switch used to specify whether to periodically report to the ACS.
Periodic inform interval	The interval for reporting to the ACS.
Connection request URL	The address used for the ACS to connect back to the device.
Connection request username	The account used for the ACS to connect back to the device, for example, admin.
Connection request password	The password used for the network management server to connect back to the device.

2.6.2 Security Configuration

After login, choose **Advanced** > **Security** to open the security configuration interface.

Figure 2-22 Security configuration interface

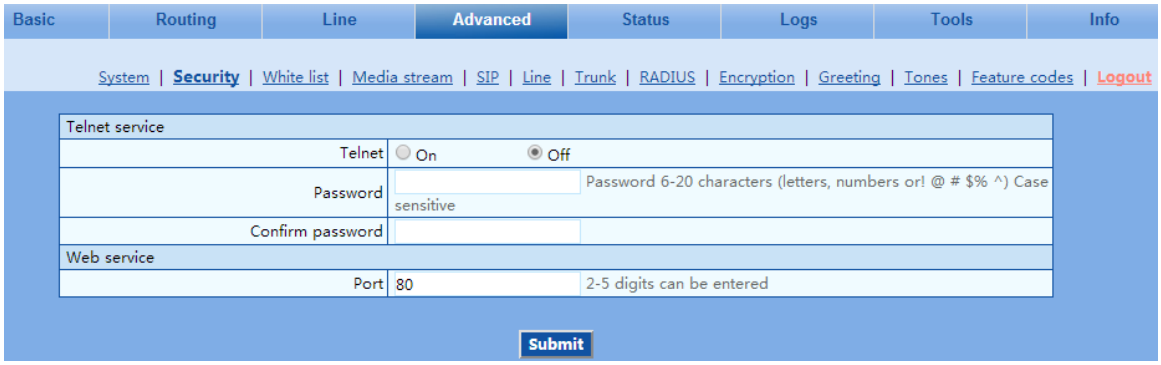


Table 2-22 Parameters for security configuration

Parameter	Description
Telnet	Enables or disables the Telnet function.
Password	The Telnet password consists of 6 to 20 characters (letters, digits, or !@#$%^) and is case-sensitive.
Confirm password	Repeat the Telnet password again.
Port	Enter 2 to 4 digits.

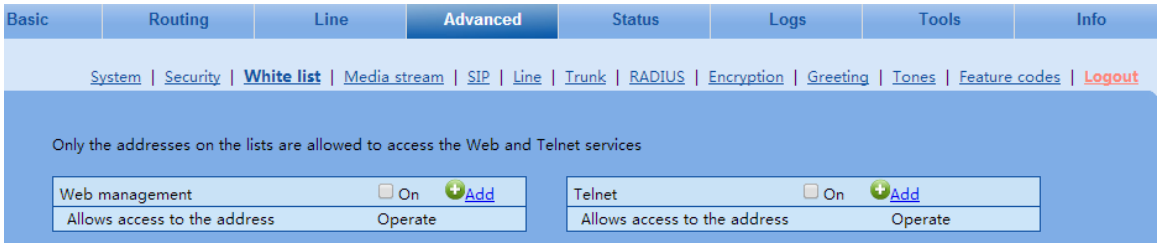
Note

If the gateway is placed in a public network environment, you should disable the Telnet function to prevent hacker attacks.

2.6.3 White List

After login, choose **Advanced** > **White list** to open the white list configuration interface.

Figure 2-23 White list configuration interface



To configure white list with Web GUI as an example, perform the following steps:

Step1 Click **Add**.

Step2 In the displayed box, enter the addresses that are allowed for access, and click **OK**.

Step3 Select enable.

Note

- This function takes effect after the device restarts.
- The device allows a white list of 20 entries.

2.6.4 Media Stream

After login, click the label of **Advanced** > **Media Stream** to open this interface.

Figure 2-24 Media stream configuration interface

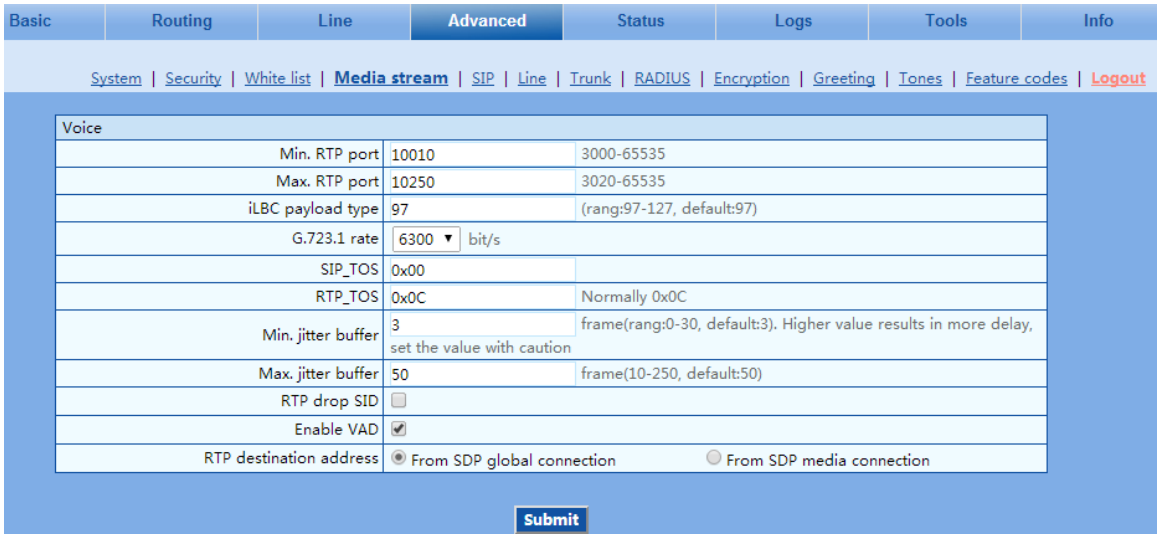


Table 2-23 Media stream configuration parameter

Title	Explanation
Min. RTP port	The lowest port number of UDP ports for RTP transmission and receiving. The parameter must be greater than or equal to 3000. This is a required field. Note: each phone call will occupy RTP and RTCP ports. If the gateway is equipped with 4 subscriber lines (or trunk line), then at least 8 UDP ports are needed.

Title	Explanation
Max. RTP port	The highest port number of UDP ports for RTP's transmission and receiving. This is a required field. The value must be greater than or equal to "2× number of lines＋min. RPT port".
iLBC payload type	Set the RTP payload type of iLBC，and the default value is 97. Accepted value is 97 ~ 127. The parameter shall be configured in conformity to that of platform.
G.723.1 rate	Set G.723.1 coding rate，the default value is 6300(bit/s). The optional parameters are followings: • 5300(bit/s): the Bit rate is 5.3k per second; • 6300(bit/s): the Bit rate is 6.3k per second
SIP_TOS	For SIP signaling, set the service level quality guarantee for different priorities. The default value is 0x00.
RTP_TOS	For RTP voice streams, set the service level quality guarantee for different priorities. The default value is 0x0c.
Min. Jitter buffer	RTP Jitter Buffer is constructed to reduce the influence brought by network jitter. This default value is 3.
Max. Jitter buffer	RTP Jitter Buffer helps to reduce the influence brought by network jitter. The default value is 50.
RTP drop SID	Determine whether to discard received RTP SID voice packets. By default, SID voice packets will not be dropped. Note: RTP SID packets should be dropped only when they are in nonconformity to the specifications. Nonstandard RTP SID data could generate noise for calls.
Enable VAD	Only applicable to G.723, GSM, iLBC. In case of selecting this parameter, it will not send any voice packet during mute period. By default, this is selected.
RTP destination address	This parameter determines where to obtain the IP address of the receiving side for RTP packets. By default, the IP address is obtained **From SDP global connection**. • **From SDP global connection:** obtain the IP address from SDP global connection; • **From SDP media connection:** obtain the IP address from SDP Media Description.

2.6.5 SIP Related Configuration

SIP messages consist of request message and response message. Both include a SIP message-header field and SIP message-body field. The SIP message header mainly describes the message sender and receiver; SIP message body mainly describes the specific implementation method of the dialog.

Message of request: the SIP message sent by a client to the server, for the purpose of activating the given operation, including INVITE, ACK, BYE, CANCEL, OPTION and UPDATE etc.

Message of response: the SIP message sent by a server to the client as response to the request, including 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses.

Message header: Call-ID.

Parameter line: Via, From, To, Contact, Csq, Content-length, Max-forward, Content-type, White Space, and SDP etc.

MX gateways provide good flexibility in content setting in order to improve compatibility with the SIP register server.

After login, click the label of **Advanced > SIP** to open this interface.

Figure 2-25 SIP related configuration interface

Table 2-24 SIP related configuration parameter

Title	Explanation
SIP related configuration	
MWI Re-subscription timer	The default is 86400 seconds. The gateway will send the platform a message to confirm that it has subscribed to MWI service at intervals of the time period set here. This parameter should be used in conjunction with voice mail subscription on the page of the subject subscriber line.
PRACK	Determine whether to activate Reliable Provisional Responses. (RFC 3262)
Session timer	Choose to activate session refresh (RFC 4028). By default, session timer is not activated.
Session interval	Set the session refresh interval, the gateway will enclose the value of Session-Expires into INVITE or UPDATE messages. Default value is 1800 seconds.
Minimum timer	Set the minimum value of session refresh interval.
Request/Response configure(SIP header)	
Contact field in REGISTER	Choose the registration mode of gateway under LAN traversal circumstance, the default is **NAT IP Address**. • **NAT IP address:** use the NAT information returned by registration server. • **LAN IP address:** keep original content of **Contact** when register;
Domain name in REGISTER	The default is **Domain name**. • **Domain name:** complete domain name used for registration (for example: 8801@registrar.newrock.com); • **Sub domain name:** only use the common part of the name of domain (for example: 8801@newrock.com).
Via field	Choose whether to use NAT IP address or LAN IP address for **Via** header field value, the default is **NAT IP address**.
To field	Choose whether to apply Sub domain name or Outbound proxy to **To** header field, the default is **Sub domain name**.

Title	Explanation
Address in Call ID field	Choose whether to fill Call ID field with Host name or Local IP address, the default is **Local IP address**.
Called party number	Choose whether the gateway acquires the called number from Request Line header field or To header field. The default is **From Request line field**.
Calling party number in call transfer	Under call forwarding, the calling party number sent can be chosen from Originating number or Forwarding number being set for sending, the default is **Forwarding number**. For example: the subscriber line 2551111 on the gateway activates call forwarding feature and set the destination to 3224422. When caller with 13055553333 calls 2551111, the call will be forwarded to 3224422: • if **Originating number** is chosen，the number 13055553333 will be sent to 3224422 as calling party number； • if **Forwarding number** is chosen， the number 2551111 will be sent to 3224422 as calling party number.
Do not validate Via	Set whether to ignore Via field, By default, Via is ignored.
Register upon invite timeout	Set whether to activate registration when SIP message of INVITE is failed or time expired, and by default, re-registration is not selected.
Selecting the receiving port for response	Use the receiving port of proxy or use the sending port of proxy.

2.6.6 Characteristics of Subscriber line

After login, click the label of **Advanced > Line** to open this interface.

