DINSTAR

MTG2000 User Manual v1.0

Dinstar Technologies Co., Ltd.

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

Postal Code: 518052

Telephone: +86 755 61919966

Fax: +86 755 2645 6659

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

Website: www.dinstar.com, www.dinstar.cn

Revision Records

Document Name	MTG2000 User Manual v1.0
Document version	V1.0
Firmware version	2.05.01.04
Revised by	Ivanka Yuan
Date	2014.6.18

Table of Contents

1 Product Description

1.1 Overview

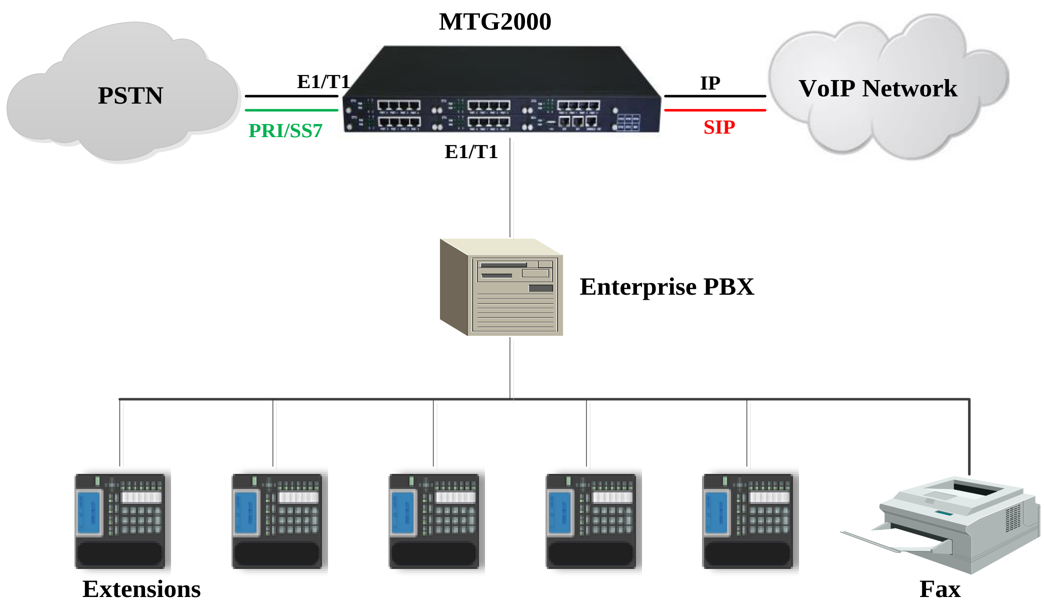
MTG2000 is a new-generation intelligent VoIP trunk gateway, featuring high integration and large capacity. Focusing on a concept of maintainable, manageable and operable, it provides carrier-grade VoIP and FoIP services, as well as value-added functions such as smart voice recognition, signal encryption and modem function.

MTG2000 supports a range of signaling protocols, realizing the interconversion between SIP and traditional signals like SS7 and PRI from the PSTN. Allowing users to choose 4, 8, 12, 16 or 20 E1/T1 ports, the trunk gateway also supports multiple codec methods and can improve the utilizing efficiency of trucking resources while ensuring voice quality. It can be connected with multiple devices such as softswitches, PBX and those servers equipped with digital trunk boards.

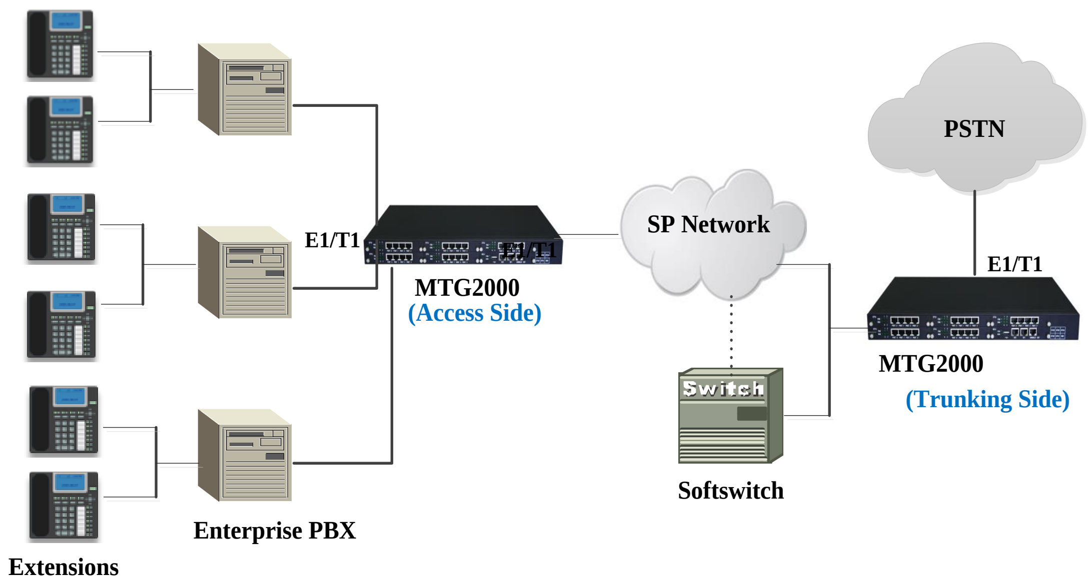
Compared to other similar products, MTG2000 has more advantages in terms of performance, system reliability, compatibility and price. The trunk gateway is ideally suitable for various access networks of SMEs, call centers, telecom operators and large-scale enterprises.

1.2 Application Scenario

The application scenario for enterprises is shown as follows:



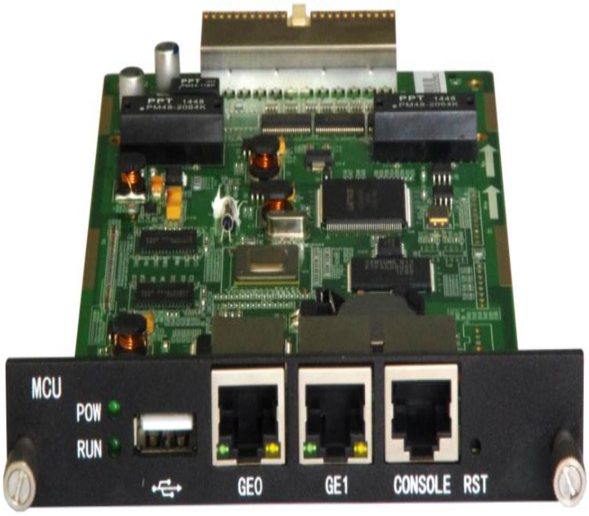
The application scenario for services providers is shown as follows:



1.3 Product Appearance

1.3.1 Image of MTG2000

1.3.2 Image of MCU and DTU



MCU (Main Control Unit)

DTU (Digit Trunk Unit)

MTG2000 allows users to insert or pull out DTU boards and can automatically identify the DTU boards that have been inserted. When a DTU board is inserted or pulled out, users need to re-configure the MTG2000 device.

1.3.3 Description of Ports and Indicators

MTG2000 has one MCU board and five DTU boards, which can be inserted or pulled out. Each board has four E1/T1 ports (from 0 to 3 in sequence), and there are indicators to show the status of each E1/T1 port.

MCU Board:

Indicator/Port	Status	Description
POW	On green	Power supply is normal
	On dull	There is no power supply or power supply is abnormal
RUN	Flash slowly	The MCU board has been inserted and identified by the system
	Flash quickly	The system does not identify the MCU board
CONSOLE	/	The console port used to carry out maintenance-related configurations, with a baud rate of 115200bps
GE1	/	The gigabit Ethernet port for services, which is used to realize the data transmission of signal or voice. Its default IP address is 192.168.1.111, and default netmask is 255.255.255.0. When data is transmitted at a rate of 1000Mbps, the indicator on the left flashes while the indicator on the right is on orange.

		When data is transmitted at a rate of 100Mbps, the indicator on the left flashes while the indicator on the right is on dull.
GE0	/	The gigabit Ethernet port for network management; its default IP address is 192.168.11.1, and default netmask is 255.255.255.0. When data is transmitted at a rate of 1000Mbps, the indicator on the left flashes while the indicator on the right is on orange. When data is transmitted at a rate of 100Mbps, the indicator on the left flashes while the indicator on the right is on dull.
RST	/	The button is used to restart MTG2000

DTU Board:

Indicator/Port	Status	Description
POW	On green	Power supply is normal
	On Dull	There is no power supply or power supply is abnormal
RUN	Flash slowly	The DTU board has been inserted and identified by the system
	Flash quickly	The system does not identify the DTU board
E1/T1	On dull	The corresponding E1/T1 port is not in use.
	On green	The corresponding E1/T1 port is connected normally, and can be used to receive or send data.
	Flash	The corresponding E1/T1 port is connected falsely and there are bit errors.

1.4 Functions and Features

1.4.1 Key Features

- Multi-port and high-integrated structure: up to 20 E1/T1 with 1U size;
- Provide various services such as VoIP, FoIP, Modem and POS;
- Support flexible dialing rules and operations, allowing users to customize dialing rules according to different working environments;
- Support multiple coding standards: G.711A/U, G.723.1, G.729A/B and iLBC;
- High compatibility: interoperable with PBX of Avaya, NEC and Alcatel and leading softswitches of Huawei, Cisco and ZTE.

1.4.2 Protocols Supported

- Standard SIP v2.0（UDP/TCP）
- SIP Rport
- PRI/SS7 Protocol
- Dynamic NAT
- SIP Trunk Working Mode: Peer/Access
- Hypertext Transfer Protocol (HTTP)
- ITU-T G.711A-Law/U-Law, G.723.1, G.729AB, iLBC13k/15k, AMR
- Domain Name System (DNS)
- TFTP/FTP
- RFC3262, 3263, 3264, 3265, 3515, 2976, 3311
- RTP/RTCP, RFC2198, 1889
- SIP-T, RFC3372, RFC3204, RFC3398

1.4.3 System Functions

- Packet Loss Concealment (PLC)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Echo Cancellation
- Packet Loss Compensation
- Silence Suppression
- Adaptive Jitter Buffer
- Gain Control of Voice and Fax
- Support Modem and POS
- DTMF Modes: RFC2833, SIP INFO and INBAND
- T38/Pass-Through Fax over IP
- Configurations via HTTP/Telnet
- Upgrade Firmware via TFTP/Web
- Recognition of Prompt Tone

1.4.4 Physical Interfaces

- E1/T1 Ports: 4/8/12/16/20
- DTU Module: 4 E1/T1
- Interface Type: RJ48(Impedance 120Ω）
- Ethernet Interface:
  GE1: 100/1000 Mbit BaseT Adaptive Ethernet
  GE0: 100/1000 Mbit BaseT Adaptive Ethernet

- Serial Port : 1* RS232, 115200bps

1.4.5 Software Features

- Local/Transparent Ring Back Tone
- Overlapping Dialing
- Multiple Dialing Rules
- PSTN Group Based on E1 Port or E1 Timeslot
- Configuration of IP Trunk Group
- Voice Codec Group
- Caller/Called Number White List
- Caller/Called Number Black List
- Access Rule List
- IP Trunk Priority
- RTP and Signaling Encryption (VOS RC4)

1.4.6 Call Features

- Flexible Route Methods: PSTN–PSTN, PSTN–IP, IP–IP, IP–PSTN
- Intelligent Routing Rules
- Call Routing Based on Time
- Call Routing Based on Prefix of Caller/Called Number
- Caller and Called Number Manipulation

1.4.7 Hardware Specifications & Environment

- Redundant Power
- Power Supply: 100 ~ 240V AC, 50 ~60Hz
- Power Consumption: 45W
- Operating Temperature. 0 ℃ ~ 45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90%, Non-Condensing
- Dimensions (W/D/H): 436×300×44.5mm (1U)
- Unit Weight: 3.8kg
- Compliance: CE, FCC

2 Quick Installation

2.1 Preparations before Installation

2.1.1 Attentions for Installation

The attentions for installing MTG2000 include:

- To guarantee MTG2000 works normally and to lengthen the service life of the device, the humidity of the equipment room where MTG2000 is installed should be maintained at 10%-90% (non-condensing), and temperature should be 0 ℃ ~ 45 ℃;
- Ensure the equipment room is well-ventilated and clean;
- Power supply of MTG2000 should be 100 ~ 240V AC, and its socket is a three-pin socket which should be grounded well;
- It's suggested that personnel who has experience or who has received related training be responsible for installing and maintaining MTG2000;
- Please wear anti-static wrist strap when installing MTG2000;
- Please do not hot plug or unplug cables;
- It's advised to adopt uninterruptible power supply.

2.1.2 Preparations about Installation Site

- Equipment Cabinet

  Ensure the cabinet is well-ventilated and strong enough to bear the weight of MTG2000. It's required that the width of the shelf should be 482.6mm (19 inches).
- Trunk

  Ensure telecom operator has approved to open a trunk.
- IP Network

  Ensure Ethernet PBX or router under IP network has been prepared, since MTG2000 is connected to the IP network through the standard 10/100/1000M Ethernet port.
- Socket

  Ensure the socket of MTG2000 is a three-pin socket and power supply is grounded well.

2.1.3 Installation Tools

- Screwdriver
- Anti-static wrist strap
- Ethernet cables, power wires, telephone wires
- Hub, telephone set, fax, and PBX
- Terminal (can be a PC which is equipped with hyperterminal simulation software)

2.1.4 Unpacking

Open the packing container to check whether the MTG2000 device and all accessories have been in it:

- One MTG2000 device
- One meter long of power wire (AC 250V/4A)
- E1/T1 transit boxes (the number of the transit box is the same with that of E1/T1 ports)
- One network cable
- E1/T1 cables (the number of the cables is the same with that of E1/T1 ports)
- Serial console cable

2.2 Installation of MTG2000

2.2.1 Put MTG2000 into Shelf

1. Use screws to fix a flank on the left and the right of MTG2000 respectively;
2. Put the MTG2000 device into the shelf horizontally;
3. Fix the flanks of MTG2000 on the cabinet by using screws.

2.2.2 Connect Ground Wire to MTG2000

Connect one end of the ground wire to the grounding port on the back of MTG2000 and then connect the other end to the grounding bar of the shelf.

2.2.3 Connect MTG2000 to Ethernet

MTG2000 has two network ports, namely the gigabit Ethernet port for services (GE1) and the gigabit Ethernet port for network management (GE0). It is advised to connect GE1 to the IP network.

Both GE1 and GE0 can be used to carry out management on MTG2000, but only GE1 is put in use generally. GE0 is used when there is a need to separate the management on MTG2000 from the service processing of the MTG2000.

2.2.4 Connect MTG2000 to PSTN

Generally, a distribution frame needs to be used for the connection between MTG2000 and PSTN. Firstly, connect one end of E1 cable to one of the E1/T1 ports of MTG2000, and then connect other end to the E1 port of the distribution frame. Secondly, connect one end of an electric cable to the distribution frame, and then connect other end to an exchanger or a PBX under PSTN.

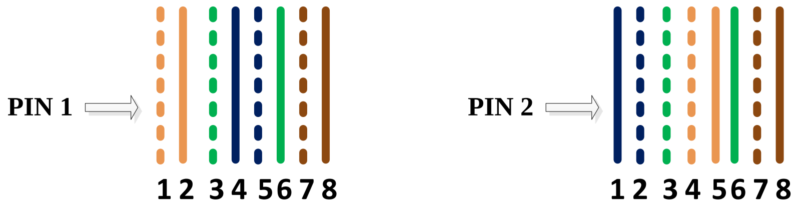
2.3 Cabling of E1/T1 Port

If there is a need to deploy multiple cables, it'd better to make a mark on each cablee and write down IP address and destination port in order to simplify the follow-up connection, debugging and maintenance.

2.3.1 How to make RJ－48 joint for E1/T1 Cable

1. Prepare a twisted-pair cable with a length of at least 0.6 meters, and then remove the shuck of the cable as follows:

2. Sequence the lines of the cable according to the following figure.



3. Put the lines into two pins of RJ—48 joint according to the abovementioned sequence of the lines.

4. Use a RJ—48 wire crimper to crimp the RJ—48 joint.

Note: Generally, a RJ—48 cable will be provided together with the MTG2000 device, and users have no need to make RJ—48 joints by themselves.

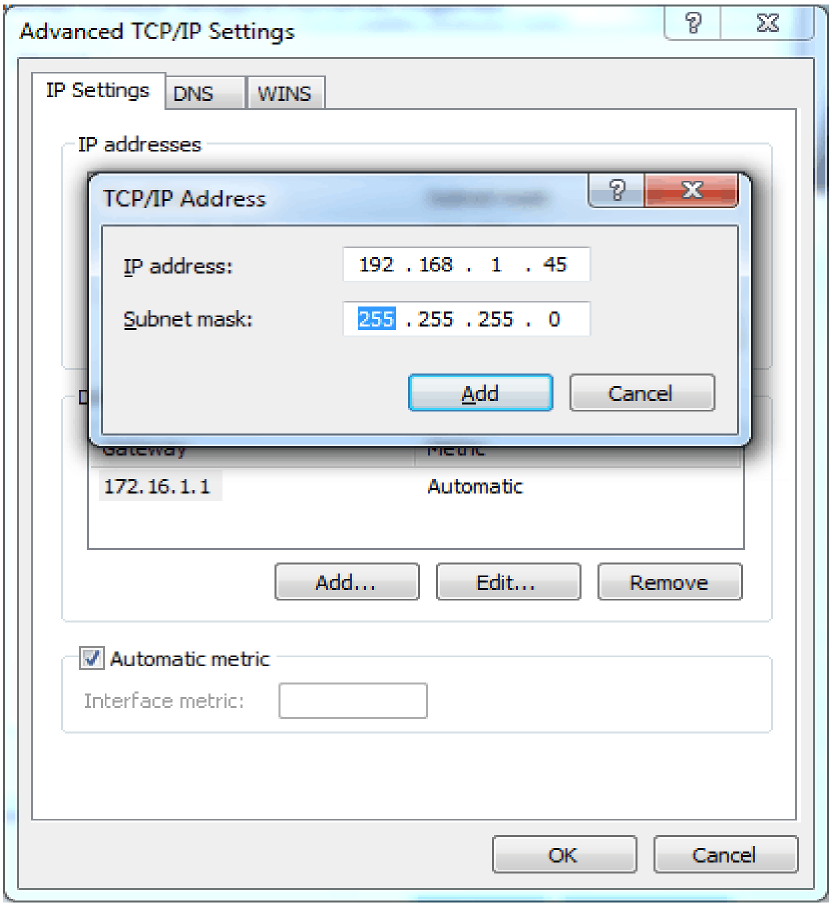
3 Basic Operation

3.1 Configuration of IP Address

The default IP address of GE1 is 192.168.1.111, while that of GE0 is 192.168.11.1.When GE1 is in use, it's required that the IP address of GE1 and the IP address of PC are at the same network segment.

1. Connect the GE1 port of MTG2000 to a PC by using a network cable.

2. Open the **TCP/IP Settings** interface, click **Advanced**, and then click **Add** to add an IP whose format is 192.168.1.XXX. Or you can open the Internet Protocol (TCP/IP) interface to modify an existing IP into 192.168.1.XXX.



3.2 Local Maintenance

To ensure easy maintenance, the MTG2000 trunk gateway provides a standard RJ48 console port, which has a Baud rate of 115200bps. Users can log in the MTG2000 to carry out maintenance-related configurations through the console port.