Figure 2-26 Subscriber-line characteristics configuration interface

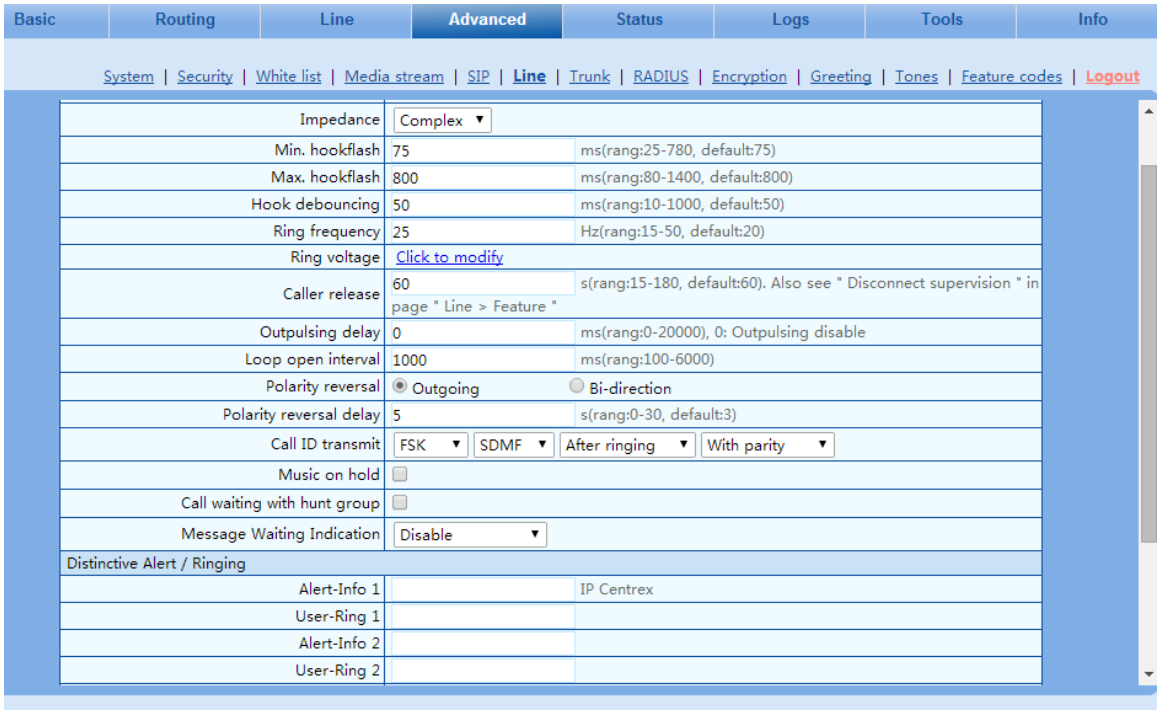


Table 2-25 Subscriber-line characteristics configuration parameter

Title	Explanation
Gain to IP	Set the voice volume gain toward the IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to terminal	Set the voice volume gain toward FXS port side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Impedance	Select the parameter of FXS port line impedance and the default value is 600 ohm. The optional values as below: • Complex • 600（ohm） • 900（ohm）
Min.hookflash	Used by the gateway to detect Hook Flash event, the default is 75 milliseconds. The gateway will ignore any flash that fall short of the shortest flash time. Generally, this value should not be less than 75 milliseconds.
Max.hookflash	Used by gateway to detect hook flash, the default is 800 milliseconds. The gateway will regard the flash duration between **Min.hookflash** and **Max.hookflash** as effective flash. Any flash lasting over the longest time will be considered by gateway as hang up. Generally, this value should not be less than 800 milliseconds.
Hook debouncing	Used by gateway to avoid a glitch of the phone status, with default of 50 milliseconds. When the duration from hang-up to off-hook falls short of this value, the gateway will ignore the status variation, and consider that the phone remains in hang-up status. In opposite case, the gateway will ignore the status variation, and consider the phone remains in off-hook status. Effective range of setting is 10~1000 milliseconds.
Ring frequency	Set the ringing frequency to be transmitted by gateway to the phone, ranging from 15 to 50 Hz, with default of 20 Hz.
Ring voltage	The ring voltage sent from the gateway to the phone is configurable. The range is from 45 to 85 Volt.
Caller release	Set the delay release time of line as caller control method, with default of 60 seconds. Effective range of setting is 15~180 seconds.
Outpulsing delay	Used when gateways' FXS port is connected with the trunk interface of PBXs. For calls from gateway to PBX, gateways will relay the extensions to PBX after the delay set here. Setting of **0** means no extension number relay. The default is **0** milliseconds.
Loop open interval	This parameter is used with the loop open disconnection request. The range is from 100 ms to 6000 ms.
Polarity reversal	Set the trigger for polarity reversal, the default is **Outgoing**. • **Outgoing:** transmit reverse polarity signal only when the outbound is connected; • **Bi-direction:** transmit reverse polarity signal for the connection of both inbound and out bound calls.
Polarity reversal delay	The delay time from a call being answered to the transmission of reverse polarity signal. The default value is 3 in seconds. Effective range of setting is 0 ~ 30 seconds.
Call ID transmit	Select transmission mode of Caller ID signal from the FXS port to the phone. • FSK or DTMF • SDMF or MDMF • Sending Caller ID data before or after ringing • Sending Caller ID data with or without parity
Music on hold	Choose whether to play the background music while call waiting, and the default is not to play.
Call waiting with hunt group	Choose whether to activate hunt group feature for call waiting, Default not selected.

Title	Explanation
Message waiting indication (MWI)	Choose the lighting method of message waiting indicator of voice mail here: None, Polarity reversed, FSK, high voltage lighting (not supported by MX8). Message waiting indicator refers to the special LED on a phone, working with voice mail function. When user receives a voice message. The gateway will light this lamp upon receiving the notice from platform; the light goes off when there's no unheard mail. It's essential to understand whether the phone supports the indicators and lighting method when selecting the lighting method.
Voltage	This parameter is used together with high voltage lighting for MWI. The range is from 60 V to 100 V. The default value is 80 V.
Distinctive Alert/Ringing	
Alert-Info 1	To match with **User-Ring 1**. Four patterns of user ring are offered. When the Alert-info value of INVITE message matches with this parameter, **User-Ring 1** is activated.
User-Ring 1	Configure user ring 1. E.g 1: if the user ring is set **2, 500, 500, 1000, 3000**, the ringing cadence is 0.5s on, 0.5s off; 1s on, 3s off. E.g 2: if the user ring is set **2000, 4000**, the ringing cadence will be 2s on, 4s off.
Alert-Info 2	To match with **User-Ring 2**
User-Ring 2	Configure user ring 2
Alert-Info 3	To match with **User-Ring3**
User-Ring 3	Configure user ring 3
Alert-Info 4	To match with **User-Ring 4**
User-Ring 4	Configure user ring 4

2.6.7 Characteristics of Trunk Line

After login, click the label of **Advanced > Trunk** to open this interface.

Figure 2-27 Trunk line characteristics configuraiton interface

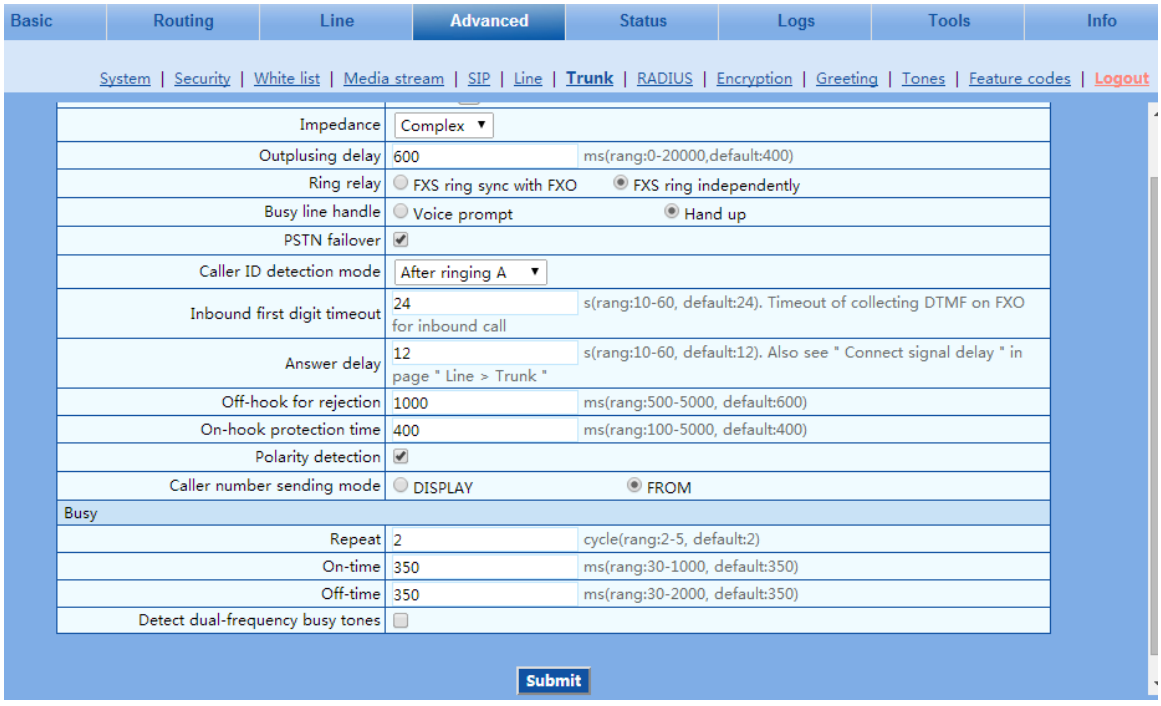


Table 2-26 Trunk line characteristics configuration parameter

Title	Explanation
Gain to IP	Set the voice volume gain toward IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +9 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to PSTN	Set the voice volume gain toward PSTN side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +9 decibels.
Impedance	Set the parameter of FXO impedance, with the default of 600 ohm. The optional settings are below: • Complex • 600（ohm） • 900（ohm）
Outplusing delay	Set the time interval between the FXO going off-hook and starting outpulsing of the first digit to the PSTN. The default is 600 in milliseconds.
Ring relay	Whether to relay the ring of inbound call to the FXS port when applying to DID. The default is **Phone ring independently**.
Busy line handle	Either a voice prompt or hanging up can be applied to FXO port when an incoming call goes to the FXS port which is in busy. This only applies to DID feature.
PSTN failover	Whether to route a call to the PSTN through an FXO port when the IP network faults or no response to the call request. Default selected.
Caller ID detection mode.	• Before ringing • After ringing
Inbound first digit timeout	Set the timeout of calling DTMF on FXO port for inbound calls, ranging from 10-60 seconds, with default of 24 seconds.
Answer delay	Set the delay time of outbound connection ranging from 10-60 seconds, with default of 12 seconds. Also see **Delay sending 200OK** on page **Phone/Line > Line**.
Off-hook for rejection	Used for binding an FXO port with an FXS port. For inbound calls to an FXO port, if the associated FXS port is busy, the gateway will hang up after off hook according to the time set by the parameter, so as to refuse the upcoming call. The duration of the off hook is 500~5000 milliseconds, with a default of 600 milliseconds.
On-hook protection time	Protection period following hang up of FXO port. During this period, gateway ignores any voltage variation of line. Value range is 100~5000 milliseconds, the default is 400 in milliseconds.
Polarity detection.	Choose whether to activate the detection of reverse polarity signal of FXO port. Note the detection will work only when the trunk supports polarity reversal.
Caller number sending mode	• DISPLAY: includes the incoming call number detected at the FXO port in the **Display** field and sends it to the peer end. The **From** field carries the phone number associated with the FXO port. • FROM: includes the incoming call number detected by FXO in the **From** field and sends it to the peer end. No Display information is carried.
Busy detection	
Repeat	Gateways will regard the busy tone signal with the repeat times specified here as a hang-up signal. Default is 2, effective range is 2 ~ 5(cycle).
On-time	Set duration of busy tone signal, the default is 350 in milliseconds.
Off-time	Set the interval time of busy tone, the default is 350 in milliseconds.
Detect dual-frequency busy tones	Default is -23(dB), effective range is -15 ~ -29 (dB).

2.6.8 Radius Call Logs

After login, click the label of **Advanced > RADIUS** to open this interface.