3.2.1 Example: Log in MTG2000 via Console Port

Step 1: Prepare a serial cable.

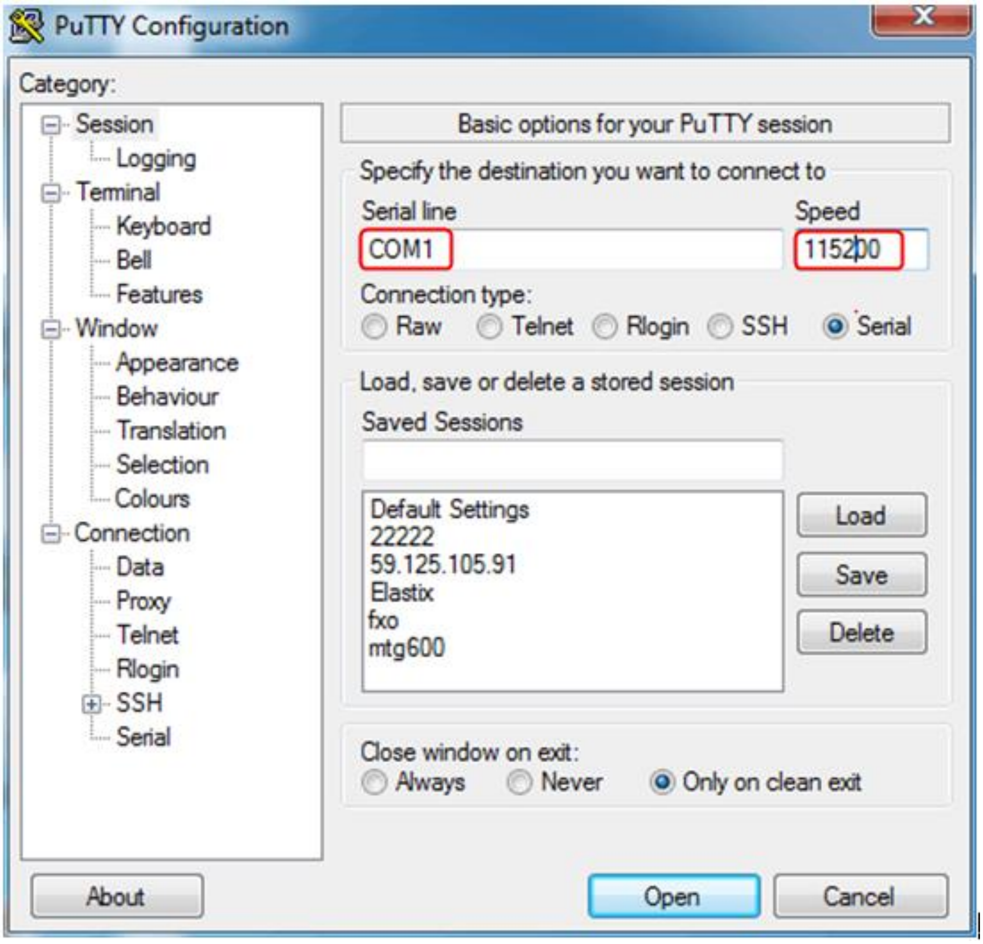
Step 2: Connect the F port of the serial cable to the COM port of PC.

If the PC does not have a COM port, please use a USB-to-COM converting line to connect the serial cable to the PC.

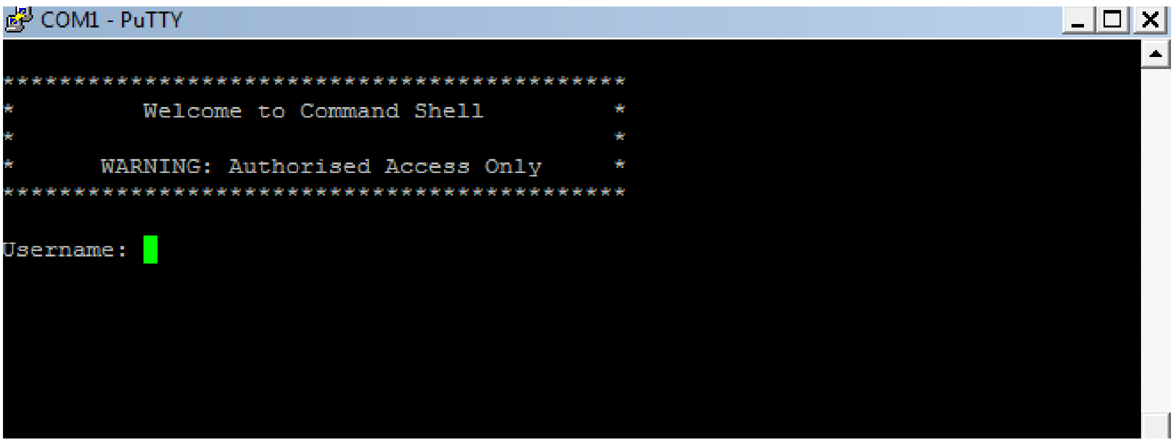
Step 3: Connect the M port of the serial cable to the console port of MTG2000.

Step 4: Conduct configurations on login software.

Herein we take the PuTTY sofeware as an example. Detailed configurations are as follows: (COM1 is an example. Please enter correct serial line according to actual conditions.)



After finishing the above configuration, click the **Open** button to enter the following interface.



Enter username and password, which are the same with the username and password of the Web of MTG2000. And then you will see a linux platform where you can carry out maintanance-related configurations.

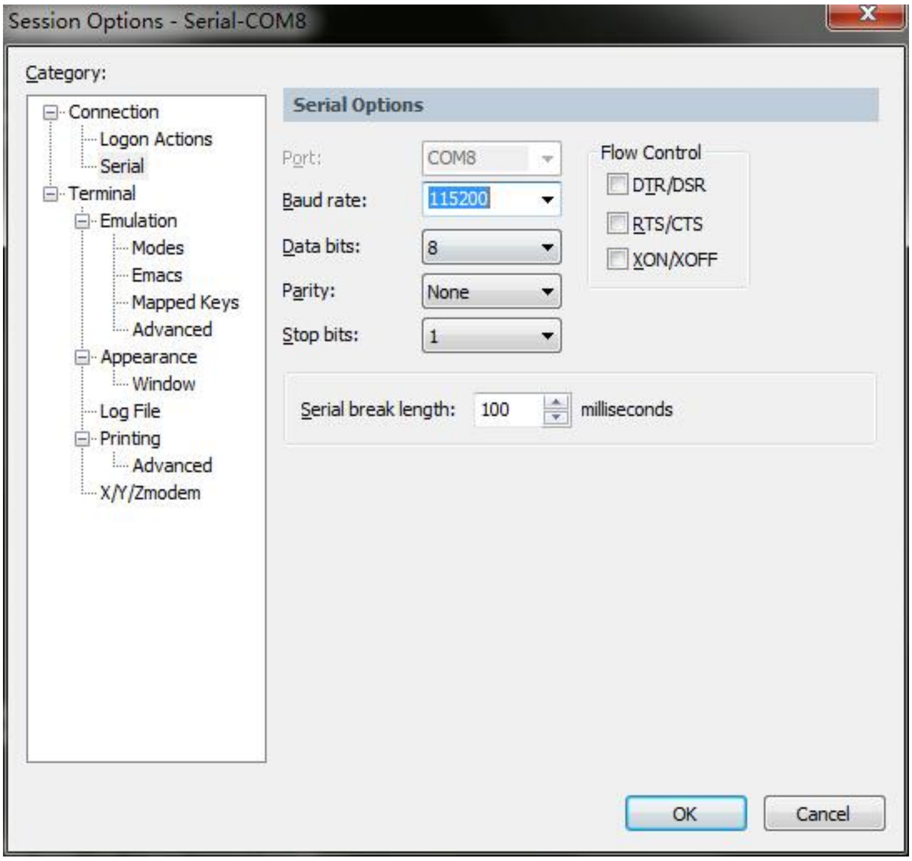
Note: For commands to query MTG2000 information, make reference to Chapter 6.

3.3 Query IP

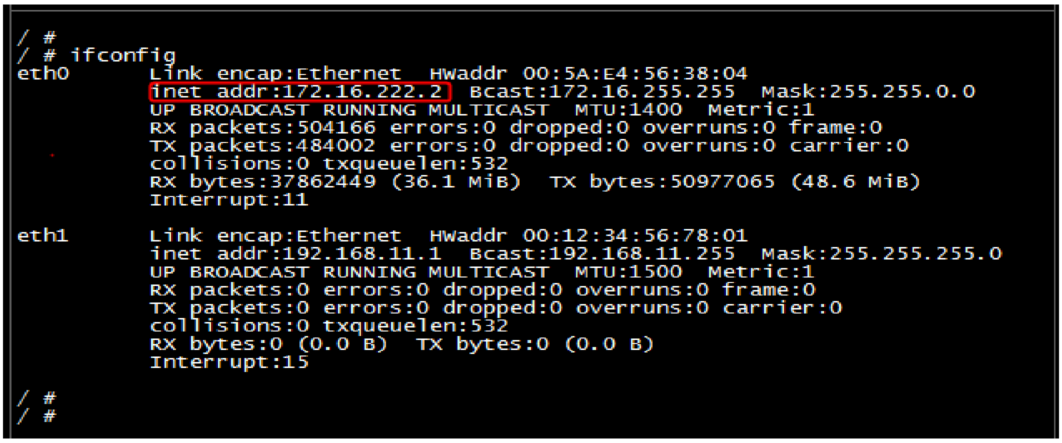
If you have changed the default IP address of GE1 or GE0 to a new IP address but forget it, you can carry out the following procedures to query the IP address.

1. Use a serial line to connect the console port of MTG2000 with a PC;

2. Modify the baud rate to 115200;



3. Click **OK**, and then enter ‘ifconfig’, and the IP address of GE1 or GE0 of MTG2000 will be displayed.

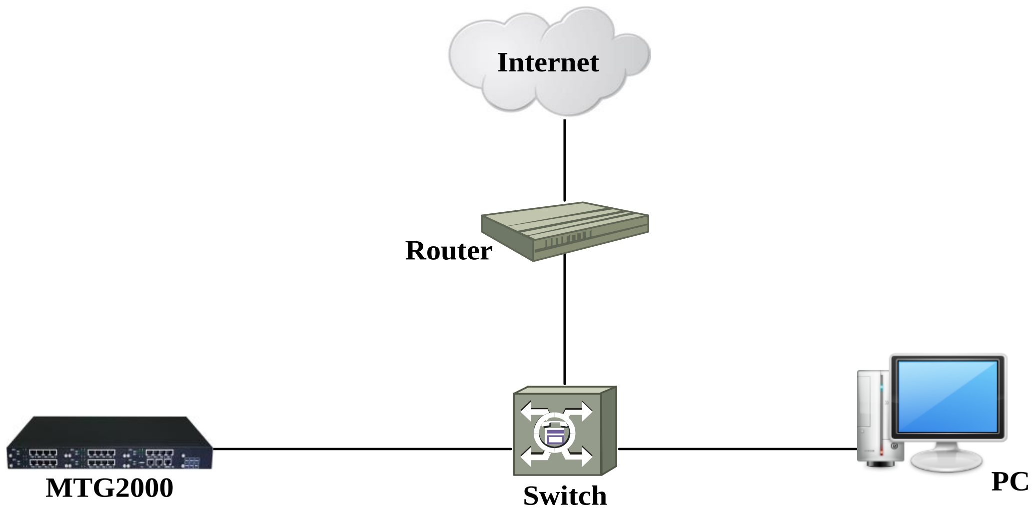


4 Configurations on Web Interface

4.1 How to Log in Web Interface

4.1.1 Network Connection

Connect MTG2000 to the network according to the following network topology:



4.1.2 Preparations for Login

Modify the IP address of the PC to make it at the same network segment with the IP address of GE1 port of MTG2000 device. The format of PC IP is 192.168.1.XXX, since the default IP of GE1 port is 192.168.1.111.

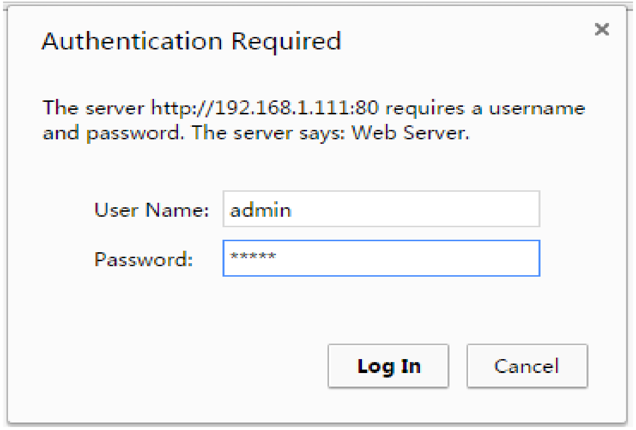
Check the connectivity between the PC and the MTG2000. Click **Start** → **Run** of PC and enter cmd to execute 'ping 192.168.1.111' to check whether the IP address of the MTG2000 runs normally.

4.1.3 Log in Web Interface

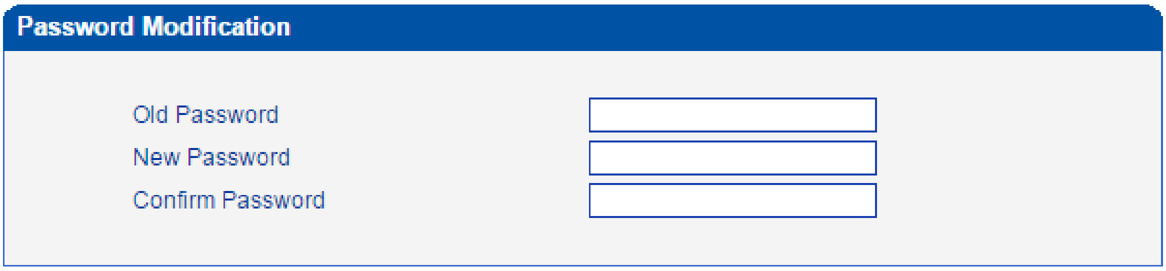
Open a web browser and enter the IP address of GE1 of MTG2000 (the default IP is 192.168.1.111). Then the login GUI will be displayed. Both the default username and password are admin.

It is suggested that you should modify the username and password for security consideration on the **Maintenance** → **Password Modification** interface.

Login GUI:



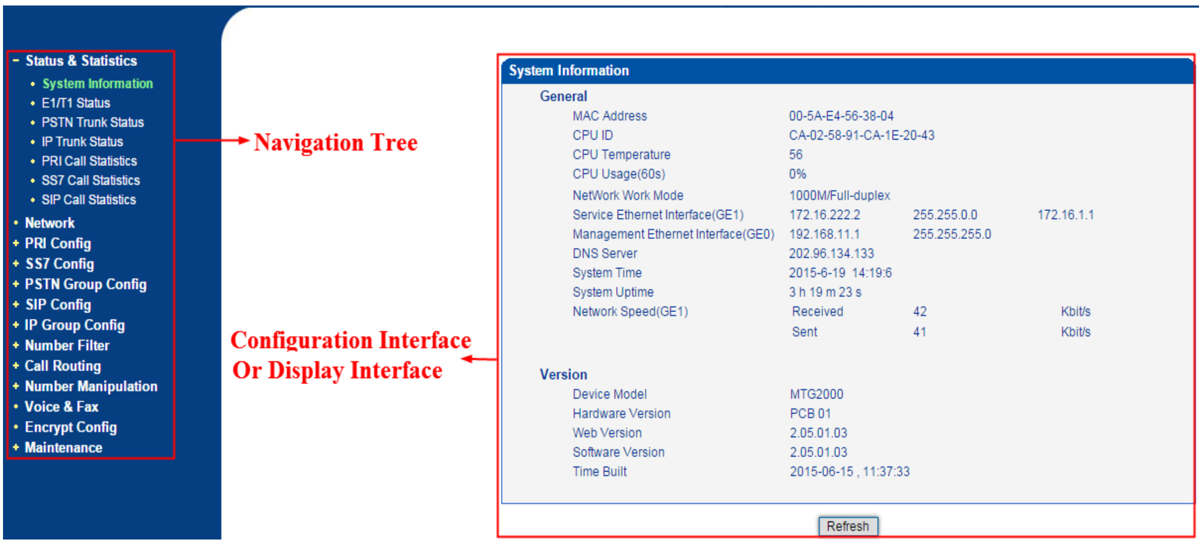
Password Modification Interface:



4.2 Introduction to Web Interface

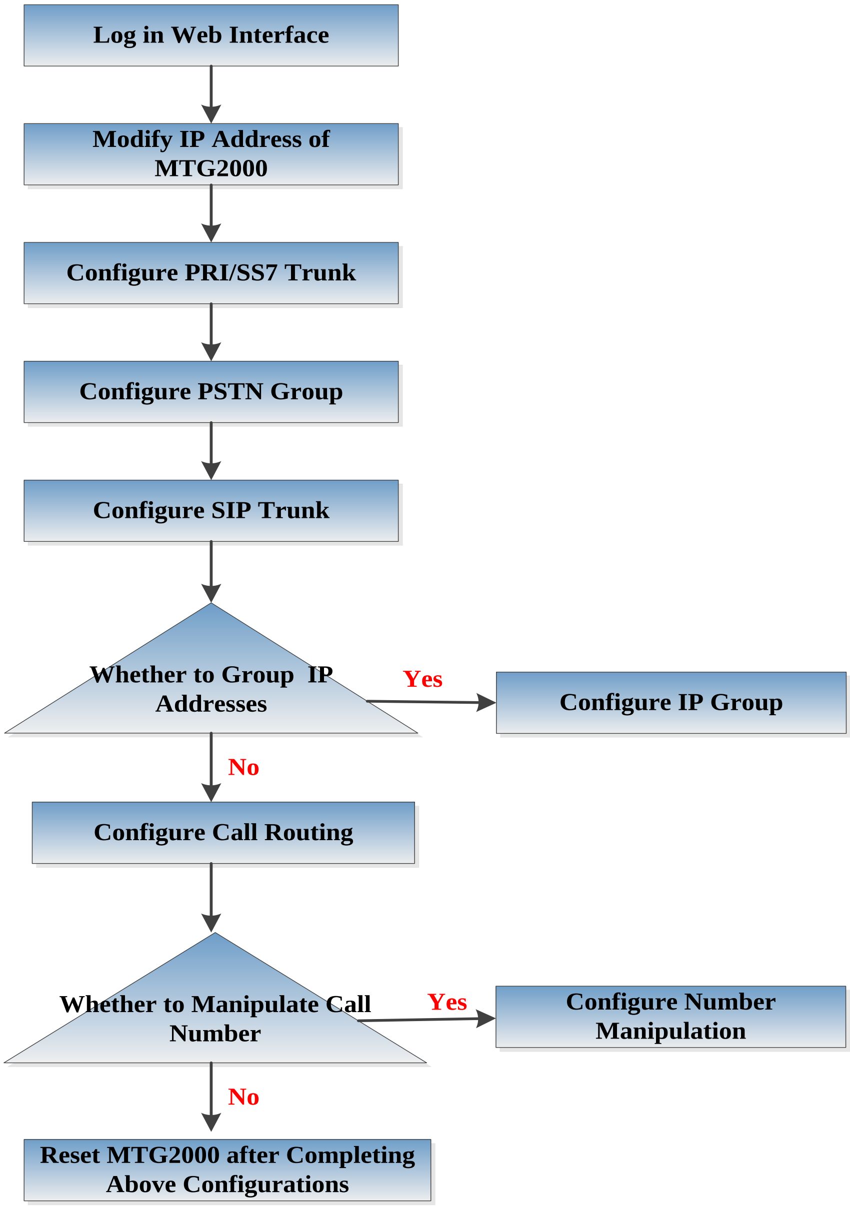
The Web Interface of the MTG2000 consists of the navigation tree and detailed configuration interfaces.