Figure 2-28 Configuration interface of Radius call logs

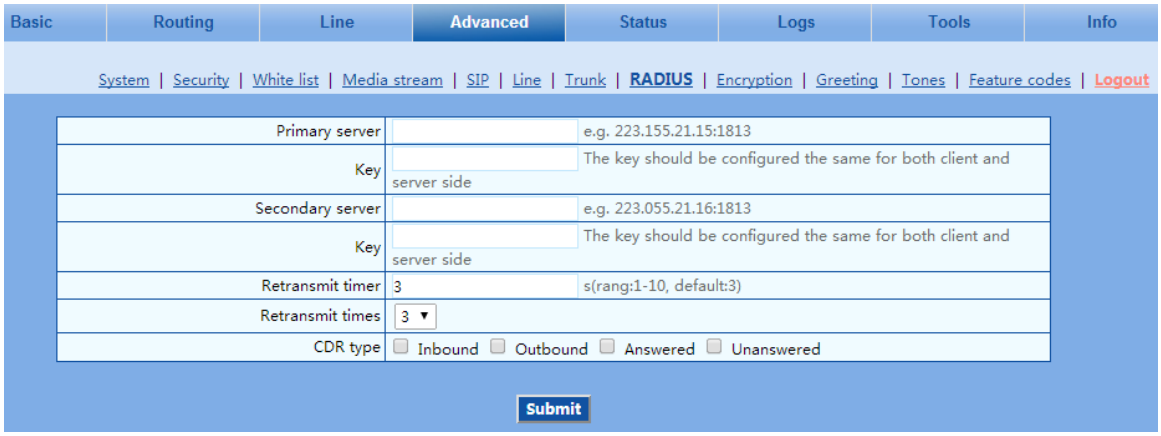


Table 2-27 Configuration parameter of Radius call logs

Title	Explanation
Primary server	Set IP address and port number of preferred Radius server. Note: if the port number is not configured yet, please use Radius default port number of 1813.
Key	Set the share key to be used for encrypted communications between Radius client and server. Note: The share key should be configured the same for both client and server side.
Secondary server	Set the IP address and port number of standby Radius server. When the fault appears in communications between gateway and preferred Radius server, the gateway will automatically activate standby Radius server. Note: In case of no configuration of port number, use default port number of 1813.
Key	The share key for communications between Radius client and standby Radius server. Note: The key should be configured the same for both client and server side
Retransmit timer	Set the amount of overtime on response after transmission of Radius message, the default is 3 seconds. The retransmission will be performed If no response is given after the timeout.
Retransmit times	Set the times of retransmission of Radius message when no response is received default is 3 times.
CDR type	• Inbound: set whether to send RADIUS charge message for inbound calls; • Outbound: set whether to send RADIUS charge message for outbound calls; • Answered: set whether to send RADIUS charge message when calls are connected; • Unanswered: set whether to send RADIUS charge message for unanswered calls.

2.6.9 Encryption

After login, click the label of **Advanced > Encryption** to open this interface.

Figure 2-29 Encryption configuration interface

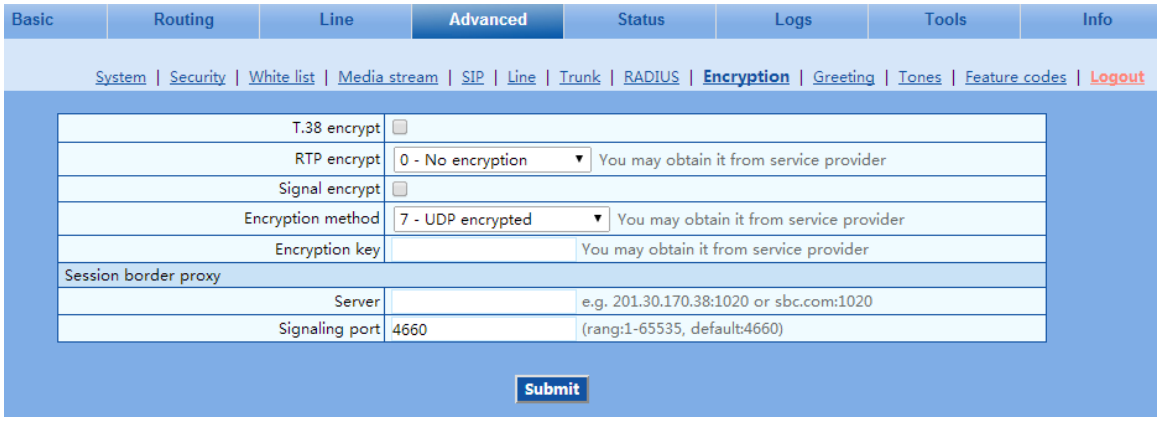


Table 2-28 Encryption configuration parameters

Title	Explanation
T.38 encrypt	Choose whether to encrypt T38 data. By default, this is not selected.
RTP encrypt	Choose whether to encrypt RTP voice pack, the default is **0**. • 0: no encryption • 1: entire message • 2: header only • 3: the data body only
Signal encrypt	Choose whether to encrypt signaling. By default, this is not selected.
Encryption method	Set the gateway encryption method, default is 7. The optional parameters as below: • 2:TCP not encrypted • 3: TCP encrypted • 6: UDP not encrypted • 7: UDP not encrypted • 8: Using keyword • 10: RC4 • 13: Encrypt13 • 14: Encrypt14 • 16: Word reverse(263) • 17: Word exchange(263) • 18: Byte reverse(263) • 19: Byte exchange(263) • 20:VOS
Encryption key	You may obtain it from service provider
Session Border Proxy	
Server	Set the IP address and port number of session border proxy server. The character of ":" must be used between IP address and port number. Server address could be set into IP address or domain name. When domain name is used, **DNS service** must be activated as shown in the page of **Network**, and **DNS server** must be configured. Example: 201.30.170.38:1020 or sbc.com:1020.
Signaling port	Signaling port assignment of the gateway, the default value is 4660. Signaling port number may be set at will, but cannot conflict with other ports of equipment.

2.6.10 Greeting

After login, choose **Advanced** > **Tones** to open the audio files interface.

Figure 2-30 Greeting interface

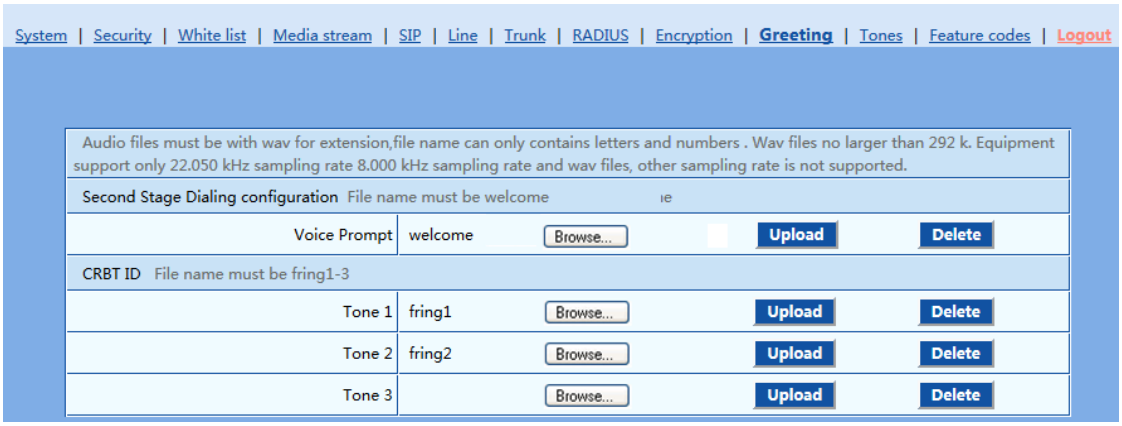


Table 2-29 Parameters for greeting

Parameter	Description
Second Stage Dialing Configuration	Click **Browse**, and then select the local audio file named **welcome.wav**. Click **Upload**. The uploaded audio file overwrites the original one. If you want to delete the current customized second stage dialing tone, click **Delete**. After the gateway restarts, the default second stage dialing tone is used.
CRBT ID	Click **Browse**, and then select the local audio file named **fring1/2/3.wav**. Click **Upload**. The uploaded audio file overwrites the original one. If you want to delete the current color ringback tone, you can click **Delete**. After the gateway restarts, the default color ringback tone is used.

2.6.11 Call Progress Tone Plan

After login, click the label of **Advanced > Tones** to open this interface.

Figure 2-31 Call progress tone configuration interface

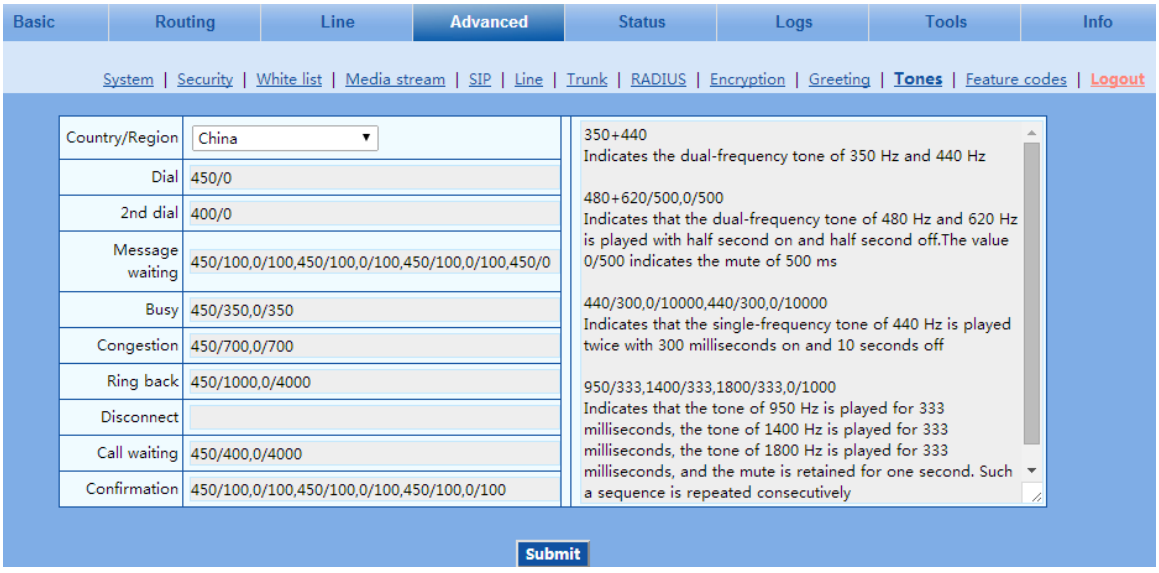


Table 2-30 Call progress tone configuration parameters

Title	Explanation
Country/Region	There are progress tone plans for several countries and regions which are pre-programmed in gateways. Users may also specify the tone plan according to the national standard. Gateways provide tone plans for the following countries and regions: China; the United States; France; Italy; Germany; Mexico; Chile; Russia; Japan; South Korea; Hong Kong; Taiwan; India; Sudan; Iran; Algeria; Pakistan; Philippines; Kazakhstan, Singapore, Israel, Malaysia, Indonesia, United Arab Emirates, Zimbabwe, Australia.
Dial	Prompt tone of off-hook dial tone.
2nd dial	Used for the second stage dial tone.
Message waiting	Used for prompt of voice mail, or when the subscriber line is set with “Don’t Disturb Service and Call Transfer”.
Busy	Used for busy line prompt. Used for busy line prompts.
Congestion	Used for notification of call set up failure due to resource limit.
Ring back	The tone sent to caller when ringing is on.
Disconnect	Used for reminding the subscriber of off-hook and no dialup status of the phone.
Call waiting	Used for notification in call waiting.
Confirmation	Used for confirming function keys being entered.

Here are examples that illustrate the various call-progress tones

350+440 (dial tone)

Indicates the dual–frequency tone consisting of 350 and 440 Hz

480+620/500,0/500 (busy)

Indicates the dual–frequency tone consisting of 480 and 620 Hz, repeated playing with 500 milliseconds on and 500 milliseconds off.

Note: 0/500 indicates 500 milliseconds mute.

440/300,0/10000,440/300,0/10000

Indicates 440 Hz single frequency tone, repeated twice in terms of 300 milliseconds on and 10 seconds off.

950/333,1400/333,1800/333,0/1000

Indicates repeated playing 333 milliseconds of 950 Hz, 333 milliseconds of 1400 Hz, 333 milliseconds of 1800 Hz, and mute of 1 second.

2.6.12 Feature Codes

The feature codes consist of system feature codes and service feature codes. The system feature codes are used for acquiring gateway information, and the latter is used for users to activate and inactivate supplementary services.

After login, click the label of **Advanced > Feature codes** to open this interface.

The following are the examples of the dialing rule for the feature codes:

Using *xx (dial * and 2 digits number) to activate a service

Using #xx (dial # and 2 digits number) to cancel a service

This is illustrated with the following defaults for various parameters, which may be modified according to requirements.

Figure 2-32 Feature codes configuration interface

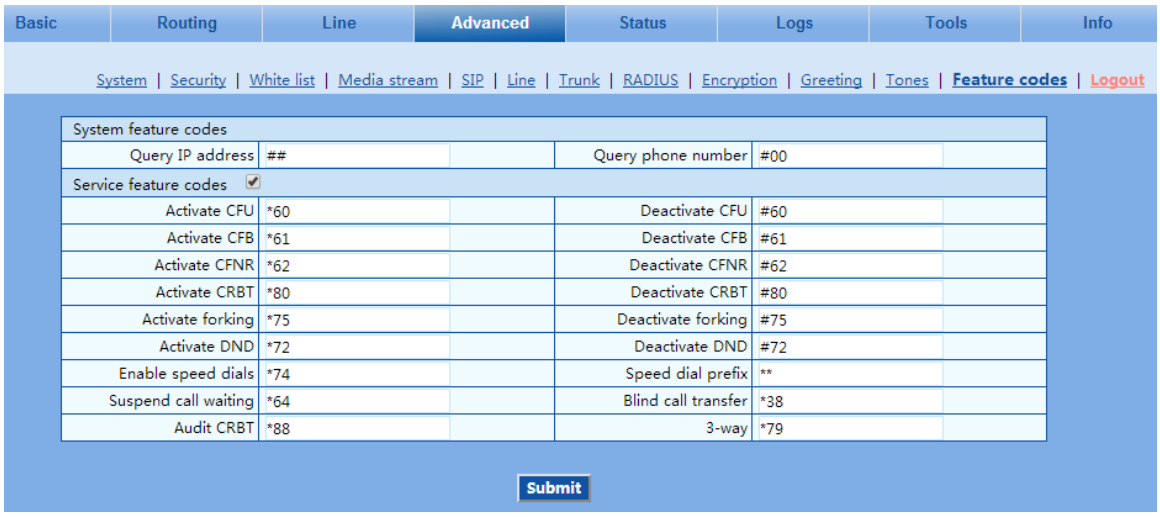


Table 2-31 Feature codes configuration parameter

Title	Explanation
System feature codes	
Query IP address	The function key for determining the IP address of gateway, with a default of ##. Dialing this key, users can hear the gateway voice the IP address and system-software version number. Narrative: if the gateway is only equipped with FXO port, connect FXO port through the PBX extension line or PSTN direct line, and dial the number of this line accordingly, press ## immediately after hearing the second dial tone, users may thus hear the IP address and system software version number of the gateway.
Query phone number	The function key for determining the phone number of this subscriber line, with default of #00. By dialing this key, your will hear the phone number of the subscriber line voiced by the gateway.
Service feature codes	
Activate CFU	The function key for activating unconditional call forwarding, with a default of *60. Dialing this key will activate unconditional call forward of the line and set the destination number for call forwarding. User operation: off hook → press *60 →enter the destination number. Users can determine the latest destination number set by dialing *60*. Note: It's required to enable call forwarding service before using this function (please see the instructions on the relevant configuration of **subscriber line**).
Deactivate CFU	The function key for deactivating unconditional call forwarding, with default of #60. User operation: off hook → press #60 → hang up.
Activate CFB	The function key for activating call forwarding on busy， with default of *61. Dialing this key may activate CFB, and specify the destination number. Note: It's required to enable call forwarding on busy service before using this function (please see the instructions on relevant configuration of **subscriber line**).
Deactivate CFB	The function key for deactivating call forwarding on busy, with default of #61. User operation: off hook → press #61 → hang up.

Title	Explanation
Activate CFNR	The function key for activating call forwarding on no answer, with default of *62. Dialing the function key may activate call forwarding on no answer and specify destination number. Note: It's required to enable call forwarding on no answer service before using this function (please see the instructions on relevant configuration of **subscriber line**).
Deactivate CFNR	The function key for deactivating call forwarding on no answer, with default of #62.
Activate CRBT	The function key for activating color ringback tone, with default of *80. Subscribers may select their favorite color RB tone by using this key. Note: It's required to start color ring service before using this function (please see **Phone** for how to assign the feature to the phone). User operation: upon off hook, the subscriber may press the function key (e.g. *80), then, input the two-digit index numbers of color ring; *80* is used for hearing and inquiring the color ring that has been previously set.
Deactivate CRBT	The function key for deactivating the color ring, with default of #80. The subscriber may use such key to recover the normal ring of phone. User operation: off hook → press #80 → hang up.
Activate forking	The function key for activating the double-ring/forking feature, with default of *75.
Deactivate forking	The function key for deactivating the feature, with default of #75.
Activate DND	Activate "Don't Disturb", with default of *72. With DND selected, the gateway will reject all coming calls by sending busy tone to the caller. Note: It's required to start "Don't Disturb" prior to using this function (please see the instructions on relevant configuration of **subscriber line**).
Deactivate DND	The function key to cancel "Don't Disturb", with default of #72. Dialing the function key may recover normal ringing upon the arrival of incoming calls.
Enable speed dials	Define the function key of dial, with default of *74. This key allows the user to build a table of 2-digits（20~49）speed-dial numbers. Note: It's necessary to get the dial-up service under way before applying this function (please see **Phone** for how to assign the feature to the phone). User operation: upon dialing the function key (*74), dial the two-digit speed dial followed by the expanded number terminated with #.
Speed dial prefix	The prefix number for applying abbreviated dialing, with default of **. The said prefix should be added ahead of abbreviated dialing numbers when using abbreviated dialing. User operation: off hook → dial the prefix number of abbreviated dialing (**) and dial abbreviated dialing number (20).
Suspend call waiting	The function key for cancelling the call waiting feature for next call, with default of *64. Dialing this function key will temporarily shield the user from a call-waiting distraction for next call, avoiding the possible intervention. Note: The function key works only for single cancel, if to cancel the call waiting completely, please refer to the instructions on relevant configuration of **subscriber line**.
Blind call transfer	Function key of blind call transfer，with default of *38. User operation: during the call, tap the phone hook switch or press R button→ dial *38→ dial the called number and then hang up.
Audit CRBT	The function key for hearing the color ring，with default of *88. User operation: off hook → press *88 → input color ring number.
3-way	*1

2.7 Status

2.7.1 Call Status

After login, click **Status > Call Status** to open this interface.

Figure 2-33 Interface of call status

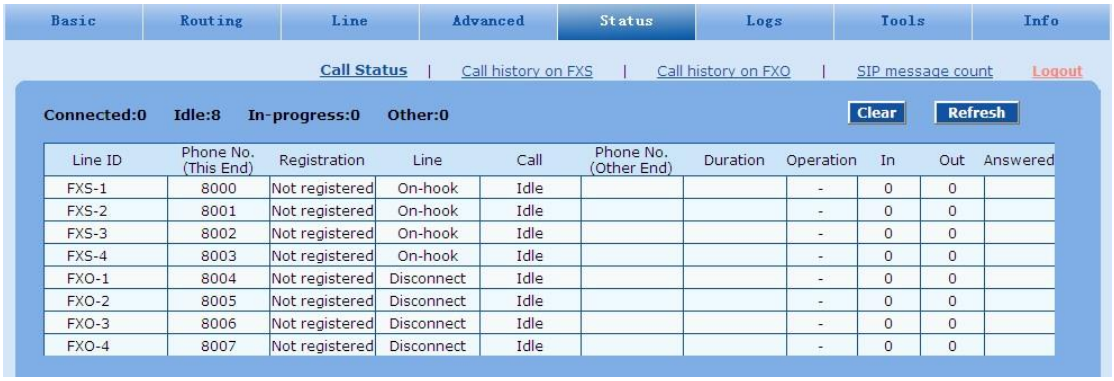


Table 2-32 Parameters of call state

Title	Explanation
Line	There are six types of line status, On-hook, Off-hook, Ringing, Maintenance, Disconnect, Parallel line in-use.
Call	The call state includes Idle, Outpulsing, Ring, Entering number, In progress, Ring back, Talk, Near end hung up, Far end hung up, and Timeout.

2.7.2 Call History on FXS

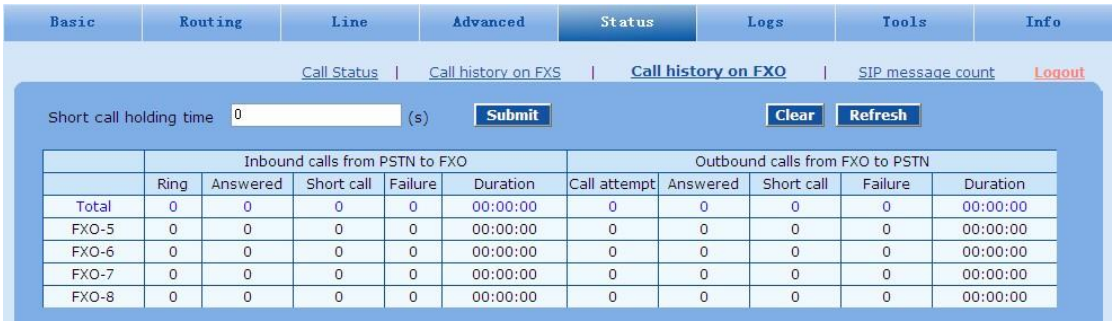
After login, click **Status > Call history on FXS** to open this interface.

Figure 2-34 Interface of call on FXS

2.7.3 Call History on FXO

After login, click the label of **Status > Call history on FXO** to open this interface.

Figure 2-35 Interface of call on FX0



	Inbound calls from PSTN to FXO					Outbound calls from FXO to PSTN				
	Ring	Answered	Short call	Failure	Duration	Call attempt	Answered	Short call	Failure	Duration
Total	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-5	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-6	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-7	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-8	0	0	0	0	00:00:00	0	0	0	0	00:00:00

2.7.4 SIP Message count

After login, click **Status > SIP message count** to open this interface.

Figure 2-36 Interface of SIP message count

Request							
	REGISTER	INVITE	ACK	BYE	CANCEL	INFO	Other
Send	0	0	0	0	0	0	0
Resend	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	19
Multiple receive	0	0	0	0	0	0	0

Response							
	200 OK	100 Trying	180 Ringing	183 Session progress	302 Moved temporarily	486 Busy here	487 Request terminated
Send	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	0

Other							
	1xx Provisional	2xx Success	3xx Redirection	4xx Client error	5xx Server error	6xx Global failure	
Send	1	19	0	0	0	0	-
Receive	0	0	0	0	0	0	-

2.8 Logs

2.8.1 System Status

Critical runtime information of gateways can be obtained in this interface, including:

The information about login interface (including IP address and permissions of the user)

SIP registration status

Call-related signaling and media (RTP) information

After login, click the label of **Logs > System Status** to open this interface.