Click a node of the navigation tree, and you will see a detailed display interface or configuration interface:



4.3 Configuration Flows

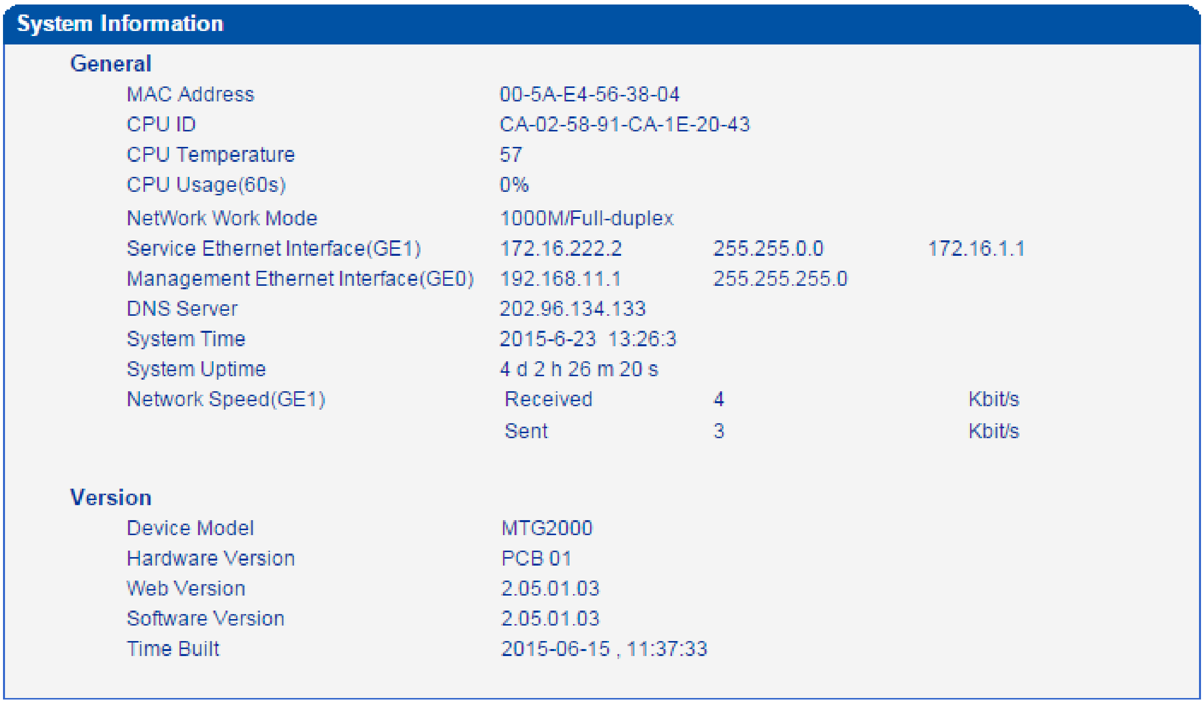
The following is the configuration flows of MTG2000:



4.4 Status & Statistics

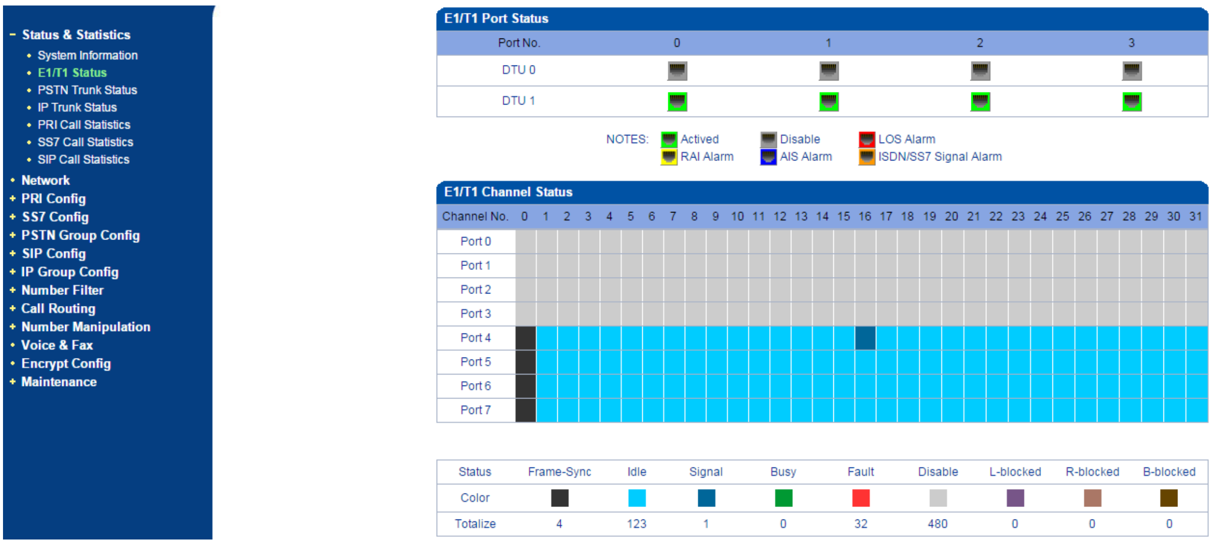
4.4.1 System Information

Click **Status & Statistics → System Information** in the navigation tree on the left, and the following interface will be displayed. On the interface, information about the system, such as Mac address, CPU usage, hardware version and software version, are shown.



4.4.2 E1/T1 Status

Click **Status & Statistics** → **E1/T1 Status** in the navigation tree, and the status of each E1/T1 port is displayed.



	Actived	Both physical connection and signal connection of the E1/T1 port are normal, and the port is activated.

Status of E1/T1 Port	Disable	The E1/T1 port is not used.
	LOS Alarm	Alarm for loss of signal. If the LOS alarm is raised, please check physical network connection.
	RAI Alarm	RAI (Remote Alarm Indication) is an alarm for lost of remote signal. The alarm is sent by the remote device and received by MTG2000.
	AIS Alarm	AIS (Alarm Indication Signal) is an alarm raised by MTG2000, indicating the peer device malfunctions, or signal/physical connections are abnormal.
	ISDN/SS7 Signal Alarm	This alarm means physical connection is normal while signal connection is abnormal.
E1/T1 Channel Status	Frame-Sync	Frame synchronization
	Idle	The channel is available, and related cables are connected normally.(The channel is used to transmit voice)
	Signal	The channel is used to transmit signal.
	Busy	The E1/T1 channel is being used by voice.
	Fault	The channel is normal while cables are not successfully connected.
	Disable	The E1/T1 trunk is not used.
	L-blocked	The E1/T1 channel is blocked at local end, but not blocked at remote end.
	R-blocked	The E1/T1 channel is blocked at remote end, but not blocked at local end.
	B-block	The E1/T1 is blocked at both local end and remote end.

4.4.3 PSTN Trunk Status

On the **PSTN Trunk Status** interface, the statuses of PRI/SS7 trunks are displayed. The PRI/SS7 trunks under PSTN need to be established at the **PRI Config → PRI Trunk** interface or the **SS7 Config →SS7 Trunk** interface first.

PRI Link Status

PRI Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
1	pri0	15	Fault

SS7 Link Status

SS7 Trunk No.	Trunk Name	E1/T1 Port No.	Link Status
0	ss7-2	4	Established

4.4.4 IP Trunk Status

On the **IP Trunk Status** interface, the statuses of SIP trunks are displayed. The SIP trunks need to be established at the **SIP Config → SIP Trunk** interface first.

SIP Trunk Status

Trunk No	Trunk Name	Trunk Mode	Protocol Type	Username	Incoming Authentication Type	Link Status
2	118.159	Peer	UDP	---	IP Address	Established

Parameter	**Explanation**
Trunk Name	This trunk name is the name used to register the SIP trunk. If the SIP trunk is not registered, the trunk name is displayed as "---".
Trunk Mode	There are two trunk modes: peer (peer-to-peer) and access.
Incoming Authentication Type	Incoming calls can be authenticated through password or IP address.
Link Status	There are two link statuses: Established and Fault.

4.4.5 PRI Call Statistics

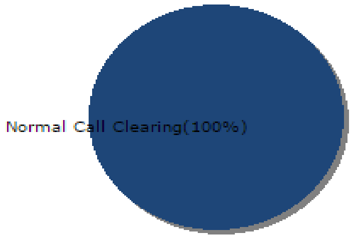
On the **PRI Call Statistics** interface, information about PRI calls and statistics about call release causes are displayed.

ASR (**Answer-seizure Ratio):** is a call success rate, which reflects the percentage of answered telephone calls with respect to the total call volume. ASR = answered call/total attempts of calls.

ACD (**Average Call Duration):** is a measurement in telecommunication, which reflects an average length of telephone calls transmitted on telecommunication networks. ACD = total call duration/total connected calls.

PRI Trunk Call Statistics					
PRI Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR	ACD
1	pri0	0	0	100%	0

Release Cause Statistics	
Normal Call Clearing	0
Call Reject	0
User Busy	0
No User Response	0
No Circuit Available	0
Unassigned Number	0
Normal, Unspecified	0
Others	0

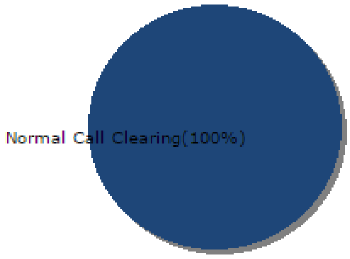


4.4.6 SS7 Call Statistics

On the **SS7 Call Statistics** interface, information about SS7 calls and statistics about call release causes are displayed.

SS7 Trunk Call Statistics					
SS7 Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR	ACD
0	ss7-2	0	0	100%	0

Release Cause Statistics	
Normal Call Clearing	5
Call Reject	0
User Busy	0
No User Response	0
No Circuit Available	0
Unassigned Number	0
Normal, Unspecified	0
Others	0



4.4.7 SIP Call Statistics

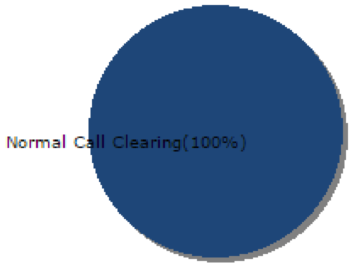
On the **SIP Call Statistics** interface, information about SIP calls and statistics about call release causes are displayed.

SIP Trunk Call Statistics

SIP Trunk No.	Trunk Name	Current Calls	Accumulated Calls	ASR	ACD
2	118.159	0	5	100%	38

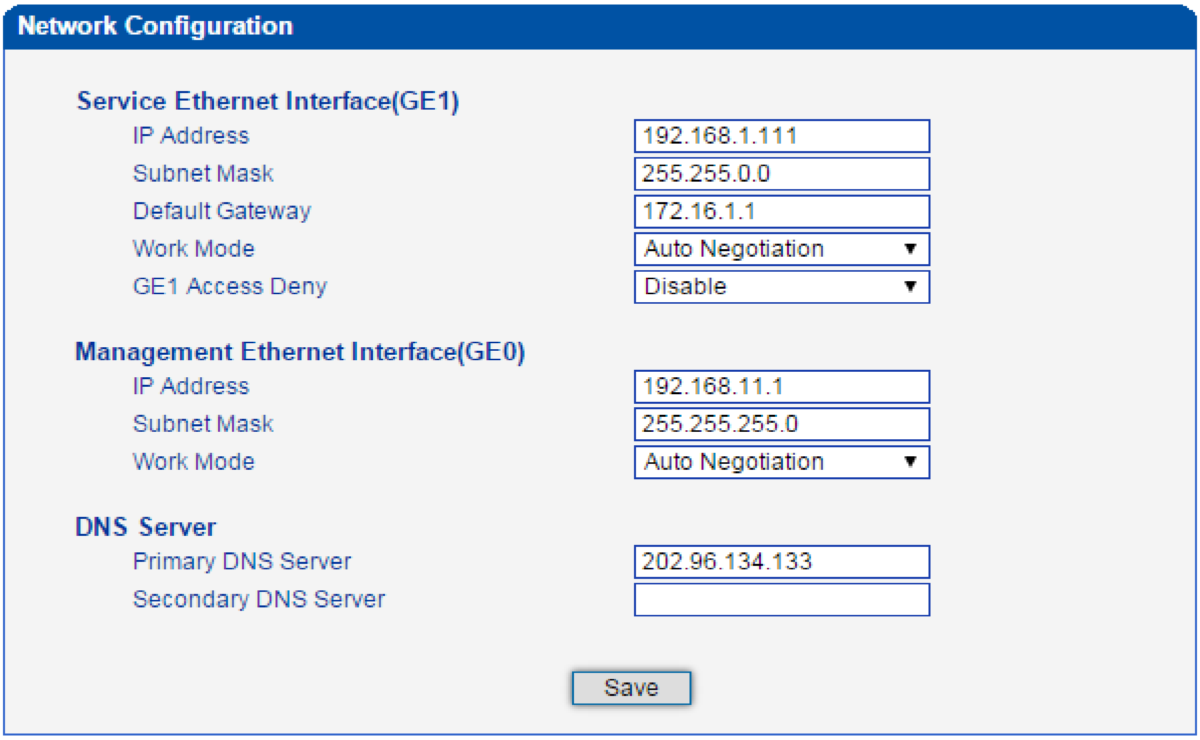
Release Cause Statistics

Normal Call Clearing	5
Temporarily Unavailable	0
Forbidden	0
Not Found	0
Busy Here	0
Internal Server Error	0
Server Time Out	0
Service Unavailable	0
Others	0



4.5 Network

Generally, it's necessary to modify the default IP address of GE1 according to actual network conditions, and then modify the IP address of PC to make it at the same network segment with the IP address of GE1. After completing the configurations, you need to restart the MTG2000 device for the changes to take effect.



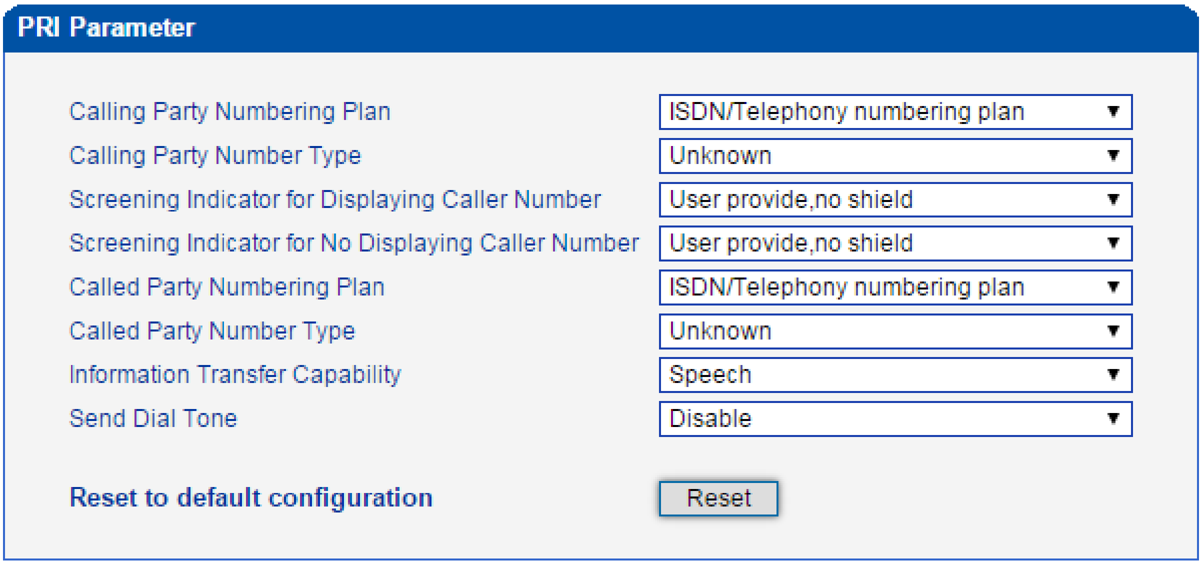
Belong to	Parameter	Explanation
GE1 Port	IP Address	The IP address of GE1, default value is 192.168.1.111
	Subnet Mask	Subnet mask of GE1
	Default Gateway	The IP address of network gateway
	Work Mode	Include Auto Negotiation, 1000M/Full-Duplex, 100M/Full-Duplex, 100M/Half-Duplex. Full-Duplex: Communication in both directions simultaneously; Half-Duplex: Communication only in one direction.
	GE1 Access	Deny: Users can not access the Web interface through GE1, but MTG2000 works normally. Allow: All users can access the Web interface through GE1.
GE0 Port	IP Address	The IP address of GE0, default value is 192.168.11.1
	Subnet Mask	Subnet mask of GE0
	Work Mode	Same with Word Mode of GE1
DNS	Primary DNS Server	The IP address of the primary DNS server
	Secondary DNS Sever	The IP address of the secondary DNS server. It is optional to fill in.

Note: The IP address of GE1 and that of GE0 cannot be at the same network segment.

4.6 PRI Config

4.6.1 PRI Parameter

Configure PRI parameters according to actual data which are provided by telecom operators.

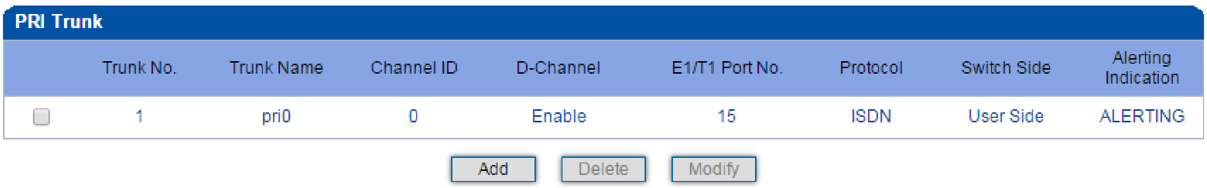


Parameter	Options
Calling Party Numbering Plan	Include ‘ISDN/Telephony Numbering Plan’, ‘Data Numbering Plan’, ‘Telex Numbering Plan’, ‘National Standard Numbering Plan’, ‘Private Numbering Plan’ and ‘Unknown’.
Calling Party Number Type	Include ‘International Number’, ‘National Number’, ‘Network Specific Number’, ‘Subscriber Number’, ‘Abbreviated Number’ and ‘Unknown’.
Screening Indicator for Displaying Caller Number	Include ‘User-provided, not screened’, ‘User-provided, verified and passed’, ‘User-provided, verified and failed’, ‘Network-provided’
Screening Indicator for No Displaying Caller Number	Include ‘User-provided, not screened’, ‘User-provided, verified and passed’, ‘User-provided, verified and failed’, ‘Network-provided’
Called Party Numbering Plan	Include ‘ISDN/Telephony Numbering Plan’, ‘Data Numbering Plan’, ‘Telex Numbering Plan’, ‘National Standard Numbering Plan’, ‘Private Numbering Plan’ and ‘Unknown’.
Called Party Number Type	Include ‘International Number’, ‘National Number’, ‘Network Specific Number’, ‘Subscriber Number’, ‘Abbreviated Number’ and ‘Unknown’.
Information Transfer Capability	Include ‘Speech’ and ‘3.1 kHz audio’
Send Dial Tone	Enable and Disable

4.6.2 PRI Trunk

On the PRI Trunk interface, you can configure PRI trunks for PRI calls. The statuses of PRI Trunks can be seen at the **Status & Statistics → PSTN Trunk Status** interface.

Click the **Add** button, and you can add a PRI trunk. If you want to delete or modify the information of a PRI trunk, select the checkbox on the left of the trunk, and then click the **Delete** button or the **Modify** button.

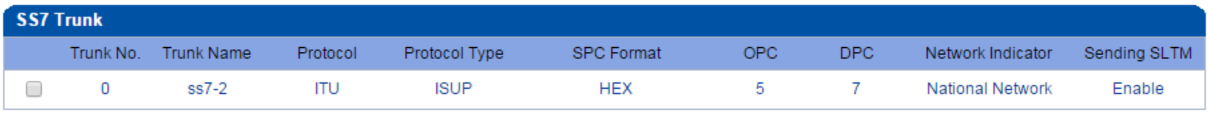


Parameter	Explanation
Trunk No.	Trunk No. starts from 0 to 19, it means you can establish 20 PRI trunks at most. The trunk No. is decided by the No. of the E1/T1 port linked to the trunk. But if D-channel is not enabled for a trunk, the No. of the trunk must be the same with a trunk under which D-channel has

	been enabled.
Trunk Name	The trunk name is used to distinguish the trunk from other trunks.
Channel ID	The ID of the channel selected for the PRI trunk. The channel ID is used for the switch to identify a PRI trunk in case that the Trunk No. of two trunks are the same.
D-Channel (Delta Channel)	The channel used to carry control information and signaling information
E1/T1 Port No.	The No. of E1/T1 port linked to the PRI trunk
Protocol	Support two protocols: ISDN and QSIG. Default value is ISDN.
Switch Side	The EI/T1 port of the PRI trunk is taken as User Side or Network Side.
Alerting Indication	Include Alerting and Progress Alerting: Play ring-back tone when receiving alerting signal Progress: Play ring-back tone when receiving progress signal

4.7 SS7 Config

On the **SS7 Config → SS7 Trunk** interface, you can configure SS7 trunks for SS7 calls. The statuses of SS7 Trunks can be seen at the **Status & Statistics → PSTN Trunk Status** interface.



SS7 Trunk

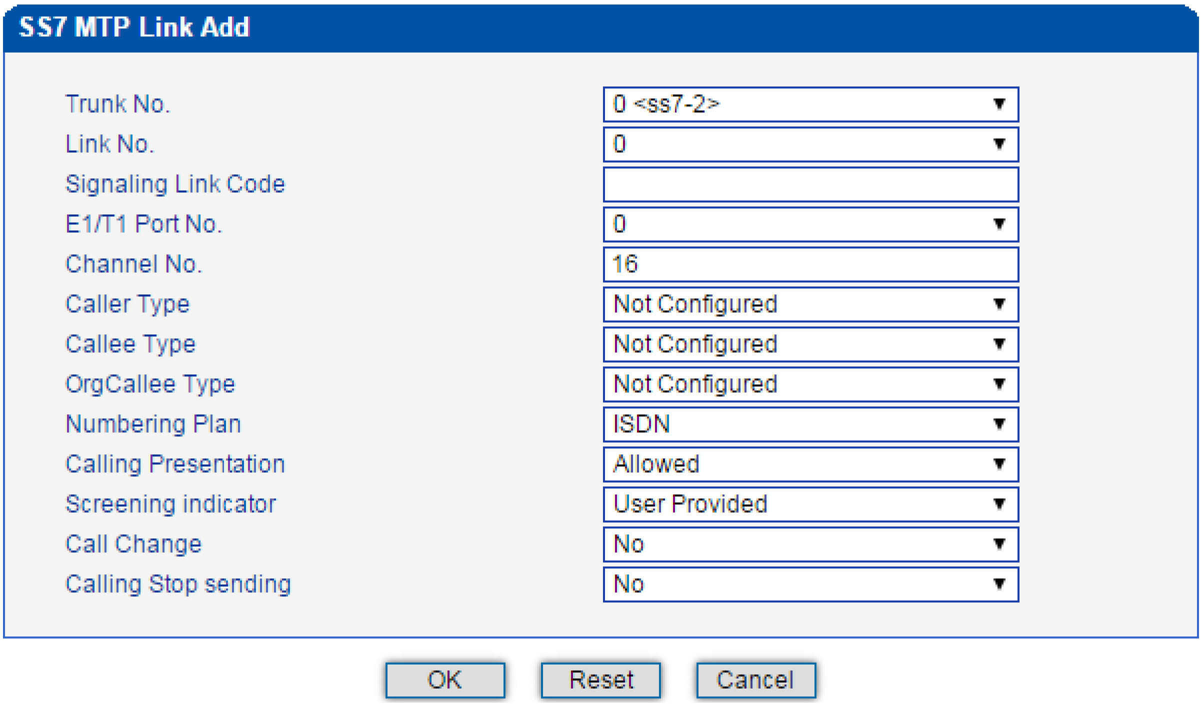
	Trunk No.	Trunk Name	Protocol	Protocol Type	SPC Format	OPC	DPC	Network Indicator	Sending SLTM
☐	0	ss7-2	ITU	ISUP	HEX	5	7	National Network	Enable

Parameter	Explanation
Trunk No.	The No. of the SS7 trunk. Generally, one SS7 trunk is for one DPC.
Trunk Name	The trunk name is used to distinguish the trunk from other trunks.
Protocol	SPC types: ITU-T (14 bit), ANSI (24 bit), ITU-CHINA (24 bit) SPC: Signaling Point Code
Protocol Type	ISUP (ISDN User Part) and TUP (Telephone User Part)
SPC Format	SPC: Signaling Point Code SPC format includes Hex (Hexadecimal system) and ITU point code structure (decimal system)
OPC	OPC: Original Point Code The signaling point code of MTG2000, which is generally assigned by telecom operators.
DPC	DPC: Destination Point Code

	The signaling point code of the peer device, which is generally assigned by telecom operators.
Network Indicator	Include International Network, International Spare, National Network and National Spare. Default value is National Network, which is mainly used in China, America and Japan.
Sending SLTM	Whether to send signaling link test message.

4.7.1 SS7MTP Link

On the **SS7 Config → SS7 MTP Link** interface, click the **Add** button, and you will see the following interface. On the interface, you can select an E1/T1 port for an existing trunk and establish two links between them.



Parameter	Explanation
Trunk No.	The No. of the SS7 trunk
Link No.	Each SS7 trunk supports two links which share the loading equally. If one link malfunctions, the other link will automatically bear all the loading until the faulty link is restored.
Signaling Link Code	If the Link No. of the trunk cannot match with that of the peer device, the SS7 trunk will be linked to the peer device according to signaling link code.
E1/T1 Port No.	The No. of E1/T1 port linked to the SS7 trunk
Channel No.	The No. of the channel under which signal is transmitted. Default value is 16.

Caller Type	The type of the caller number. Options include 'Not Configured', 'Subscriber', 'International" and "National'.
Callee Type	The type of the called number. Options include 'Not Configured', 'Subscriber', 'International" and 'National'.
OrgCallee Type	The type of the original called number in case of number manipulation. Options include 'Not Configured', 'Subscriber', 'International' and 'National'.
Numbering Plan	Options include 'ISDN', 'Data', 'Telex' and 'Private'.
Calling Presentation	If 'Allowed' is selected, the calling number will be presented. If 'Restricted' is selected, the calling number will not be presented. If 'Not Config' is selected, the parameter does not work.
Screening Indicator	Options include "User Provided" and "Network Provided".
Calling Stop Sending	'Stop Sending' is an end mark. If 'Yes' is selected for 'Calling Stop Sending', it means there will be an end mark following the calling number.

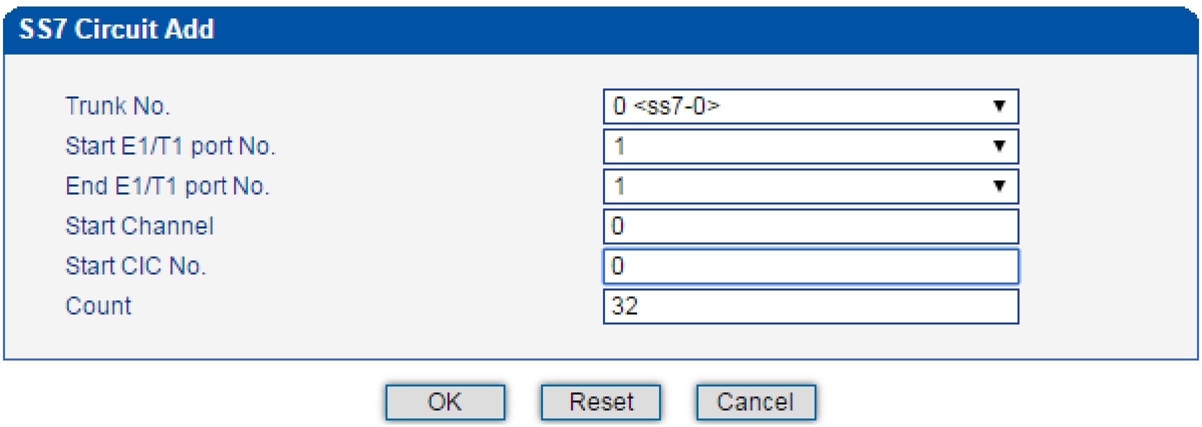
4.7.2 SS7 CIC

On the **SS7 Config → SS7 CIC** interface, click the **Add** button, and you will see the following interface. You can determine which channels will be used by an SS7 trunk on the interface.

➢ Procedures for adding SS7circuit that only involves an E1/T1 port:

Step 1: Click **Add** on the **SS7 CIC** interface.

Step 2: Select a trunk and an E1/T1 port. (Trunk 0 and Port 1 are taken as example in the following figure

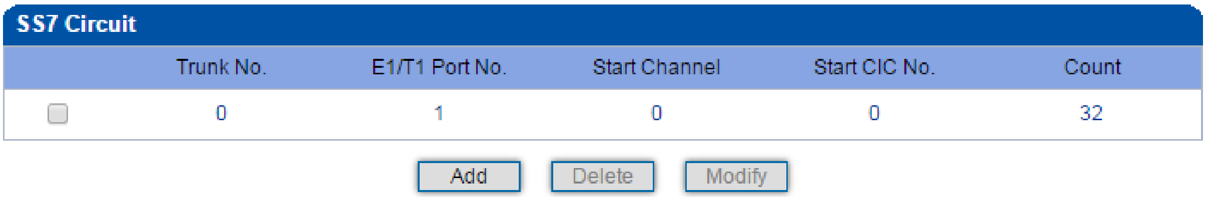


Note:

As there are 32 channels (from 0 to 31) for one E1/T1 port, so the value for **Count** is 32. When start E1/T1 port is the same with end E1/T1 port, it means only one E1/T1 port is connected to the SS7 trunk.

Parameter	Explanation
Trunk No.	The No. of the SS7 trunk
Start E1/T1 Port No.	The No. of the start E1/T1 port
End E1/T1 Port No.	The No. of the end E1/T1 port
Start Channel	When the start E1/T1 port is also the end E1/T1 port, it's required to set the start channel, and the channels starting from the set channel to the No.31 channel of the E1/T1 port will be used by the SS7 trunk.
Start CIC No.	CIC: Circuit Identification Code The CIC No. of the start channel, which is generally 0, 32, 64, 96, 128, 160, 192, 224, 256, 288, 320, 352, 384, 416, 448…
Count	The total number of the channels used by the SS7 trunk

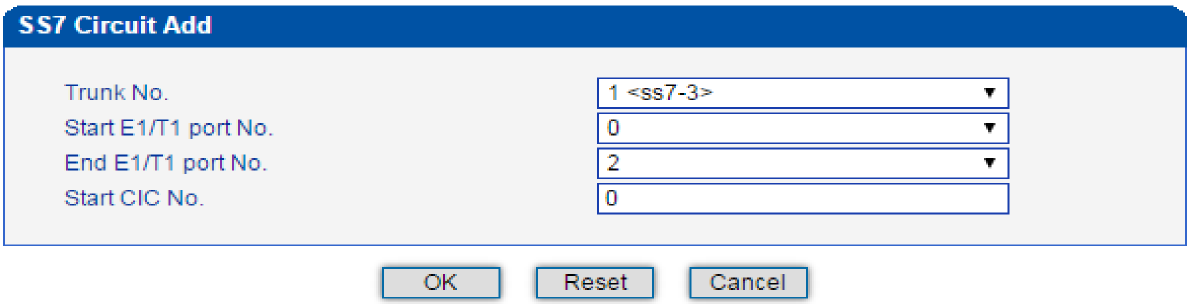
Step3: Click **OK**. And then you can see the following data on the **SS7 CIC** interface.



➢ Procedures for adding SS7circuit that involves multiple E1/T1 ports:

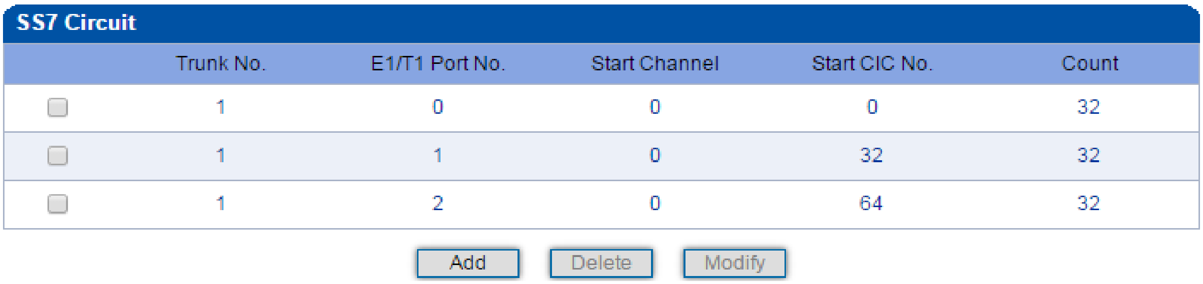
Step 1: Click **Add** on the **SS7 CIC** interface.

Step 2: Select a trunk and E1/T1 ports. (Trunk 1, Port 0, Port1 and Port 2 are taken as example in the following figure.



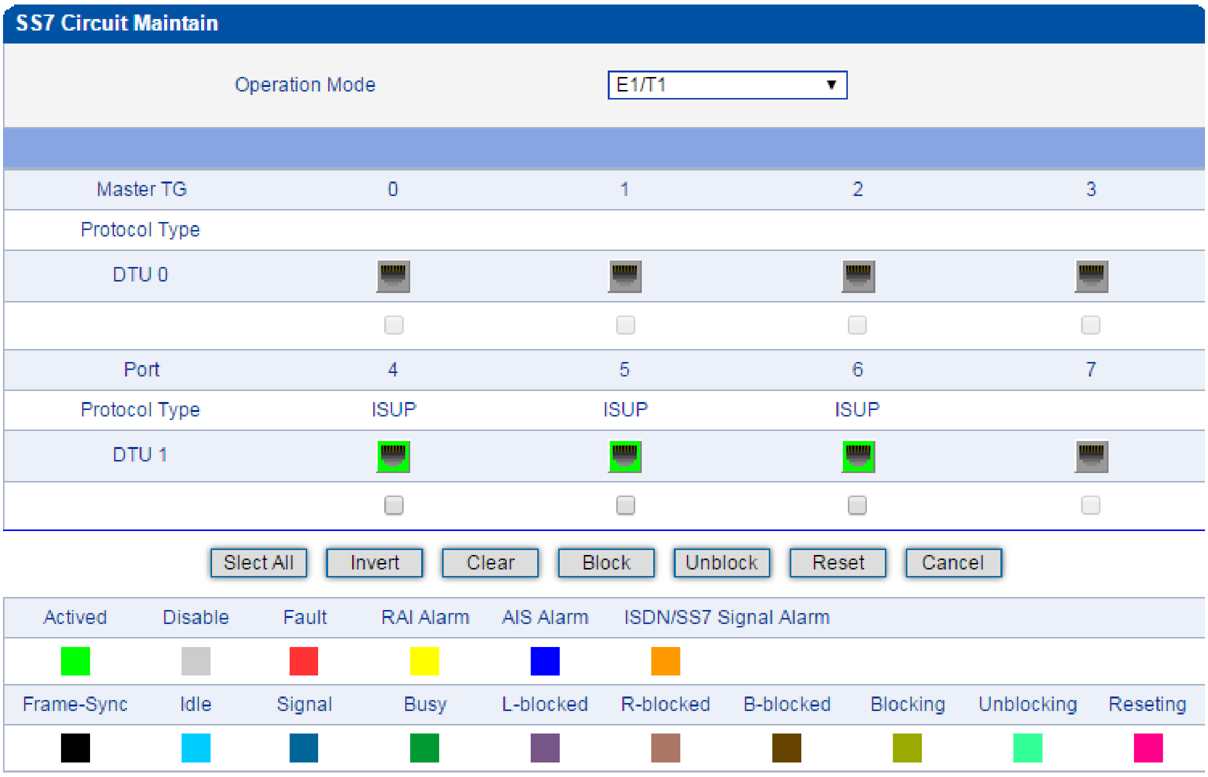
Note: If multiple E1/T1 ports are involved, it defaults that all the 32 channels of each E1/T1 port involved are used by the SS7 trunk.

Step3: Click **OK**. And then you can see the following data on the **SS7 CIC** interface.



4.7.3 SS7 CIC Maintain

There are two objects to be maintained for SS7 CIC, namely E1/T1 ports and channels. Select **E1/T1** on the right of **Operation Mode**, and the following interface will be displayed.