Figure 2-37 Interface of System Status

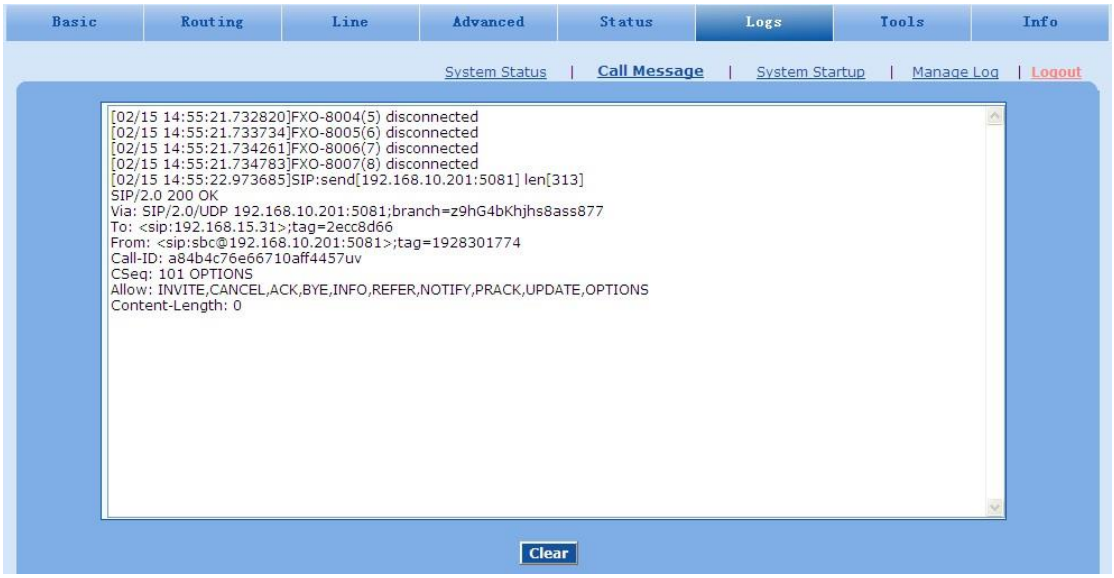
Table 2-33 Parameters of System Status

Title	Explanation
Login User Info	Show the IP address and permissions of the login user. The numbers following the IP address show the online permission level of the user: 1- administrator; 2 - operator; 3 – viewer. The viewer can only read the configuration. When more than one administrator logs in at the same time, the first login's permission level is 1; others are 3; also, when more than one operator logs in at the same time, the first one's permission is 2, others are 3.
SIP Registration Info	Show registration status: • Not enabled: the registration server's address is not entered yet; • Latest response: the latest response message for the registration. 200 means registered successfully; • No response: no response from registration server. The cause may contribute to 1） incorrect address for the registration server; 2) IP network fault; or, 3) the registration server is not reachable.
Latest Call Info	Show the latest call.
Call Context Info	Show the call status.
Rtp Context Info	Show the voice channel related to the calls.

2.8.2 Call Message

After login, click **Logs > Call Message** to open this interface.

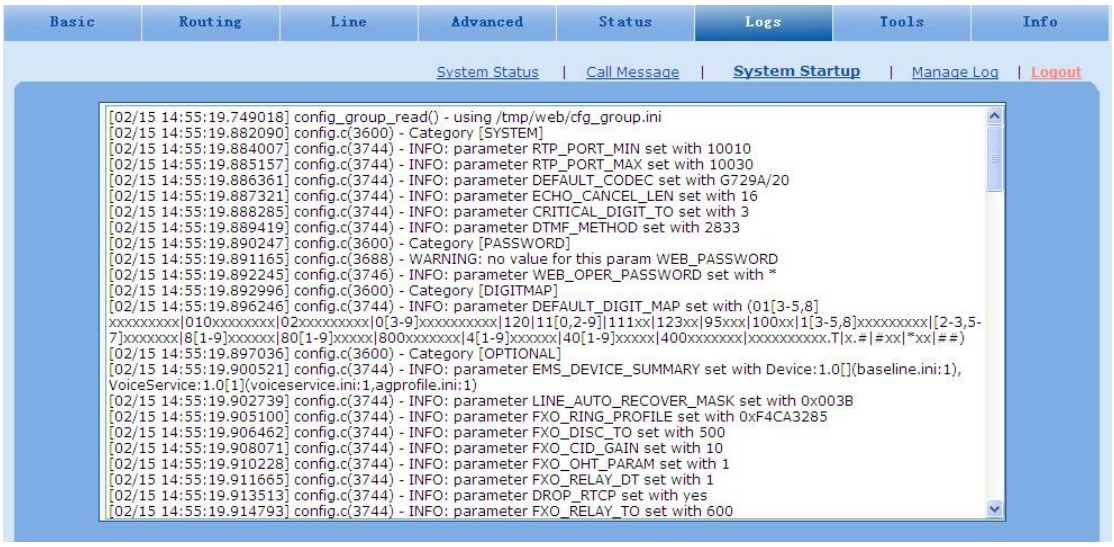
Figure 2-38 Call Message interface



2.8.3 System Startup

After login, click **Logs > System Startup** to open this interface. Log files can be downloaded through this interface.

Figure 2-39 Interface of System Startup



2.8.4 Manage Log

After login, click **Logs > Manage Log** to open this interface. Log files can be downloaded through this interface.

Figure 2-40 Interface of Manage Log

Table 2-34 Configuration parameters of Manage Log

Title	Explanation
Log level	Select the log file level of gateway, default is 4. The higher the level the more details the log file will be. Note: Log level should be set to 4 or lower when gateway is used in normal operation, avoiding reducing the system performance.
System log server	Set the IP address of the system log server.
Local log port	The port used to send logs.
Log server	IP address of debugging log server.

Procedure for downloading the log:

Step1 Click **Download**, the gateway begins to assemble the logs.

Step2 After a few seconds, the interface of log saving will appear.

Step3 Click **Save**, and select path to save.

Step4 The user may review the log from the server.

Note

The procedure of downloading log files described hereof is only applicable to release 1.9.x.238 of MX series or updated version of software.

2.9 Tools

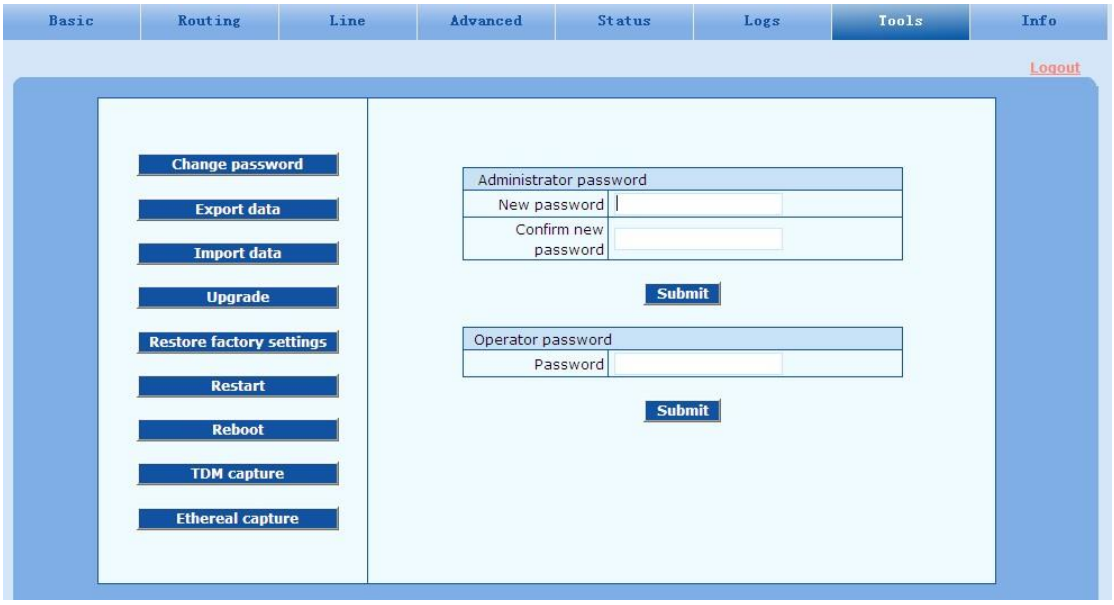
2.9.1 Change Password

After login, click **Tools** to open this interface. Only administrator is entitled to change the password of login.

For changing administrator password, it's required to enter new password into **New password** field and **Confirm new password** field, and then click **Submit**.

The password being used by the operator will be displayed as hidden codes, which could be changed by the administrator at any time. The administrator is allowed to change the operator's password by entering the new password into **Operator password > password**.

Figure 2-41 Interface for password changing



2.9.2 Export Data

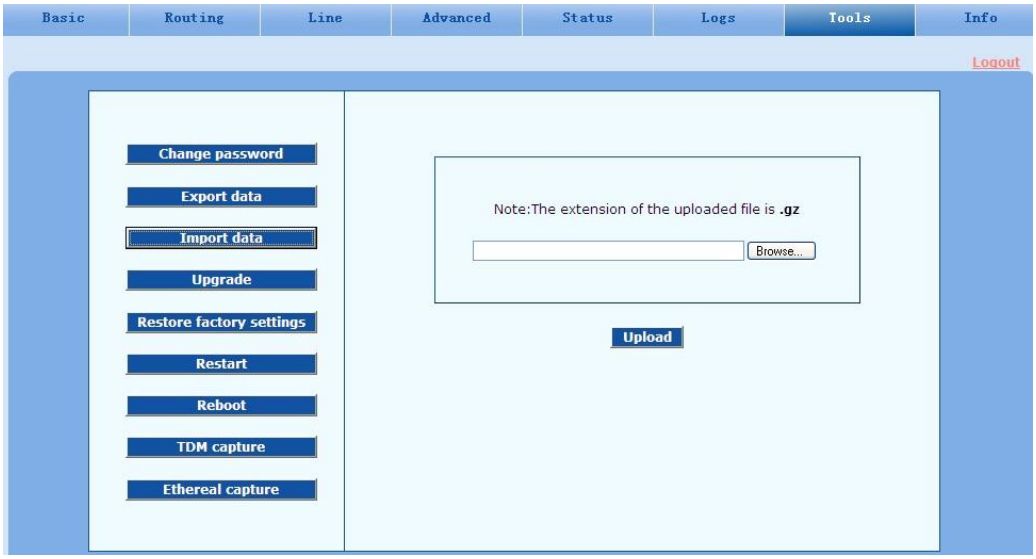
After login, click **Tools > Export data** to open this interface. The download procedure is similar to the download procedure of log files.

Figure 2-42 Interface of export data

2.9.3 Import Data

After login, click **Tools>Import data** to open this interface. Operating procedure is the same as that of software upgrade.

Figure 2-43 Interface of import data



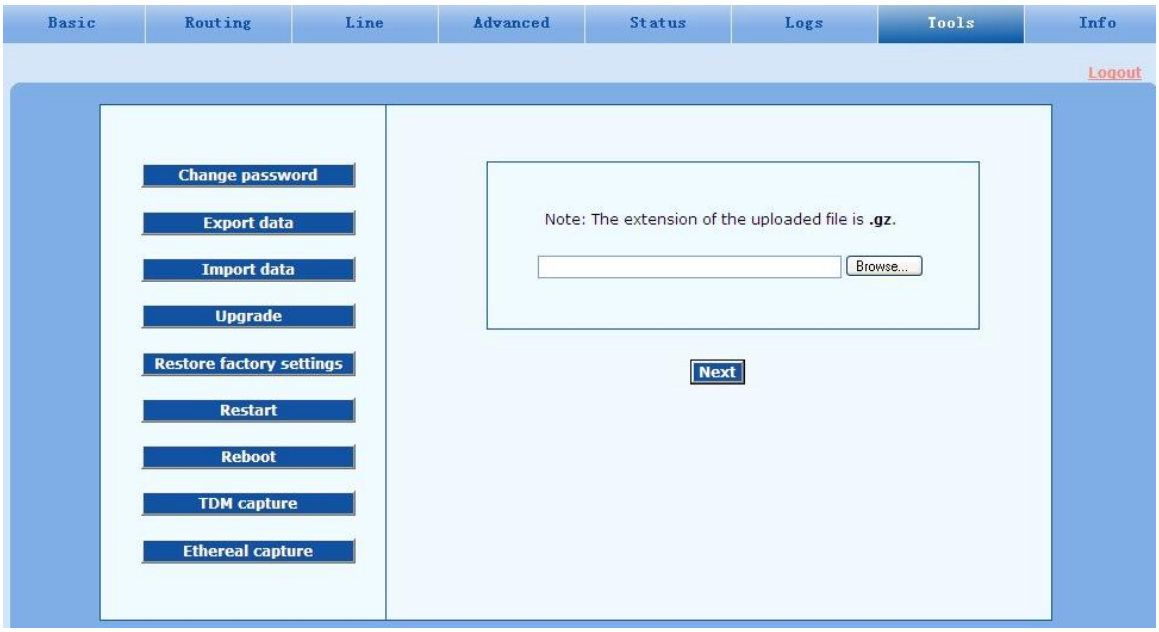
2.9.4 Upgrade

After login, click **Tools > Upgrade** to open this interface. The software upgrade procedure is presented as below:

Step1 Obtain the upgrade files (tar.gz file), and save the file onto a local computer.