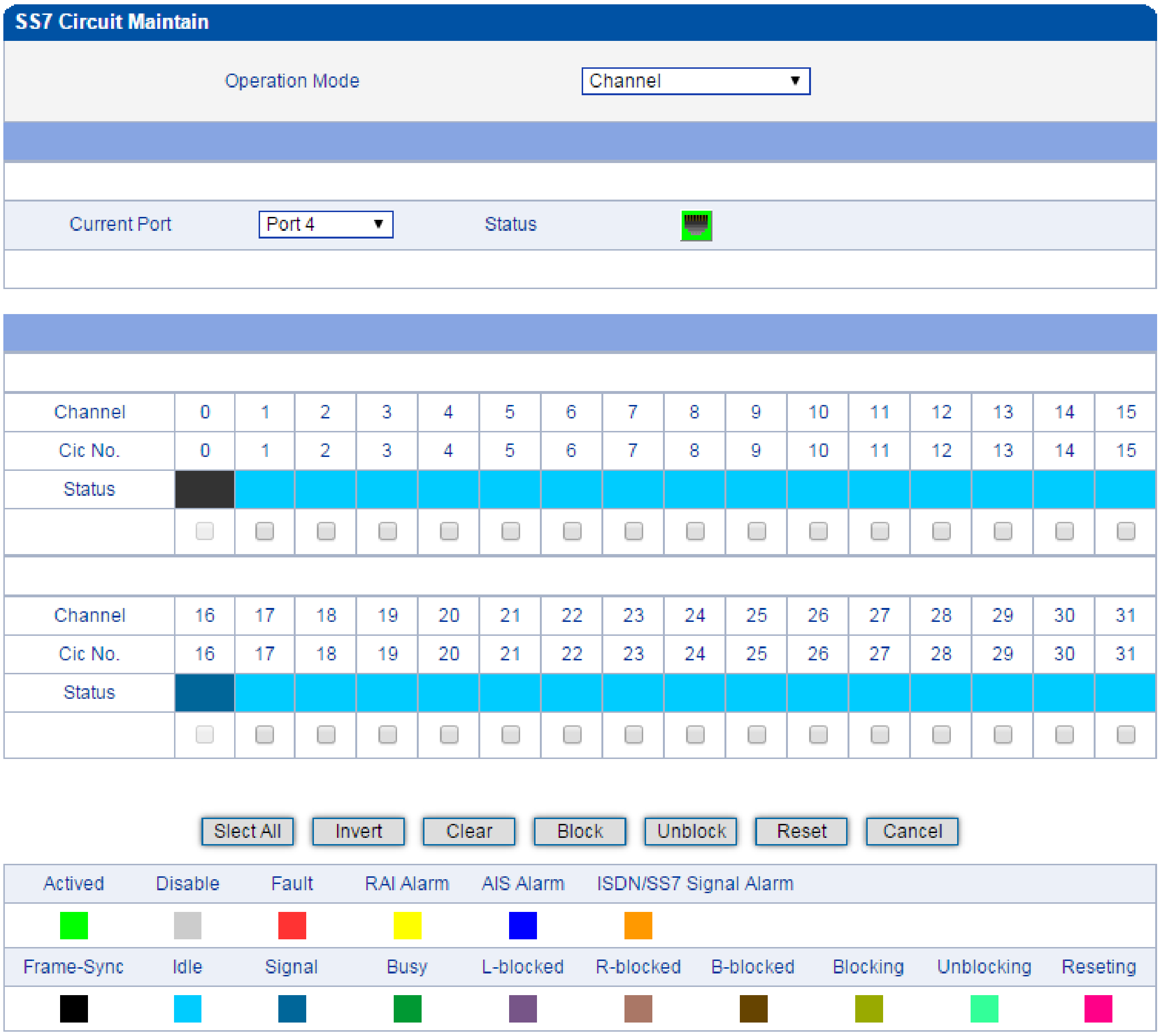


Parameters	**Explanation**
Operation Mode	E1/T1
Port	The No. of E1/T1 port
Protocol Type	ISUP or TUP
DTU	The No. of DTU which the E1/T1 ports belong to
Status	The E1/T1 ports have 16 statuses, including Activated, Disabled, Fault, RAI Alarm, AIS Alarm, ISDN/SS7 Signal Alarm, Frame-Sync, Idle, Signal, Busy, L-block, R-blocked, B-blocked, Blocking, Unblocking and Resetting.

	The meaning of each status, please make reference to 4.4.2.

Meanwhile, you can carry out maintenance on the E1/T1 ports through the following buttons: **Select All**, **Invert**, **Clear**, **Block**, **Unblock**, **Reset** and **Cancel**.

Select **Channel** on the right of **Operation Mode**, and then select an E1/T1 port, the channels of the E1/T1 port and their statuses are displayed.



Parameter	**Explanation**
Operation Mode	Channel
Current Port	The No. of the current E1/T1 port
Channel	The No. of channels
CIC No.	The CIC No. of channels
Status	The statuses of channels, including Activated, Disabled, Fault, RAI Alarm,

	AIS Alarm, ISDN/SS7 Signal Alarm, Frame-Sync, Idle, Signal, Busy, L-block, R-blocked, B-blocked, Blocking, Unblocking and Resetting.

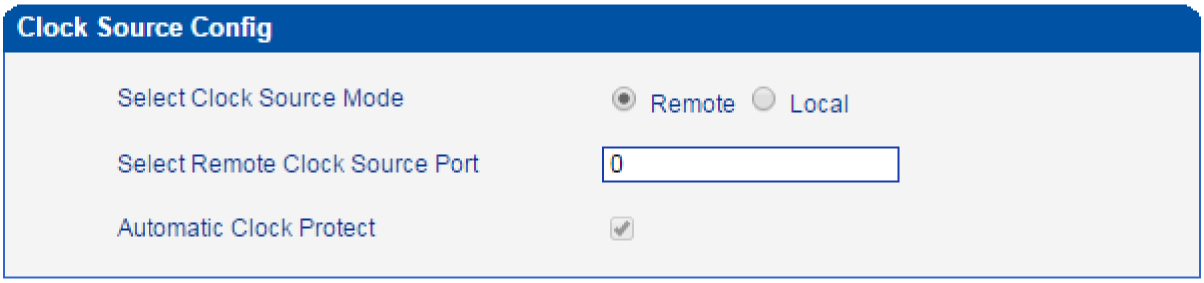
Meanwhile, you can carry out maintenance on the channels of E1/T1 ports through the following buttons: **Select All**, **Invert**, **Clear**, **Block**, **Unblock**, **Reset** and **Cancel**.

4.8 PSTN Group Config

In this section, you can group several PRI trunks or SS7 trunks together, so when one trunk is in an outage, communication can turn to another trunk in the same group.

4.8.1 Clock Source

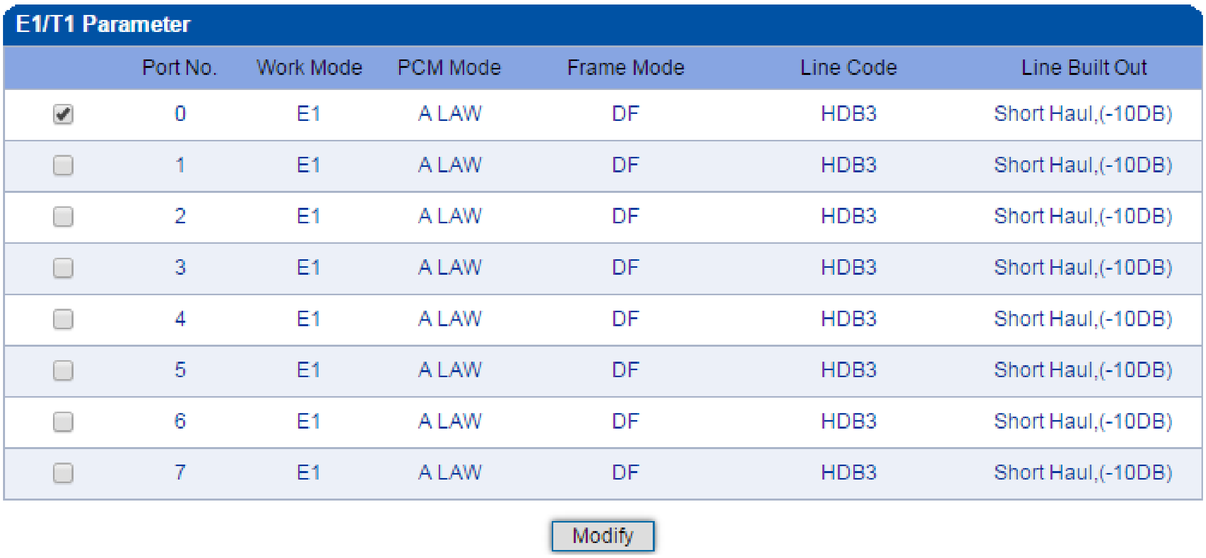
When clock source is produced by the local crystal chip of MTG2000, it is regarded as local clock source. When clock source is obtained from the data received by E1/T1 ports, it is regarded as remote clock source. Each E1/T1 port can obtain one clock source.



Parameter	Explanation
Select Clock Source Mode	If Remote is selected, clock source is produced by crystal chip; if local is selected, clock source is obtained from the data received by E1/T1 port.
Select Remote Clock Source Port	The No. of the E1/T1 port from which clock source is obtained.
Automatic Clock Protect	Clock source is protected automatically.

4.8.2 E1/T1 Parameter

Select the checkbox on the left of an E1/T1 port, and click the **Modify** button to modify E1/T1 parameters.

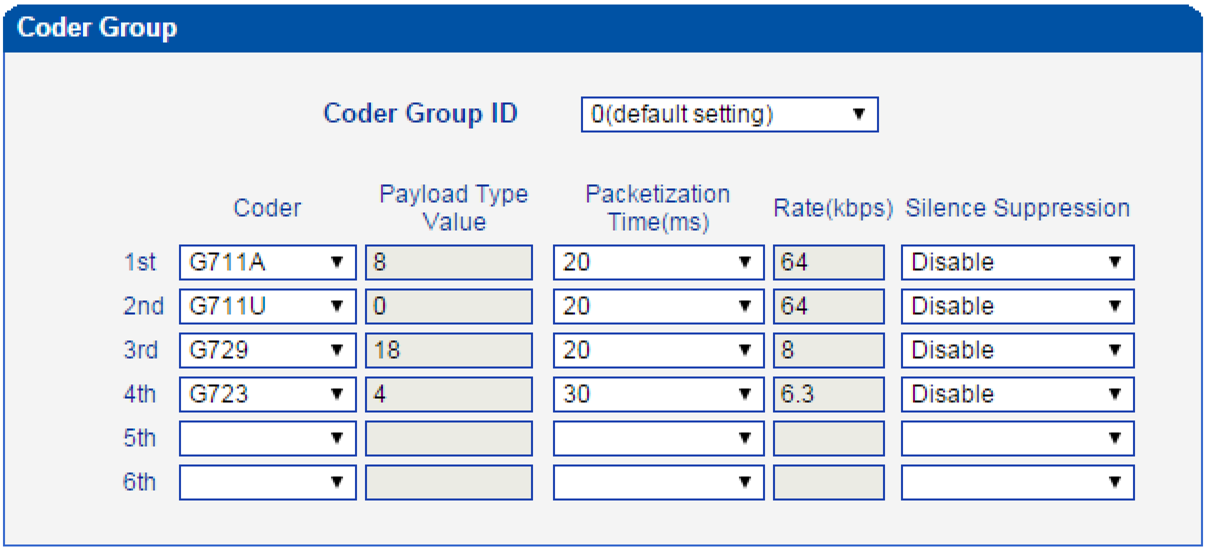


E1/T1 Parameter						
	Port No.	Work Mode	PCM Mode	Frame Mode	Line Code	Line Built Out
☑	0	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	1	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	2	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	3	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	4	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	5	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	6	E1	A LAW	DF	HDB3	Short Haul,(-10DB)
☐	7	E1	A LAW	DF	HDB3	Short Haul,(-10DB)

Parameter	Explanation
Port No.	The No. of each E1/T1 port
Work Mode	E1 or T1 If E1 is selected for one port, the work modes of all ports are E1.
PCM Mode	PCMA(A LAW) or PCMU(Mu LAW) If A LAW is selected for one port, the work modes of all ports are A LAW.
DF CRC-4 CRC4_ITU	Frame formats of E1 port include DF, CRC-4, CRC4_ITU, and the default value is CRC-4; Frame formats of T1 port include F12, F4, ESF, F72, and the default value is F4.
Line Code	Line codes of E1 include NRZ, CMI, AMI, HDB3, and the default value is HDB3; Line codes of T1 include NRZ, CMI, AMI, B8ZS, and the default value is B8ZS.
Line Built-out	Short Haul (-10DB)
Batch Configure	If Disable is selected, E1/T1 parameter cannot be configured at batch; If Enable selected, E1/T1 parameter can be configured at batch;

4.8.3 Codec Group

On the **Codec Group** interface, you can group several voice codecs together, so when one voice codec is faulty, another voice codec in the same group can be used. Except codec group 0, the parameters of other codec groups can be modified.

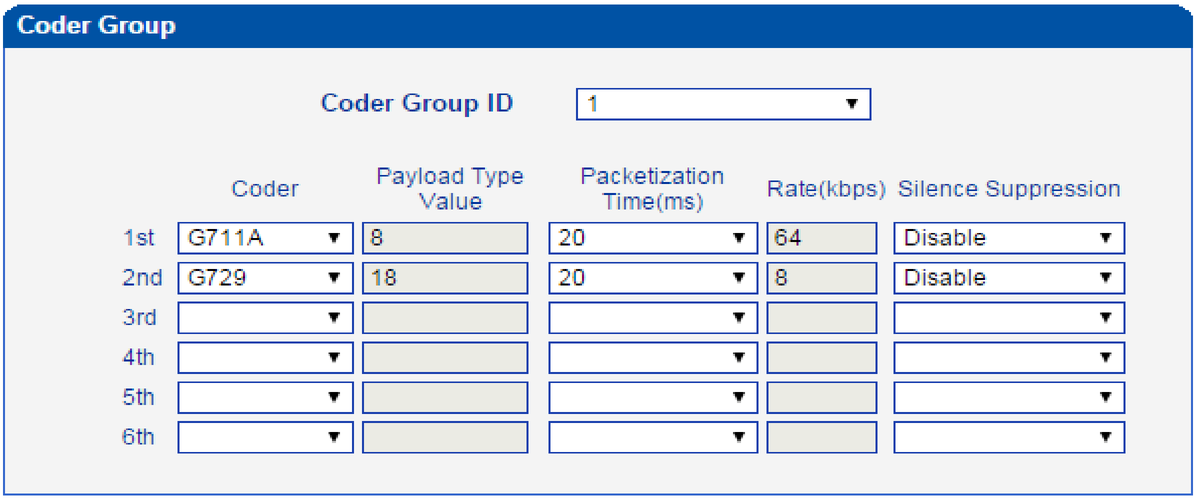


Parameter	Explanation
Codec Group ID	ID of each codec group for voice ability, from 0 to 7. The codec group 0 is default setting which cannot be modified.
Codec	MTG2000 supports three kinds of voice codec: G711A, G711U, G729, G723, iLBC 13k and iLBC 15k.
Payload Type Value	Each codec has a unique payload type value (make reference to RFC3551).
Packetization Time (ms)	The minimum packetization time of voice codec. For example, if packetization time is 20ms, voice will be packetized every 30ms.
Rate (kbps)	Transmission rate of voice
Silence Suppression	If silence suppression is enabled, the bandwidth occupied by voice transmission will be released automatically for the silence party or when talking is paused. Default value is 'Disable'.

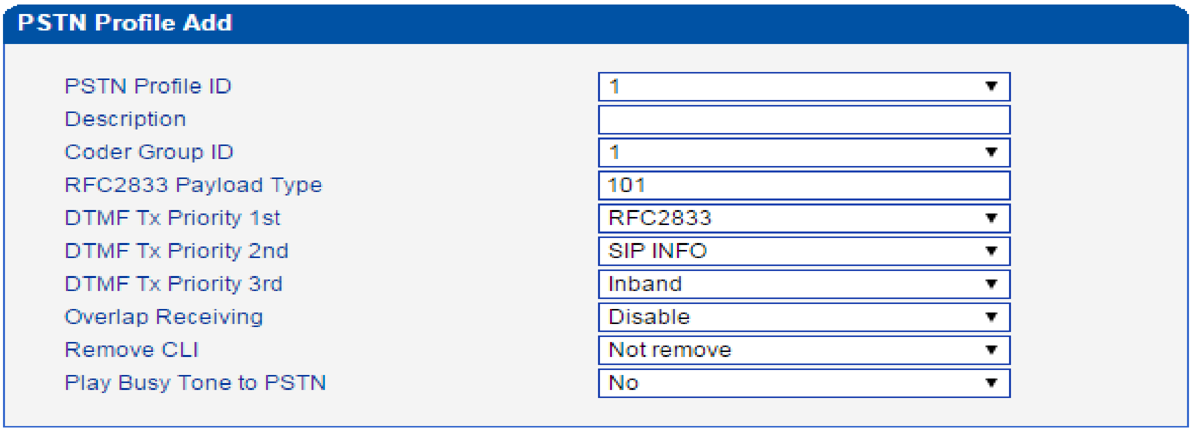
- **Example: How to configure preferred codec group**

Step1. Enter into the Codec Group interface and select codec group ID 1 to create new codec group

Step2. Select preferred voice codec (G711A and G729) in this example, as below:

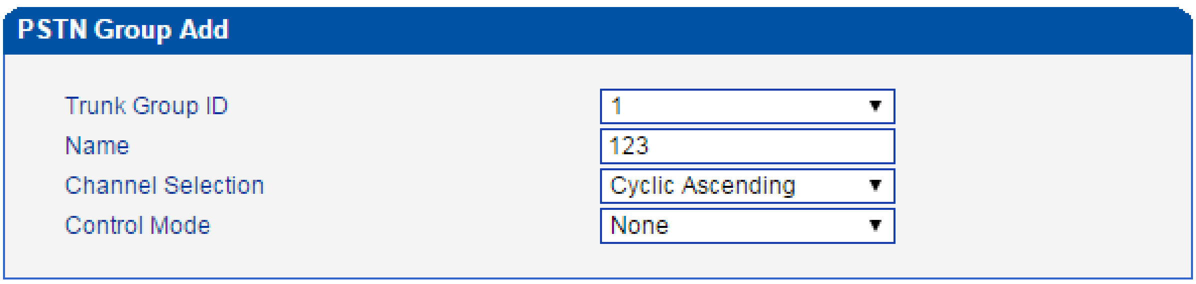


Step3. Enter into the **PSTN Profile** interface, click **Modify** to modify the default PSTN profile and change the codec group ID, or click **Add** to add a new PSTN profile.

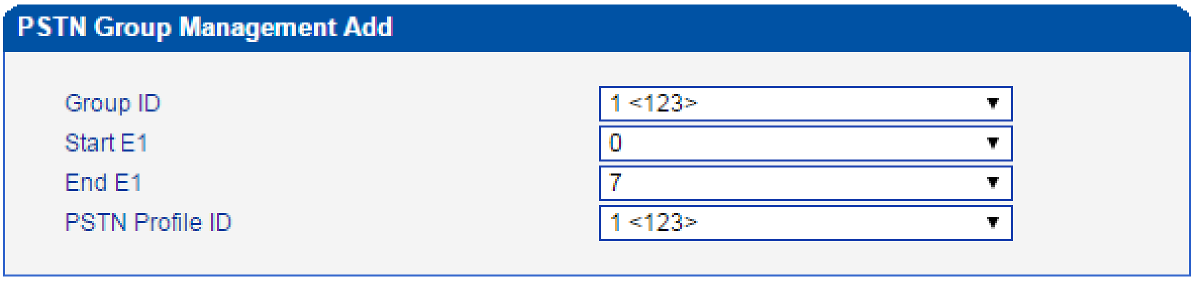


Step4. Click **OK** to save the above configuration.

Step5. Enter into the **PSTN Group** interface to establish a PSTN group



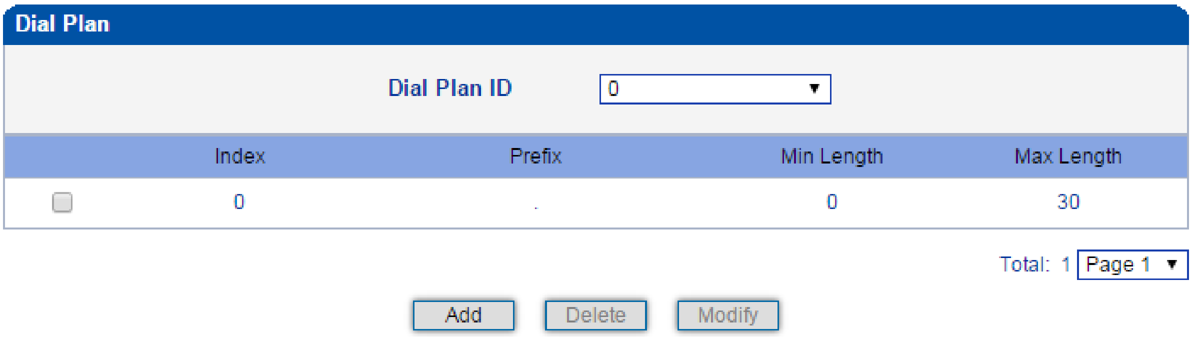
Step5. Enter into the **PSTN Group Management** interface to associate the PSTN profile and PSTN group to an E1/T1 port or multiple E1/T1 ports.



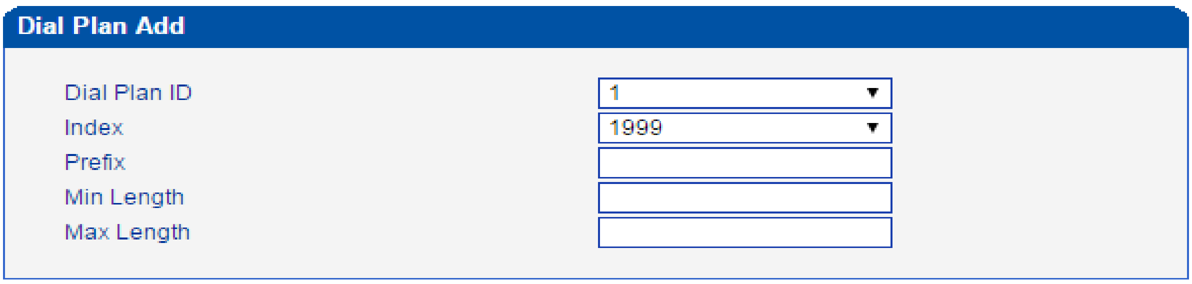
Step6. Click **OK** save the above configuration.

4.8.4 Dial Plan

Dial plan is used for the MTG2000 to identify how many digits that a received number includes. Dial rules can be divided into 5 groups with dial plan IDs. The setting in dial plan 0 is the default setting.



Click the **Add** button, and you can add a new dial plan in the following interface.



Parameter	Explanation
Dial Plan ID	The ID of the dial plan
Index	Each dial plan has a unique index. Greater index value, higher priority for the dial plan.

Prefix	The prefix matching received numbers, through which the MTG2000 can judge how many digits the received number includes.
Min Length	The minimum number of digits included in a telephone number (generally from 0 to 30). If the length of a received number falls within the range of between the set minimum length and the set maximum length, call connection will continue.
Max Length	The maximum number of digits included in a telephone number (generally from 0 to 30). If the length of a received number reaches the set maximum length, MTG2000 deems that all digits of the number have been received and will begin to analyze the telephone number, and if there are still digits being sent, MTG2000 will not received them.

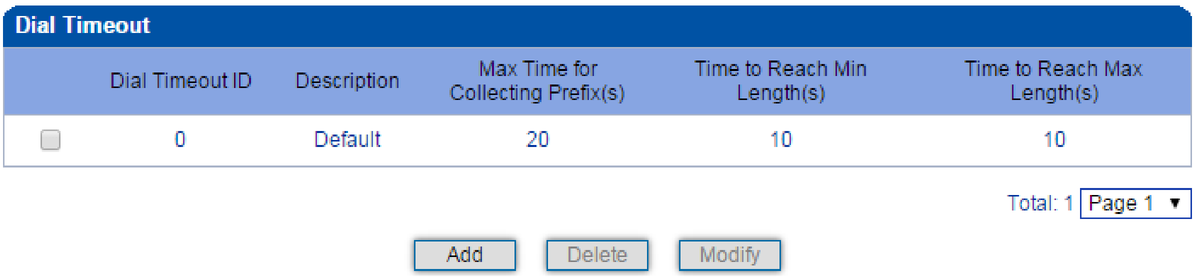
Note:

1. Dial plans can be backed up and restored at the **Maintenance → Data Backup** interface and the **Maintenance → Data Restore** interface respectively.
2. ‘Min Length’ and ‘Max Length’ does not include the length of prefix.
3. For overlapping dialing, it’d better to set ‘Min Length’ and ‘Max Length’ to a same value in order to accelerate connection rate, since the length of the called number has been known.

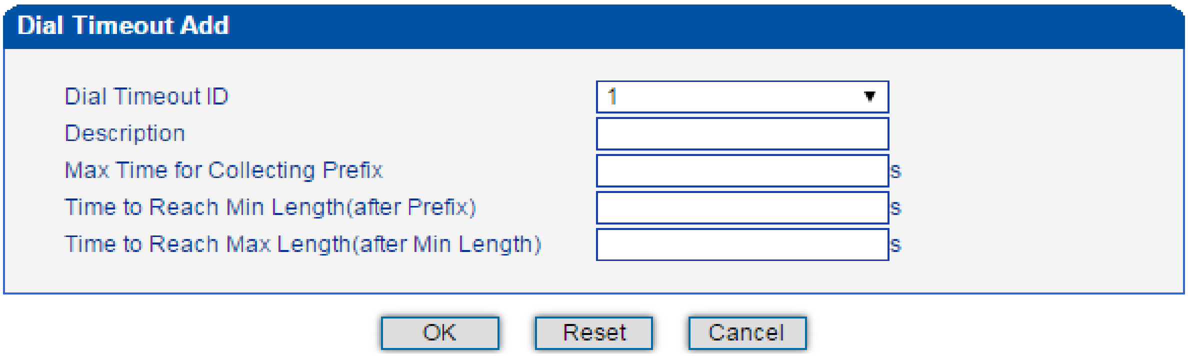
4.8.5 Dial Timeout

On the **Dial Timeout** interface, you can set the maximum time for collecting prefix and the maximum time for telephone number to reach ‘Min Length’ and ‘Max Length’.

The setting in Dial Timeout 0 is default setting, which can be modified but cannot be deleted.



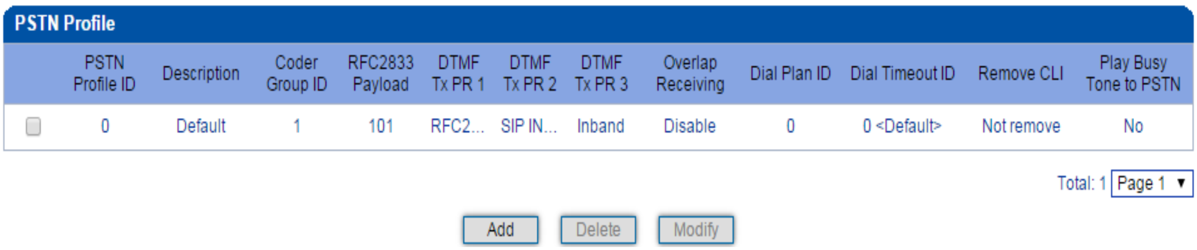
Click the **Add** button to add a new dial timeout rule.



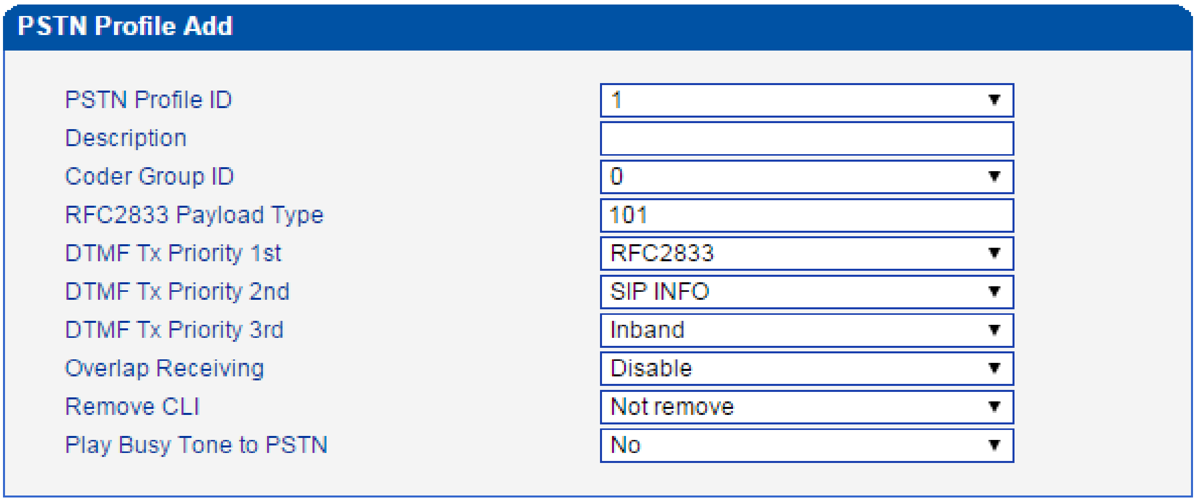
Parameter	Explanation
Dial Timeout ID	The ID of the dial timeout
Description	Description of the dial timeout
Max Time for Collecting Prefix	The maximum time for receiving all the digits of a prefix
Time to Reach Min Length (after Prefix)	After receiving the prefix, the maximum time before receiving the set minimum number of digits included in a telephone number.
Time to Reach Max Length (after Min Length)	After receiving the set minimum number of digits, the maximum time before receiving the set maximum number of digits included in a telephone number.

4.8.6 PSTN Profile

On the **PSTN Profile** interface, you can configure PSTN call number rules and related parameters, such as associating a codec group, a dial plan and a dial timeout to a PSTN profile.



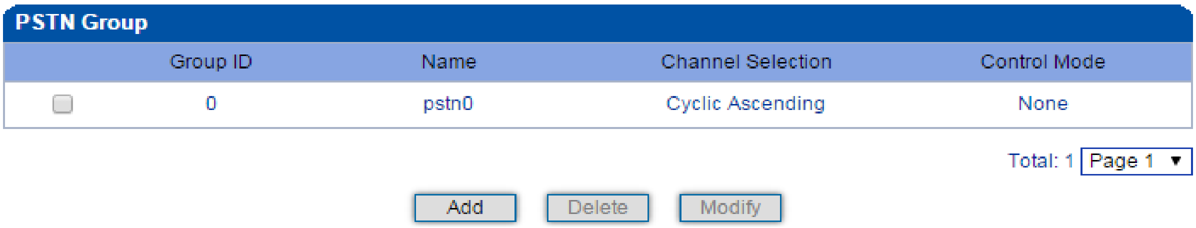
Click the **Add** button to add a new PSTN profile.



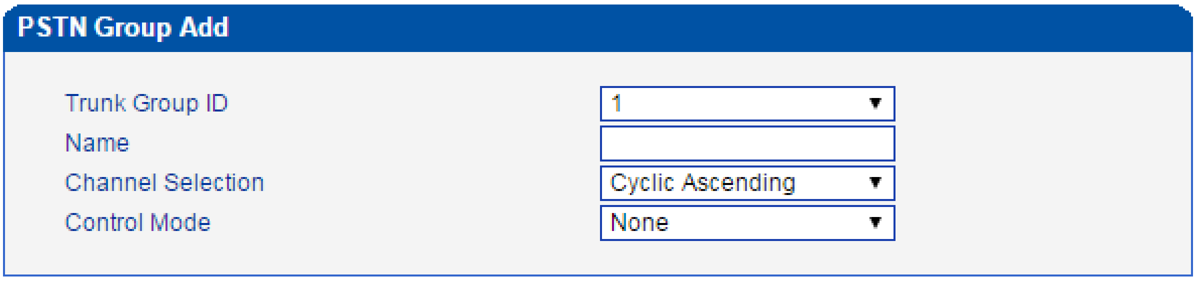
Parameter	Explanation
PSTN Profile ID	The ID of the PSTN profile
Description	The description of the PSTN profile
Coder Group ID	The ID of the coder group (the coder group needs to be created at the **Coder Group** interface first.)
RFC2833 Payload	Default value is 101.
DTMF Tx Priority 1st	There are three ways to send DTMF: RFC2833, SIP INFOR and Inband. You can set their priority. Priority 1^{st} represents the top priority.
DTMF Tx Priority 2nd	There are three ways to send DTMF: RFC2833, SIP INFOR and Inband. You can set their priority. Priority 2^{st} represents the second priority.
DTMF Tx Priority 3rd	There are three ways to send DTMF: RFC2833, SIP INFOR and Inband. You can set their priority. Priority 2^{st} represents the third priority.
Overlap Receiving	Default value is ‘Disable’; If overlap receiving is enabled, the set ‘Dial Plan’ and ‘Dial Timeout’ will work.
Remove CLI	CLI: Calling Line Identification Whether to remove CLI
Play busy tone to PSTN	If ‘Yes’ is selected, when the called phone is offhook, MTG2000 will play busy tone to the PSTN side.

4.8.7 PSTN Group

On the **PSTN Group** interface, you can create a PSTN group and set a strategy for channel selection of the group.



Click the **Add** button to add a new PSTN group.

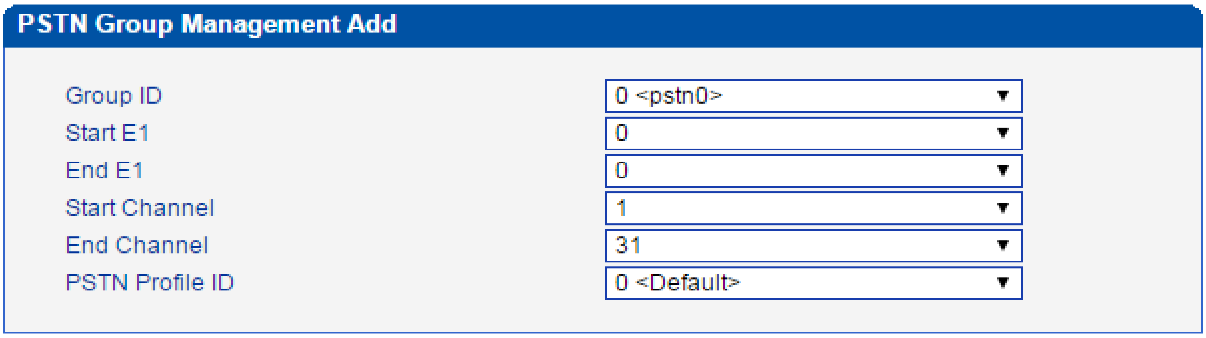


Parameter	Explanation
Trunk Group ID	The ID of the trunk group
Name	The name of the trunk group
Channel Selection	There are four selection strategies: Ascending, Descending, Cyclic Ascending and Cyclic Descending. Ascending: to search idle channels starting from channel 0 to channel 31; Cyclic ascending: to search idle channel in an ascending order, starting from the previous idle channel that has been selected
Control Mode	Control mode is also a method for channel selection and works together with the set selection strategy. Options include Master Odd, Master Even and None. Master Odd: it means channels with odd ID will be searched first, and channels with even ID will not be searched until all channels with odd ID have been searched.

4.8.8 PSTN Group Management

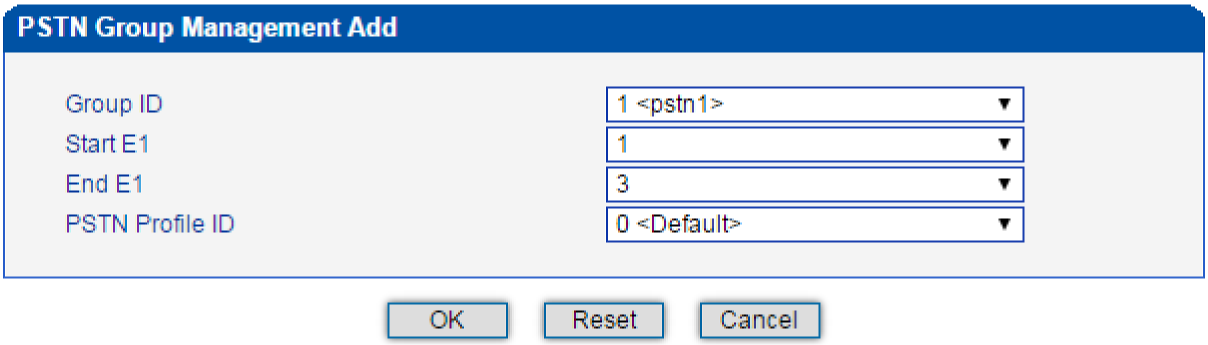
On the **PSTN Group Management** interface, you can add start E1/T1 port, end E1 /T1 port, start channel, end channel and PSTN profile to a PSTN group.

Click the **Add** button, and you will see the following configuration interface.



In the above figure, as start E1 is the same with end E1, only one E1 port is used in the PSTN group and you need to set start channel and end channel.

When there is a need to set several E1 ports, it defaults that all the 32 channels of each E1 port are used by the PSTN group.

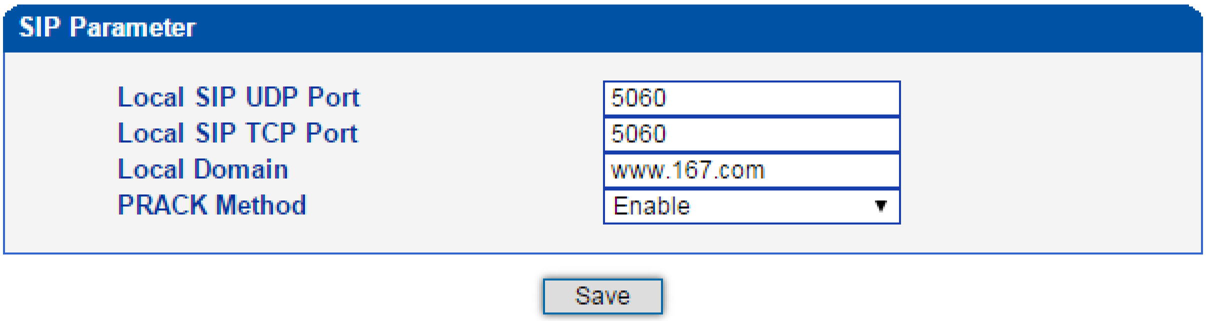


Parameter	**Explanation**
Group ID	The ID of the PSTN group
Start E1/T1	The start E1/T1 port in this PSTN group
End E1/T1	The end E1/T1 port in this PSTN group
Start Channel	The start channel in this PSTN group
End Channel	The end channel in this PSTN group
PSTN Profile ID	The ID of the PSTN profile in this PSTN group (the PSTN profile needs to be created at the **PSTN Profile** interface first.

Note: When the start E1/T1 port is different from the end E1/T1 port, the start channel is channel 0 by default and the end channel is channel 31 by default (it means there is no need to choose a start channel and a end channel).