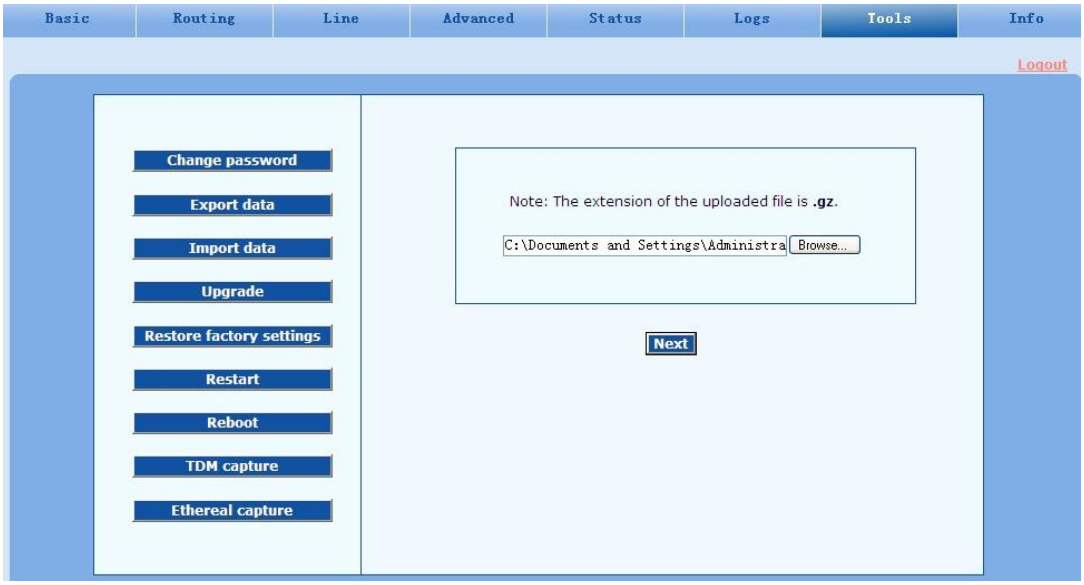
Step2 Click **Tools > Upgrade** to access to the page of software upgrade.

Figure 2-44 Interface of upgrade



Step3 Click **Browse** to select the upgrade files.

Figure 2-45 Interface of file upload



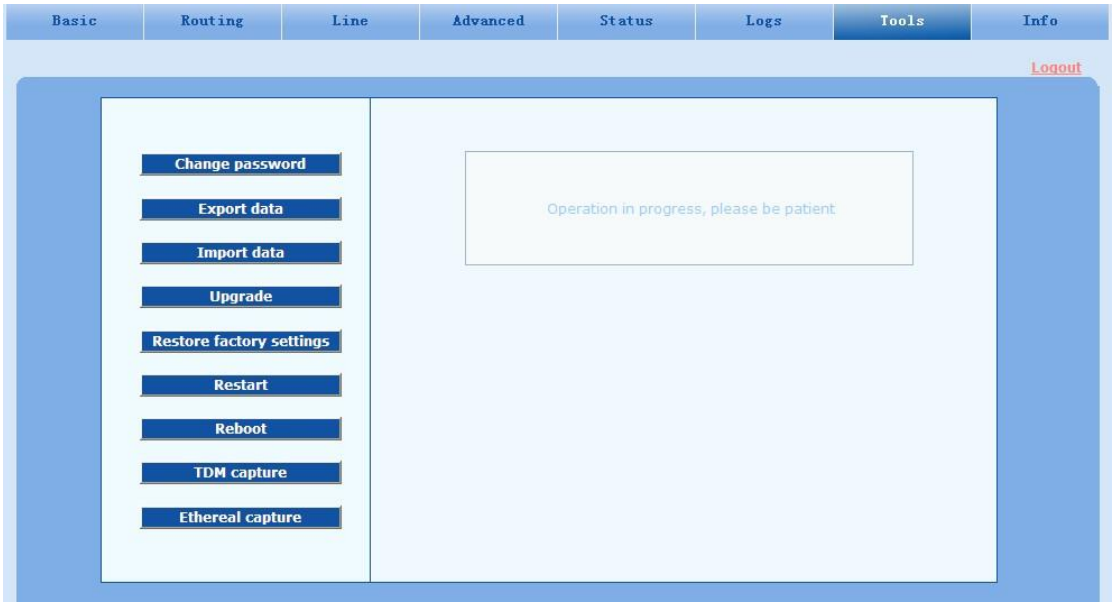
Step4 Click **Upload**.

Step5 Uploading will be completed in about 30 seconds, then click **Next**.

Figure 2-46 Upgrade interface

Step6 The following prompt appears during the upgrade.

Figure 2-47 Screen of upgrade process



Note

A few minutes are needed to upgrade the gateway. Don't operate the gateway during this period.

Step7 After success in upgrade, the following dialog will appear, click **Confirm**.

Figure 2-48 Interface of successful upgrade

Step8 The gateway is on the progress of reboot when the interface cannot be displayed.

Step9 Wait for about two minutes, and access the interface of gateway management system, click **Version info** and check the software version.

Note

For MX100 and MX120 gateways, the software upgrade operation must be conducted on the 100M Ethernet port.

2.9.5 Restore Factory Settings

After login, click **Tools > Restore factory settings** to restore the factory settings.

The factory settings are designed based on common applications, and therefore, no need to modify them in many deployment situations.

2.9.6 Software Restart

After login, click **Tools > Restart** to restart the gateway, making modified configuration come into effect.

Note

In most cases, there is no need to reset the gateway, and the modified parameters will come into effect upon confirming the **Submit**.

2.9.7 System Reboot

After login, click **Tools > Reboot** to restart the gateway. As this is a system wide reset, it takes longer time.

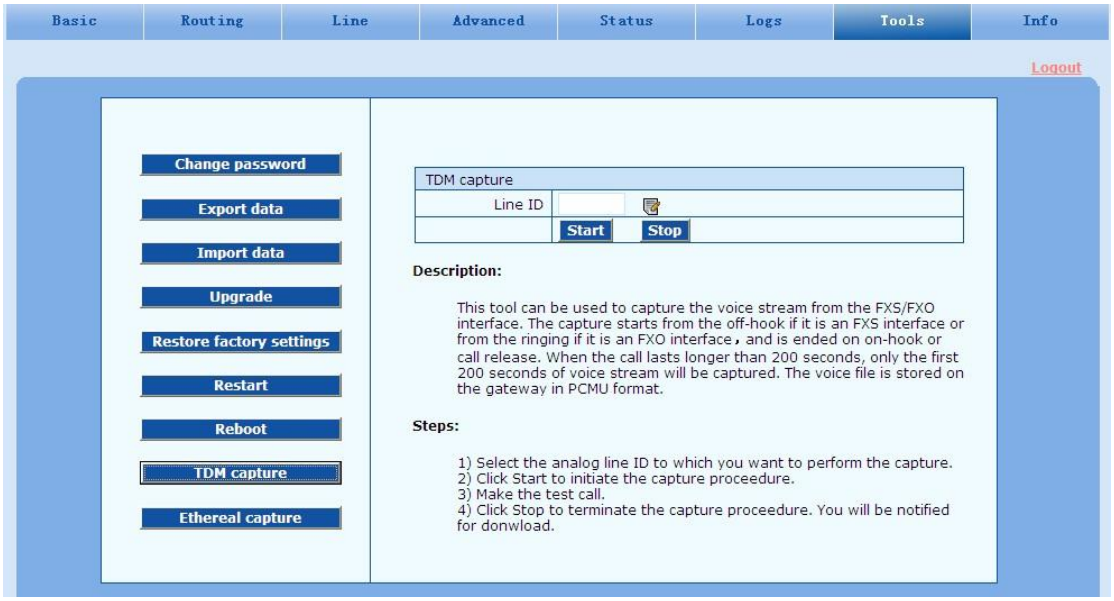
Note

Generally, it's sufficient to restart software when the gateway confirms to reset; the system reboot will be required only when network settings of the gateway are changed.

2.9.8 TDM Capture

After login, click **Tools > TDM capture** to open this interface. This tool can be used to capture the voice stream from the Phone or Line interface. The capture starts from the off-hook if it is a Phone interface or from the ringing if it is a Line interface， and is ended on on-hook or call release. When the call lasts longer than 200 seconds, only the first 200 seconds of voice stream will be captured. The voice file is stored on the gateway in PCMU format.

Figure 2-49 Interface of TDM capture



Step1 Select the analog line ID to which you want to perform the capture.

Step2 Click Start to initiate the capture procedure.

Step3 Make the test call.

Step4 Click Stop to terminate the capture procedure. You will be notified for download.

2.9.9 Ethereal Capture

After login, click **Tools** > **Ethereal capture** to open this interface. You are allowed to capture up to three IP voice data files, each with up to 2M bytes. The data files are stored on the gateway in dump.cap format under catalog **/var/log**.

Figure 2-50 Interface of Ethereal capture

Step1 Click **Start** to initiate the capture procedure.

Step2 Click **Stop** to terminate the capture procedure. You will be notified for download.

2.10 Version Information

After login, click **Version info** to view the gateway hardware and software version information.

Figure 2-51 Interface of Version info

2.11 Logout

After login, click the **Logout** at top right to exit the gateway management system and return to the login interface.

3 Appendix: High Availability Configuration

3.1 Overview

3.1.1 Function Definition

In the deployment of VoIP network, New Rock MX-Series VoIP Gateway (referred as *gateway* below) supports **high availability** architecture with **Primary-Standby**, **Active-Standby** mode and **Load balancing** mode.

Primary-Standby mode

In this mode, a backup SIP proxy server (referred as SIP server) is configured. The gateway will failover to the backup server automatically when the primary server faults. The gateway detects the failure condition of primary server by sending OPTIONS request to it.

- If the gateway does not receive the response to OPTIONS request, it will failover to the backup server.
- After failover to the backup server, the gateway will still send OPTIONS to the primary server all the same. It switches back to the primary server once the response to the OPTIONS request is received.

Active-Standby mode

In this mode, one SIP proxy server (referred as SIP server) functions as the primary server while other SIP servers function as standby servers.

Either of the following conditions could trigger the failover operation of the gateway:

- Not receiving response to the OPTIONS message from the current SIP server to which the gateway sends or receives call traffic; or
- Not receiving response to the REGISTER/INVITE message from the current SIP server to which the gateway send or receives call traffic

The administrator can manually switchover the gateway from the current SIP server to the next available one.

The gateway will redirect call traffic to the designated proxy server in responding to the re-INVITE from the server.

Load balancing mode

In this mode, the clustered SIP servers are all working in active status. Under the coarse grained scheme all endpoints of a gateway are allowed to register on one of the designated servers and under the fine

grained scheme the endpoints of a gateway are allowed to register on multiple servers, according to the administrator's load balancing plan. The following features are supported with load balancing:

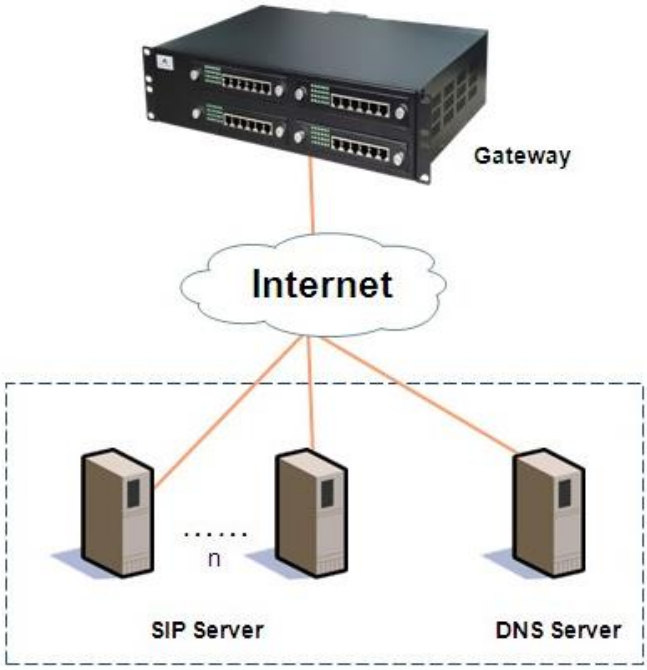
- The gateway as a whole or endpoints search for the designated sever in the server cluster (a list of servers) using REGISTER/INVITE message in forward circular scheme.
- Server failure detetion is supported by gateway sending OPTIONS to each servers, on which the gateway or endpoints are registered on.
- Upon the condition of no response to OPTIONS or REGISTER/INVITE, the gateway will search for the next available servers for the gateway or endpoints and move the calls to them accordingly

The gateway will redirect call traffic to the designated proxy server in responding to the re-INVITE from the server.

3.1.2 Server Cluster

The server cluster includes one primary SIP proxy server and one backup SIP proxy server under primary-standby mode, one primary SIP proxy server and up to *five* standby proxy servers under active-standby mode or six active servers under load balancing mode. The address of the SIP server can be configured manually by the administrator or obtained through DNS SRV record. Topology is shown as bellow:

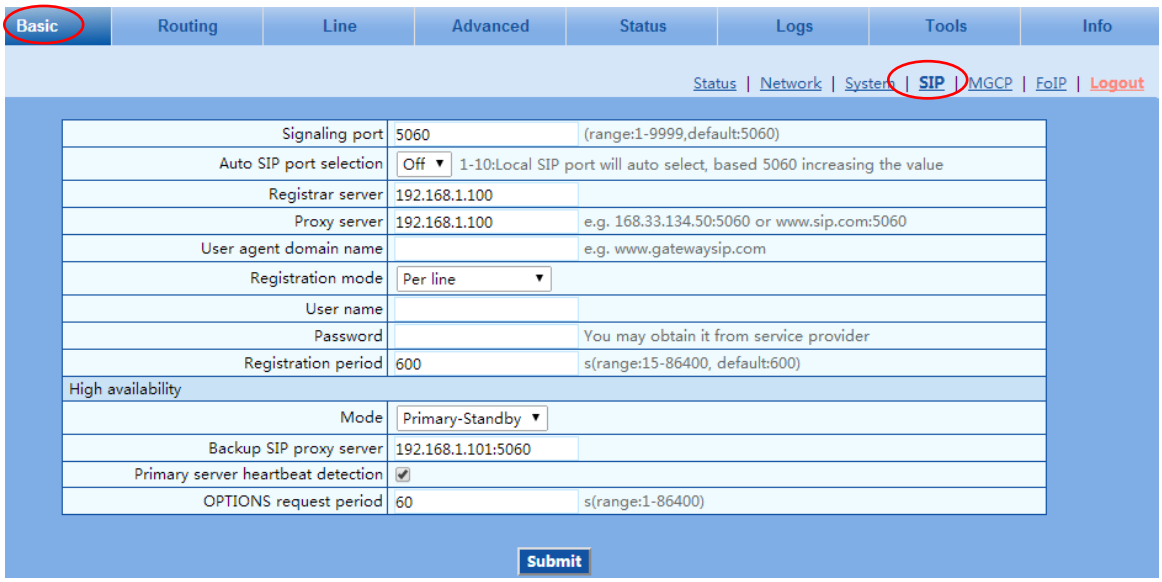
Figure 3-1 Server cluster



3.2 Configuring Primary-Standby Mode

Click **Basic** > **SIP**.

Figure 3-2 Primary-Standby configuration page



Here are the steps to configure the **Primary-Standby** mode.

Step4 Choose the **mode** as **Primary-Standby**.

Step5 Fill primary SIP server IP address or domain name in **Registrar serve**r.

Step6 Fill primary SIP server IP address in **Backup SIP proxy serve**r. (If a domain name is filled in step 2, skip this step.)

Step7 Select **Primary server heartbeat detection**, and configure the OPTIONS request period.

Step8 Click **Submit**.

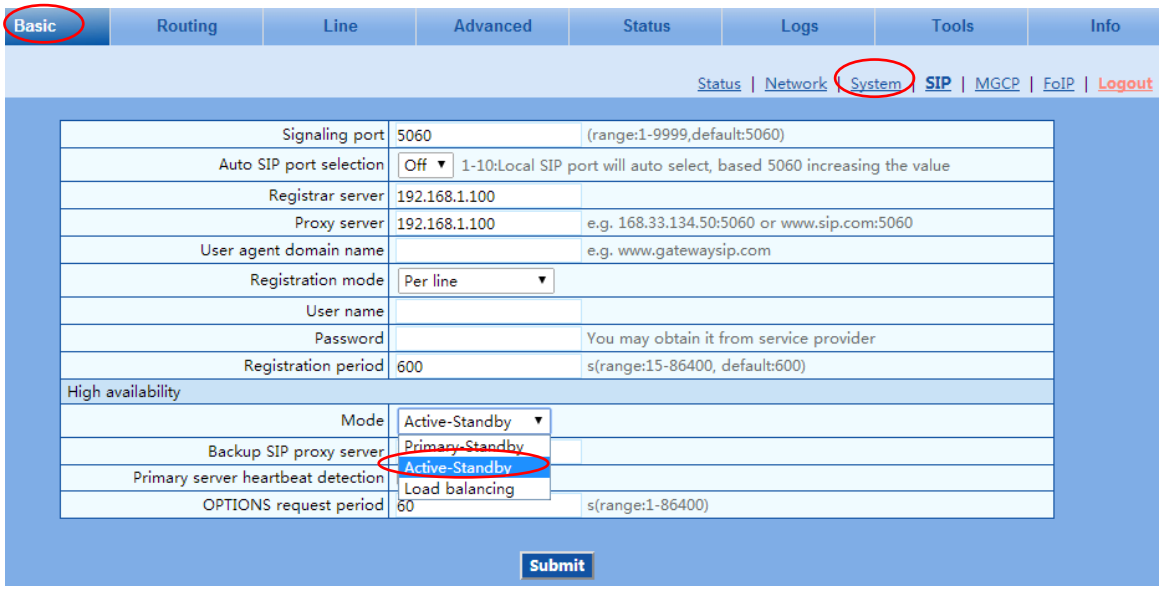
Step9 Click **Basic** > **Network**, check **Enable** in **DNS**， fill IP address in **Primary server**， and then submit. (This step is required if a domain name is filled in step 2).

3.3 Configuring Active-Standby Mode

3.3.1 Enable Active-Standby Feature

Enter the SIP trunk setting page, and click **Basic** > **SIP** > **Primary-Standby configuration** and choose **Active-standby**, then submit.

Figure 3-3 Active-Standby configuration page



3.3.2 Set SIP Servers

Configuring the IP Address of SIP Servers

Note: The IP address of the primary SIP server is configured on the top half of the SIP page.

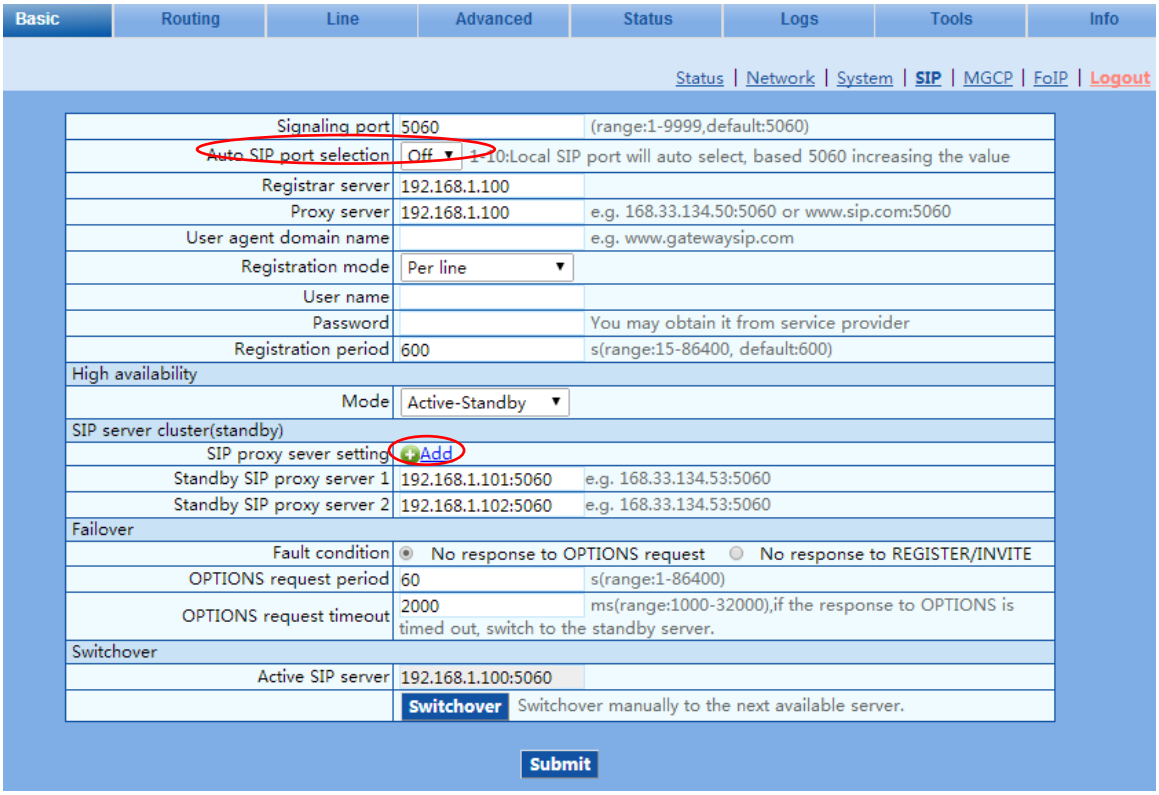
Here are the steps to configure the IP addresses of the standby SIP servers:

Step10 Ensure that active-standby feature is enabled.

Step11 Fill primary SIP server IP address in **Registrar serve**r, and then submit.

Step12 Click **Add** and fill the IP addresses for the standby SIP servers in **Standby SIP server**s.

Figure 3-4 Page to set registrar server



Configuring the Domain Name of the Primary Server

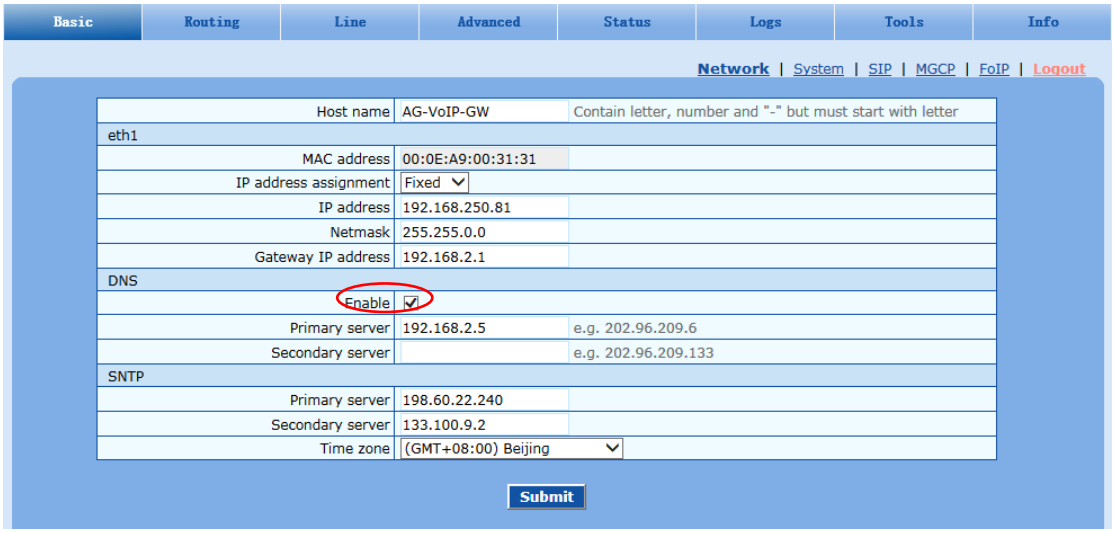
In case the primary SIP server is located through the domain name, the steps below should be follows:

Step1 Ensure that active-standby feature is enabled.