4.9 SIP Config

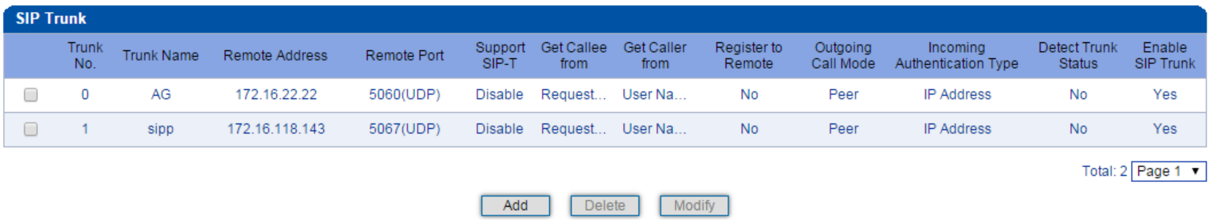
4.9.1 SIP Parameter



Parameter	Explanation
Local SIP UDP Port	5060 (default)
Local SIP TCP Port	5060 (default)
Local Domain	A local domain whose format is www.xxx.com
PRACK Method	PRACK: Provisional Response ACKnowledgement

4.9.2 SIP Trunk

SIP trunk can realize the connection between MTG2000 and PBX or SIP servers under the IP network. It provides two modes to connect MTG2000 and the IP network. One is Access (MTG2000 registers to a softswitch), and the other is Peer (MTG2000 connects to a peer device in the IP network via IP address).

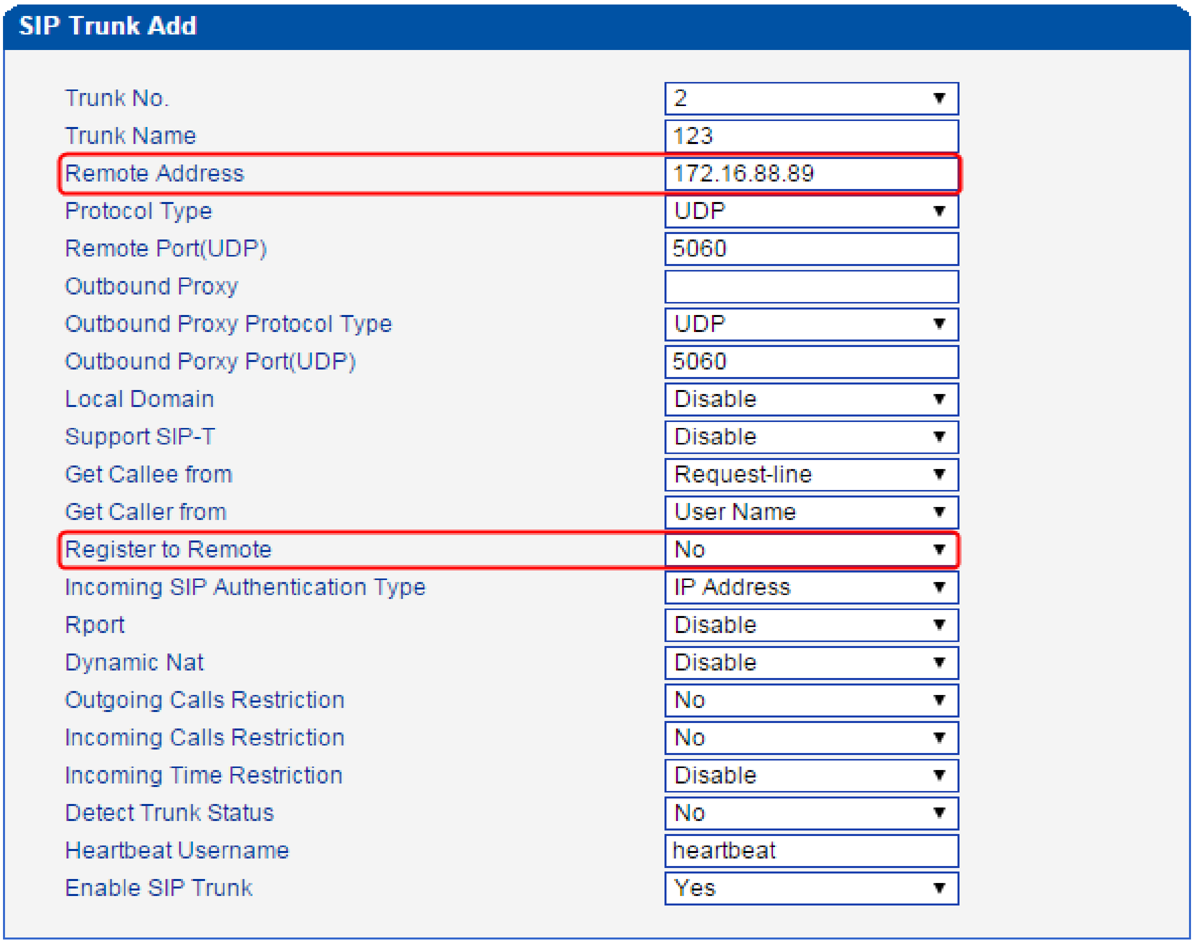


Configuration procedures for Peer Mode are as follows:

1. Click the **Add** button to add a SIP trunk.

2. Configure parameters on the **SIP Trunk Add** interface according to related explanations in the table. As it is Peer mode, you should select **No** for the **Register to Remote** parameter, and enter the IP address of the peer device.

3. After finishing the configuration of the parameters, click **OK**.



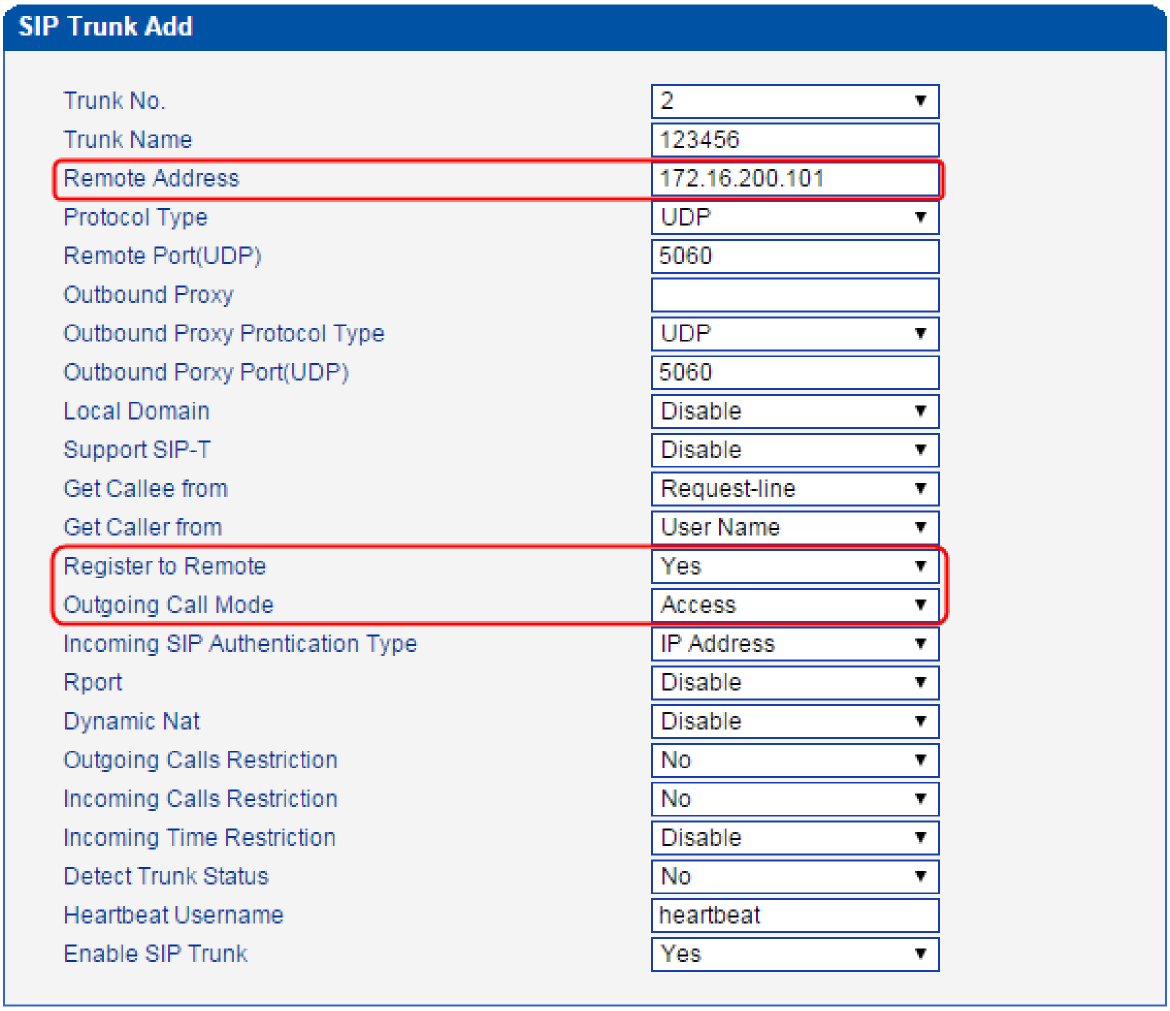
Parameter	**Explanation**
Trunk No.	The No. of the SIP trunk (range is 1 ~99)
Trunk Name	The name of the SIP trunk
Remote Address	The IP address of the peer device interfacing with the MTG2000
Protocol Type	Options include UDP, TCP and Auto If Auto is selected, the protocol type is determined by the peer device.
Remote Port (UDP)	The SIP port of the peer device interfacing with the MTG2000; The default remote port is 5060.
Outbound Proxy IP address	SIP proxy IP address If outbound proxy is used, enter the IP address or domain name of the proxy server
Outbound Proxy Protocol Type	Options include UDP, TCP and Auto If Auto is selected, the protocol type is determined by the peer device.
Outbound Proxy Port (UDP)	The default outbound proxy port is 5060.
Local Domain	The local domain set in the **SIP Parameter** interface

Support SIP-T	This parameter is for SS7. Its default value is 'Disable'.
Get Callee from	Get the called number from 'Request-line' or 'To Header Field'
Get Caller from	Get the caller number from 'User Name' or 'Display Name'
Register to Remote	It is defined by IETF RFC3372, which is a standard used to establish remote communication between SIP and ISUP; The default value is 'Yes'. If 'Yes' is selected, MTG2000 will be registered to the peer device whose IP address is filled in 'Remote Address'.
Incoming SIP Authentication Type	Incoming calls from IP network can be authenticated by IP address or password. If password is selected, you need fill in password. If IP address is selected, incoming calls will be rejected when their IP address are different from the remote address filled in.
Rport	Whether to enable the Rport of the SIP trunk
Dynamic Nat	Enable or Disable If it is enabled, a private IP address can be mapped to a public address from a pool of public IP addresses.
Outgoing Calls Registration	Whether to limit the number of the calls from PSTN to IP network. The default value is 'No'. If 'Yes' is selected, then input the number of concurrent calls that are allowed to go out. The range is 0 to 65535.
Incoming Calls Registration	Whether to limit the number of the calls from IP network to PSTN. The default value is 'No'. If 'Yes' is selected, then input the number of concurrent calls that are allowed to come in. The range is 0 to 65535.
Incoming Time Registration	The default setting is 'Disabled'. If 'Enabled' is selected, user can edit the start and stop time of a prohibition period. During this period, all calls from IP network to PSTN are prohibited. (Calls from PSTN to IP network are not limited)
Detect Trunk Status	Whether to detect the status of the SIP trunk. If 'Yes' is selected, MTG2000 will send Heartbeat message to the peer device to confirm whether the link status is OK.
Heartbeat Username	The name of the Heartbeat message
Enable SIP Trunk	Whether to enable the SIP trunk. If 'Yes' is selected, the SIP trunk is available; If 'No' is selected, the SIP Trunk is invalid.

Configuration procedures for Access Mode are as follows:

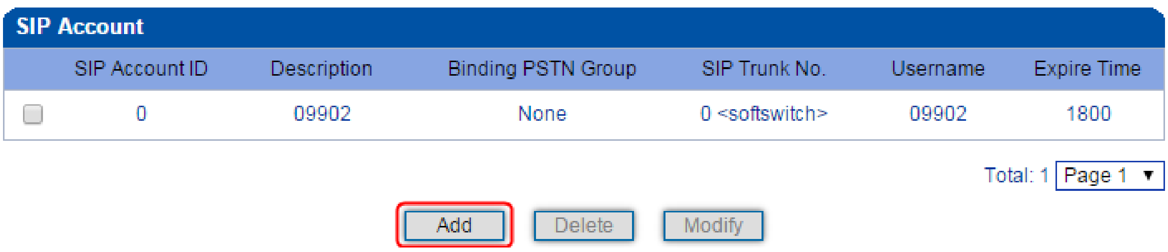
1. Click the **Add** button to add a SIP trunk.

2. Configure parameters on the following interface according to related explanations.
   As it is Access mode, you should select **Yes** for the **Register to Remote** parameter, and enter the IP address of a softswitch.

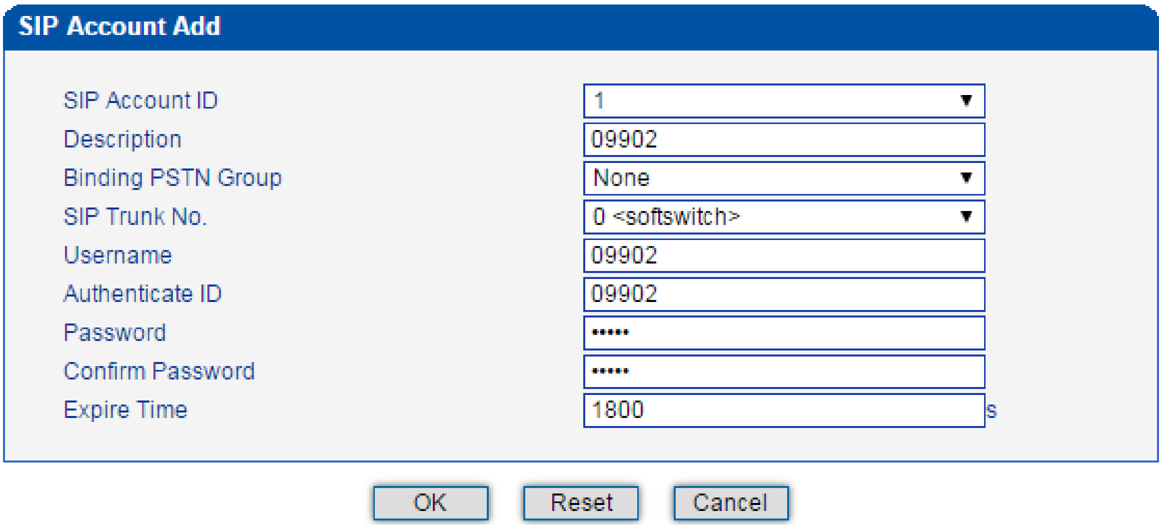


3. Click **OK.**

4. Click **SIP Account** in the navigation tree on the left, and then click **Add** to add a SIP account.



5. Configure the parameters on the **SIP Account Add** interface.



Parameter	Explanation
SIP Account ID	The ID of SIP Account, from 0 to 127
Description	Description of the SIP account
Binding PSTN Group	Choose a PSTN group that is bound to the SIP account
SIP Trunk No.	The No. of the SIP trunk bound to the SIP account
Username	The username of the SIP account, which is used to register the SIP account to softswitch
Authenticate ID	The authentication ID to authenticate the SIP account for the softswitch connected to MTG2000
Password	The password of SIP account, which is used when the SIP account is registered to softswitch
Confirm Password	Enter the password again
Expire Time	The interval to register the SIP account; Default value is 1800s.

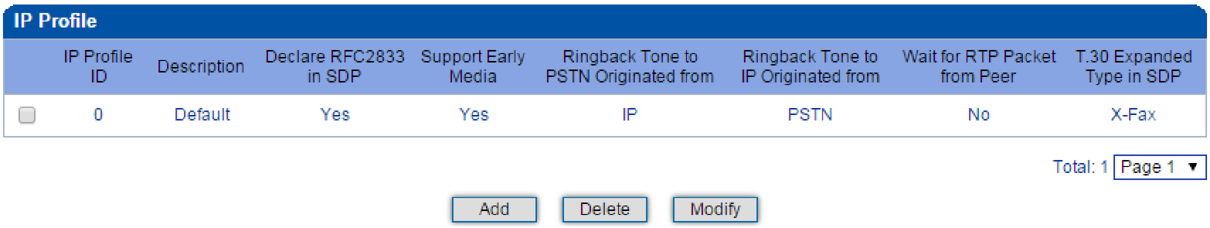
6. Click **OK**. And you can click **Status & Statistics → IP Trunk Status** to check the SIP trunk that has been established.

4.10 IP Group Config

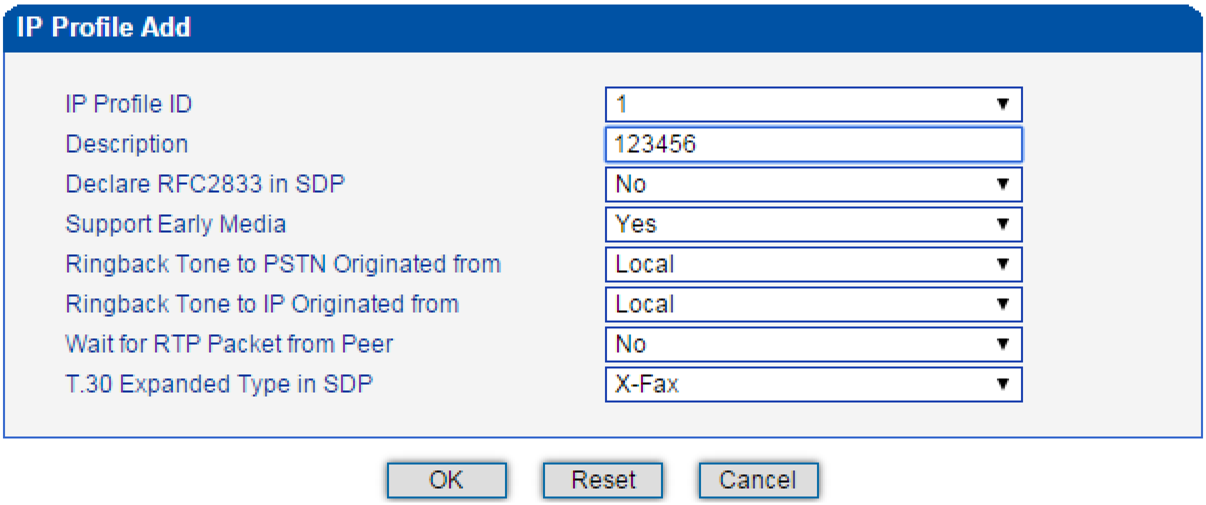
You can group several SIP trunks together, so when one SIP trunk is in an outage, communication can turn to another SIP trunk in the same group.

4.10.1 IP Profile

On the **IP Profile** interface, you can configure the parameters about IP calls, such as whether to support early media, where ringback tone to PSTN/IP is originated from and whether to wait for RTP packet from peer device.



Click **Add**, and the following interface will be displayed.