Step2 Fill registrar server domain name in **Registrar server**, then submit.

Step3 Click **Basic** > **Network**, check **Enable** in **DNS**，fill IP address in **Primary server**，and then submit.

Figure 3-5 Page to set DNS server



3.3.3 Set the Failover Condition

You should choose one of the following conditions:

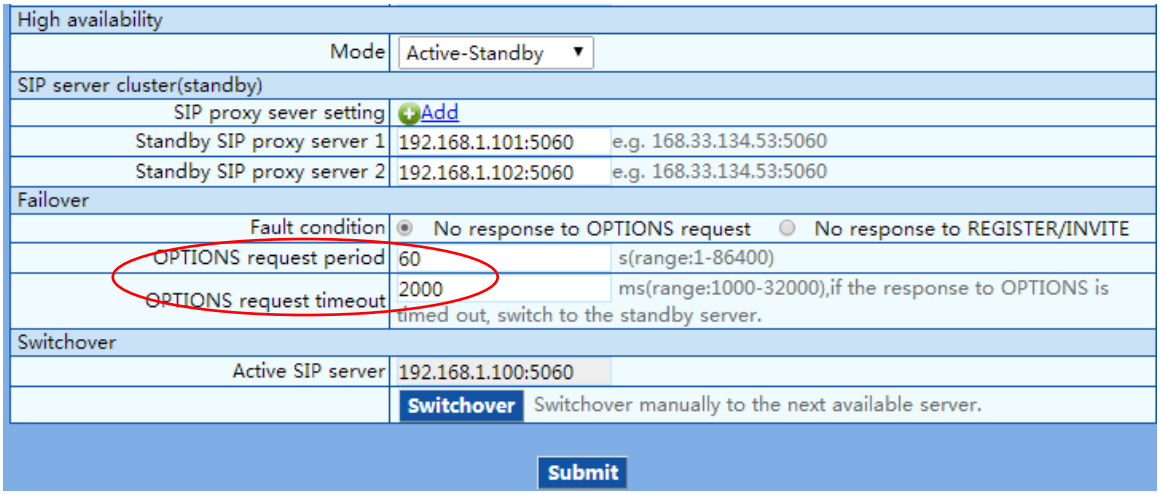
- No response to OPTIONS message
- No response to REGISTER/INVITE message

No Response to OPTIONS

When this condition is chosen the following timers need to be configured:

- **OPTIONS request period**: the interval between receiving the response (200) from the SIP server to the previous OPTIONS and sending the next OPTIONS.
- **OPTIONS request timeout**: the period since the sending of the last OPTIONS with no response by the SIP server.

Figure 3-6 Page to set failover condition



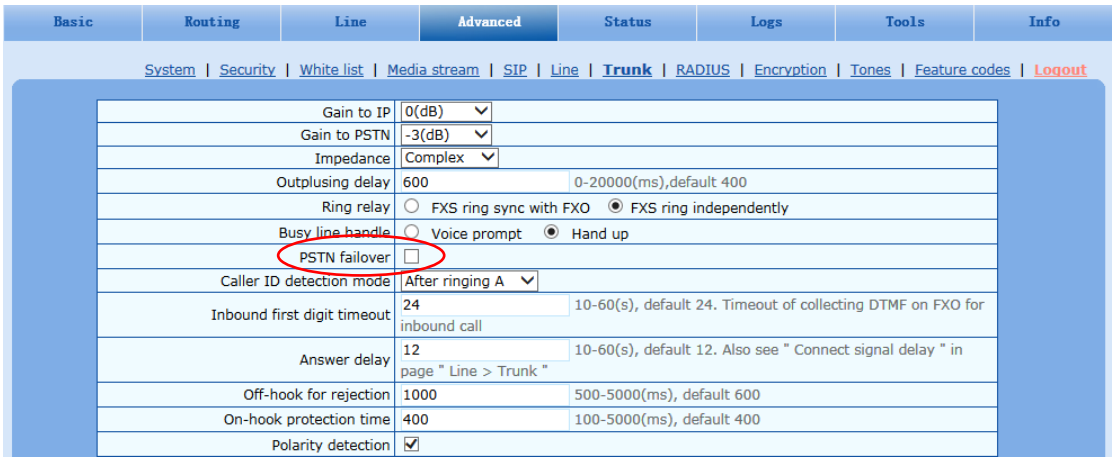
No Response to REGISTER/INVITE

When this condition is chosen, the gateway will failover to the standby SIP server if there is no response to the REGISTER or INVITE.

Note

When high availability feature is enabled, the PSTN failover feature in **Advanced > Trunk** should be disabled as shown in Figure 2-6.

Figure 3-7 Page to disable PSTN failover



3.3.4 How to Manually Perform Switchover

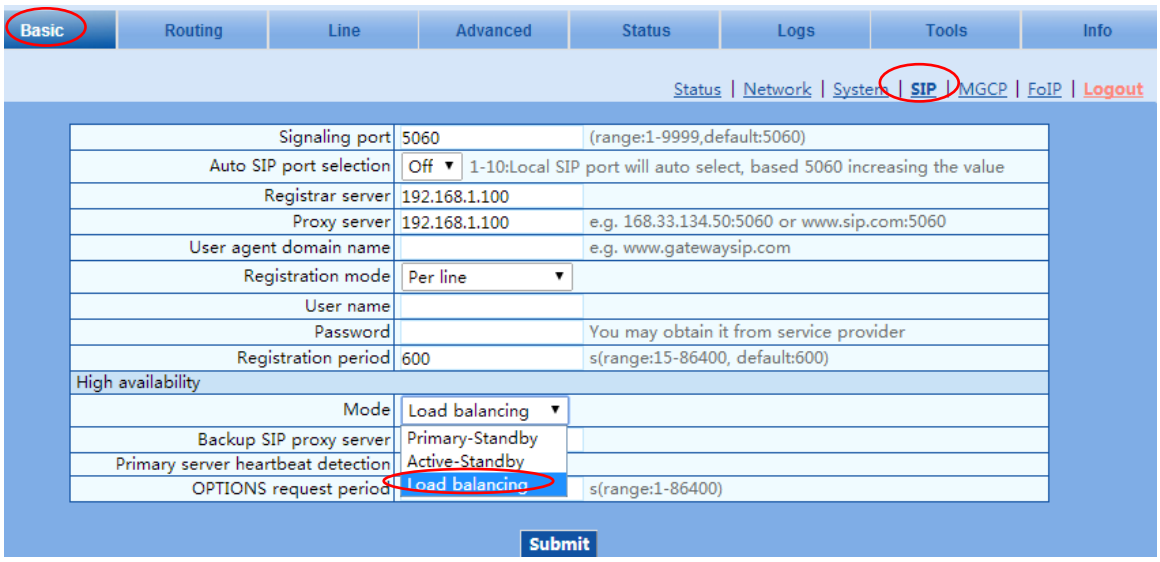
On the Wen GUI of the gateway, the **Switchover** button provides a means to manually switchover the call traffic from the current SIP server to the next available SIP server.

3.4 Configuring Load Balancing Mode

3.4.1 Enable Load Balancing Feature

Enter the SIP trunk setting page, and click **Basic > SIP > Primary-Standby configuration** and choose **Load balancing**, then submit.

Figure 3-8 Load balancing configuration page



3.4.2 Set SIP Servers

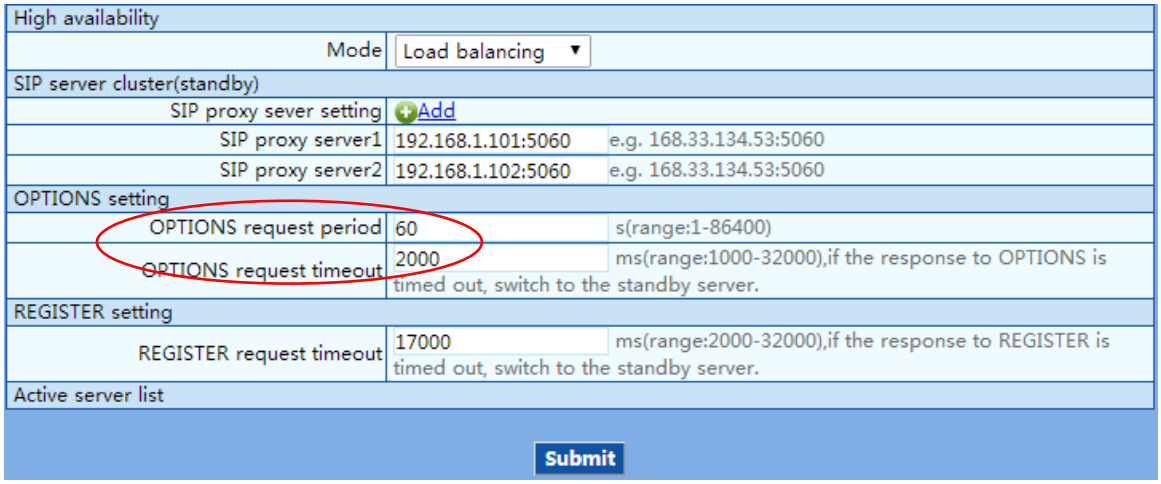
Refer to **3.3.2 Set SIP Servers**.

3.4.3 Configure OPTIONS Settings

In the active balancing mode, the following timers need to be configured:

- **OPTIONS request period**: the interval between receiving the response (200) from the SIP server to the previous OPTIONS and sending the next OPTIONS.
- **OPTIONS request timeout**: the period since the sending of the last OPTIONS with no response by the SIP server.

Figure 3-9 Page to configure OPTIONS settings

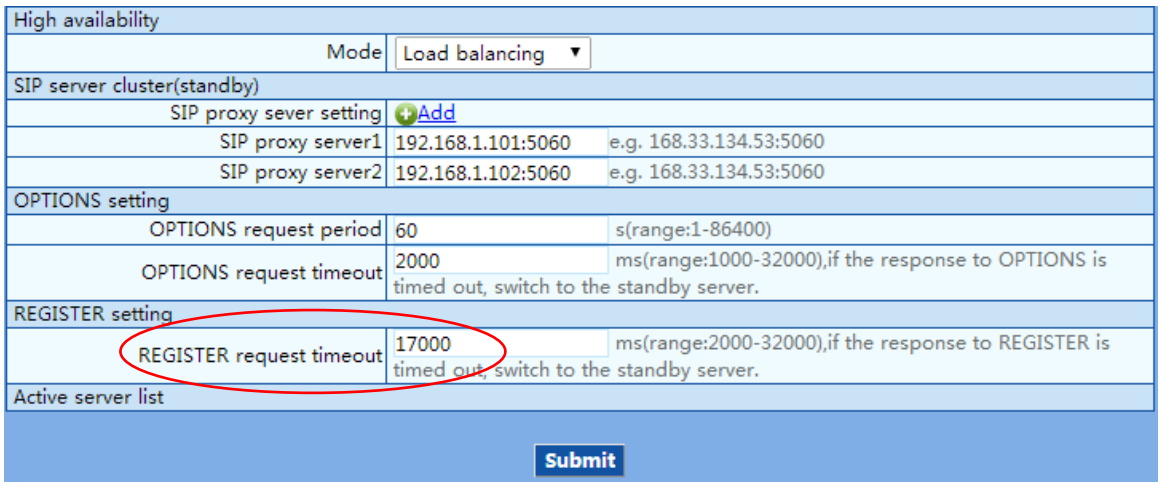


3.4.4 Configure REGISTER Settings

In the active balancing mode, the following time need to be configured:

- **REGISTER request timeout**: the period from the sending of the first REGISTER with no response by the previous SIP server to the sending of REGISTER to the next SIP server.

Figure 3-10 Page to configure REGISTER settings



3.4.5 Active Server List

All the SIP servers, on which the gateway or endpoints are registered on, will be listed in active server list.