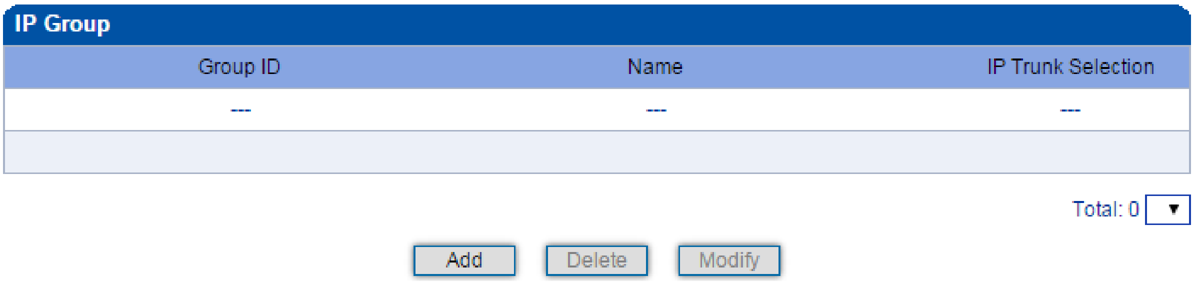


Parameter	**Explanation**
IP Profile ID	The ID of the IP profile, from 1 to 15.
Description	Description of the IP profile
Declare RFC2833 in SDP	Whether to declare RFC2833 in SDP Default value is 'Yes'.
Support Early Media	Whether to support Early Media (183) If 'Yes' is selected, ringback tone will be played to the caller before the call is successfully connected.
Ringback Tone to PSTN Originated from	Where the ringback tone to PSTN side is originated from If 'Local' is selected, the ringback tone is played from MTG2000. If 'IP' is selected, the ringback tone is played from the IP network
Ringback Tone to IP Originated from	Where the ringback tone to IP network l is originated from If 'Local' is selected, the ringback tone is played from MTG2000. If 'PSTN' is selected, the ringback tone is played from the PSTN

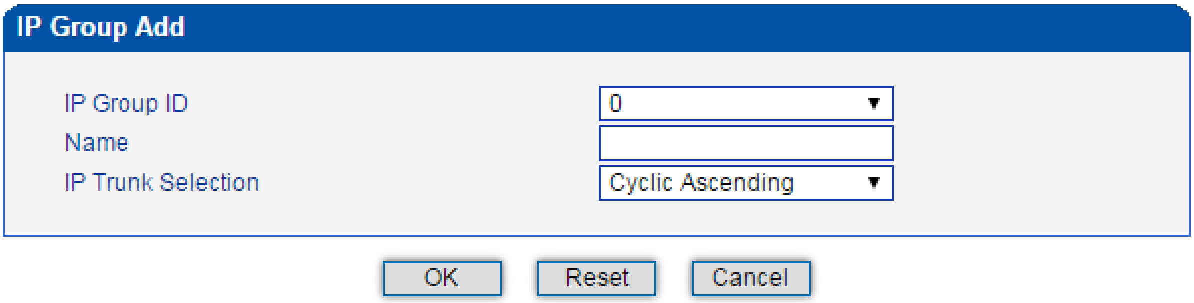
Wait for RTP Packet from Peer	If 'Yes' is selected, RTP packets will be sent from peer device to MTG200 first, and then RTP packets will be sent from MTG to peer device. If 'No' is selected, RTP packets will be sent automatically during calling;
T.30 Expanded Type in SDP	There are two T.30 expanded types: X-Fax and Fax

4.10.2 IP Group

On the IP Group interface, you can add IP groups and choose a strategy for selecting IP trunks.



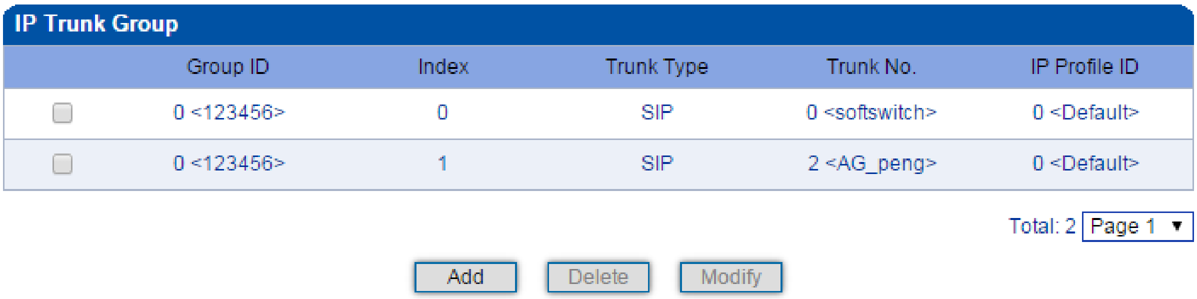
Click **Add**, and the following interface will be displayed.



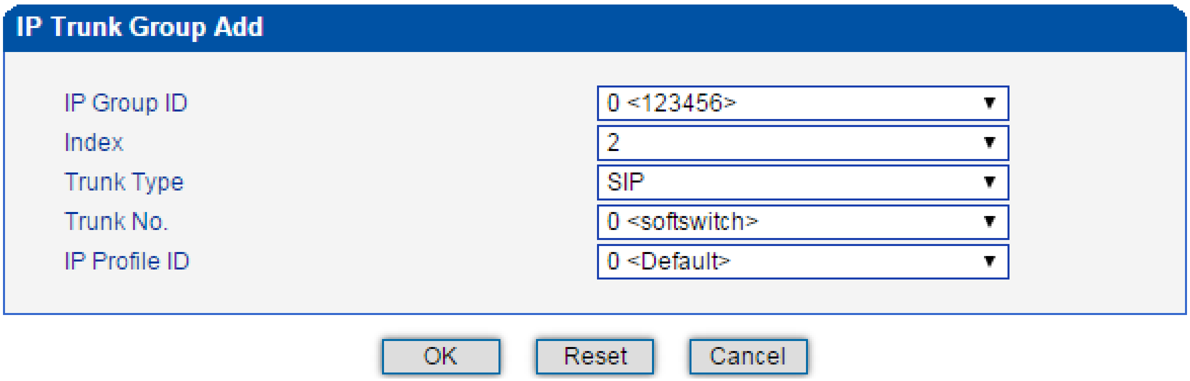
IP Trunk Selection	Ascending	To select IP trunks in an ascending order under a same group.
	Cyclic Ascending:	To select IP trunks in an ascending order, starting from the previous IP trunk that has been selected
	Descending	To select IP trunks in a descending order under a same group
	Cyclic Descending	To select IP trunks in a descending order, starting from the previous IP trunk that has been selected

4.10.3 IP Group Management

On the **IP Group Management** interface, you can add IP trunks to the IP group which have been established on **IP Group** interface.



Click **Add**, and you can see the following interface.



Parameter	**Explanation**
IP Group ID	The ID of the IP group If you want to add more IP trunks to the IP group, do not change the IP group ID.
Index	The index of the IP trunk added to the IP group
Trunk Type	SIP
Trunk No.	Select an IP trunk that has been established on **SIP Config → SIP Trunk** interface.
IP Profile ID	The ID of the IP profile that will be used by the IP trunk.

4.11 Number Filter

This section is mainly to introduce how to configure white & black lists on the MTG2000 gateway.

Caller White List: Calls from the numbers on the Caller White List will be allowed to pass. If a caller number cannot match with one of the numbers on the Caller White List, calls from the caller number will be rejected.

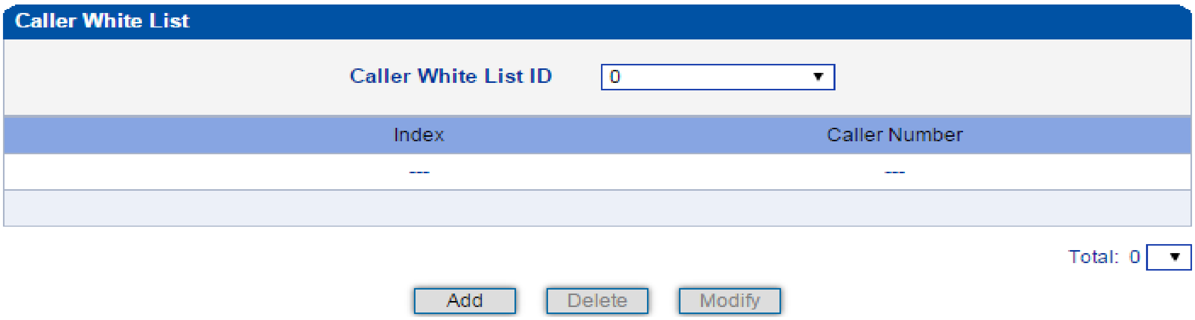
Caller Black List: Calls from the numbers on the Caller Black List will be rejected to pass. If a caller number match with one of the numbers on the Caller Black List, calls will be rejected.

Callee White List: Calls to the numbers on the Callee White List will be allowed to pass. If a callee number cannot match with one of one of the numbers on the Caller White List, calls to the callee number will be rejected.

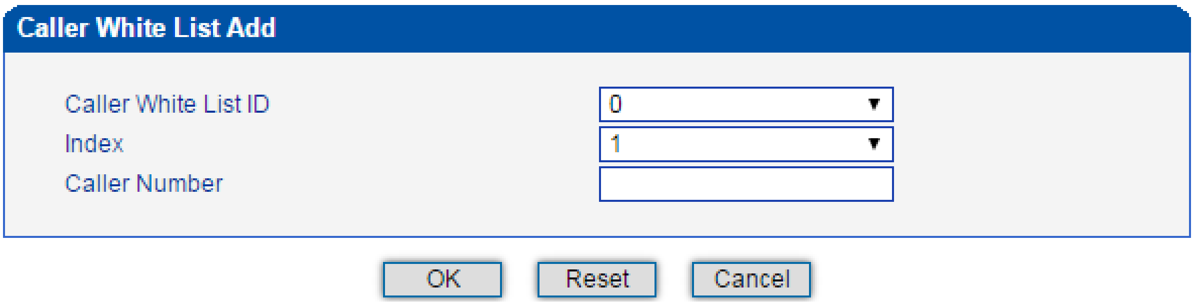
Callee Black List: Calls to the numbers on the Callee Black List will be rejected to pass. If a callee number match with one of the numbers on the Callee Black List, calls to the callee number will be rejected.

4.11.1 Procedures to add a number on the Caller White List

1. Click **Number Filter → Caller White List** to enter into the following interface.



2. Click **Add** to enter into the following interface to add a caller number on the Caller White List



3. Choose an ID for the caller white list and an index for the caller number, and then enter the caller number

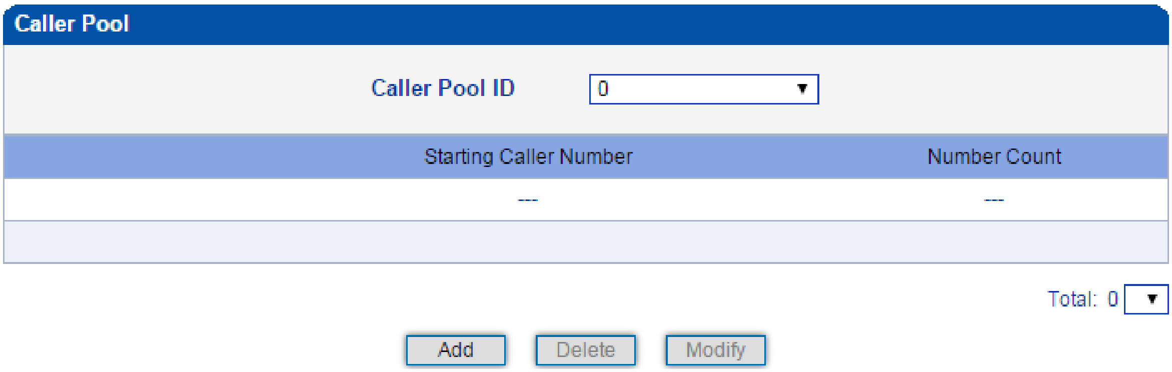
4. Click **OK**.

Note:

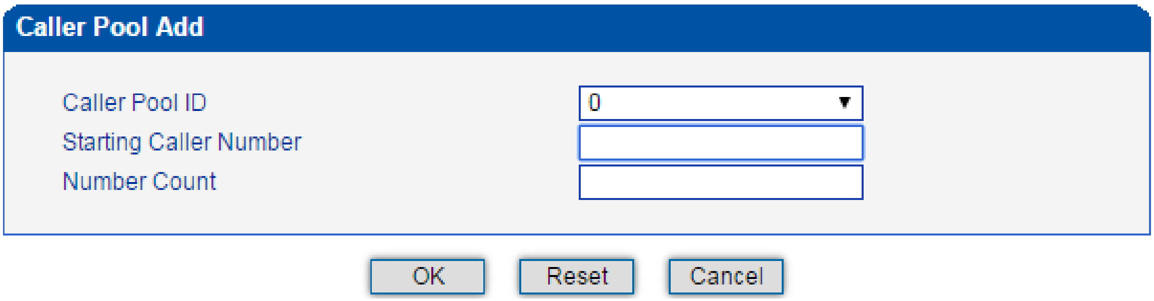
You can add 8 white or black lists at most, with ID from 0 to 7. And each white or black list can contain 1024 numbers at most.

4.11.2 Caller Pool

On the **Caller Pool** interface, you can add a batch of telephone numbers to replace the actual caller numbers when there is a need.



Click **Add** to set numbers in the caller pool.



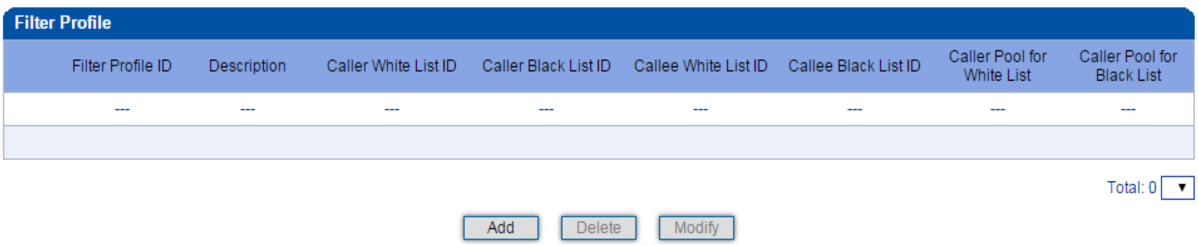
Note:

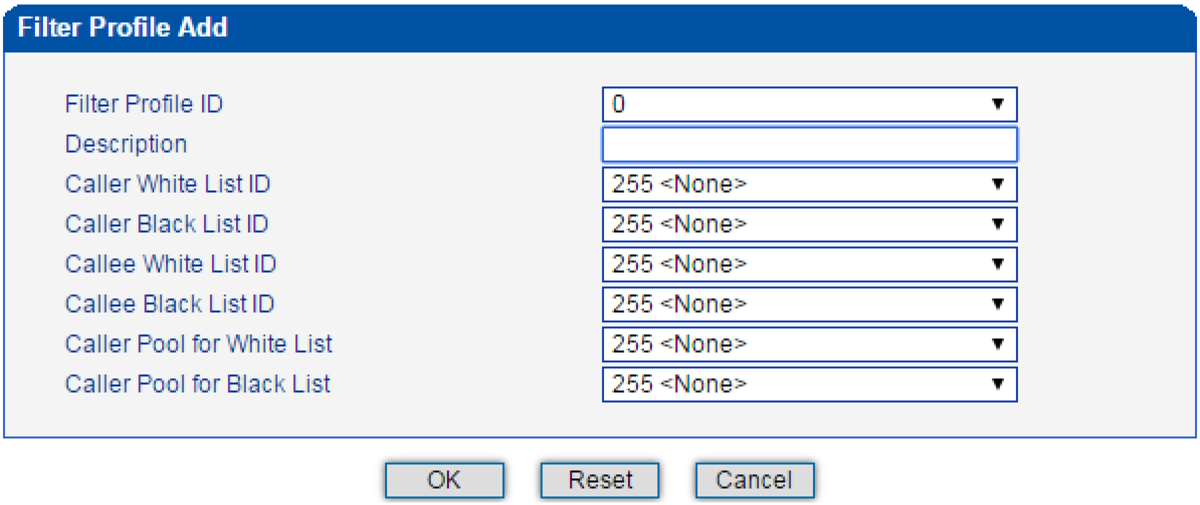
If 'Starting Caller Number' is 80080000 and 'Number Count' is 100, it means numbers from 80080000 to 80080099 are all in the caller pool.

Each caller poor can contain 512 numbers at most, and if there are multiple caller pools, the caller pools can contain up to 1024 numbers in total.

4.11.3 Filter Profile

On the **Filter Profile** interface, you can put white lists and black lists that have been set before in a filter profile or several profiles. The white lists and black lists will not take effect until you set them in filter profiles.



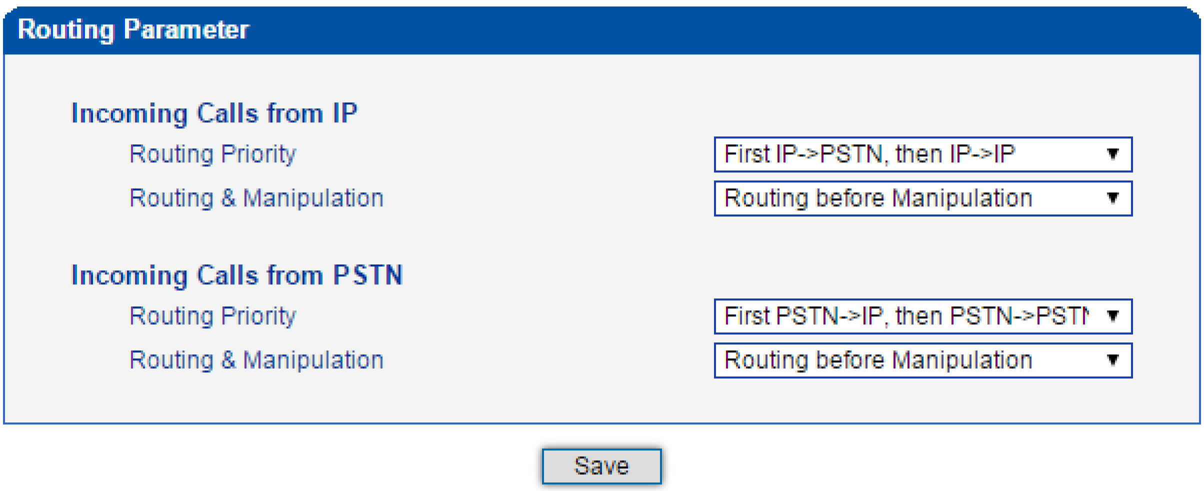


Select a white list ID, and the calls of the numbers on the white list will be passed. Select a black list ID, and the calls of the numbers on the black list will be prohibited.

If you select **255<None>**, it means no while lists or black lists are set in filter profile, and no numbers will be filtered.

4.12 Call Routing

4.12.1 Routing Parameter

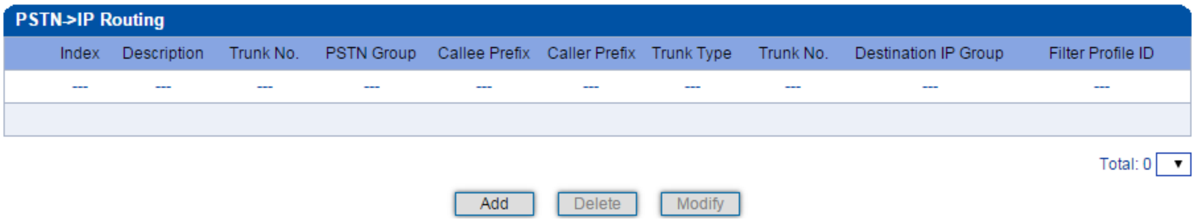


Belong To	Parameter	Explanation
Incoming Calls from IP	Routing Priority	There are two options: First IP→PSTN, then IP →IP First IP→IP, then IP →PSTN
	Routing & Manipulation	There are two options: Routing before Manipulation Routing after Manipulation

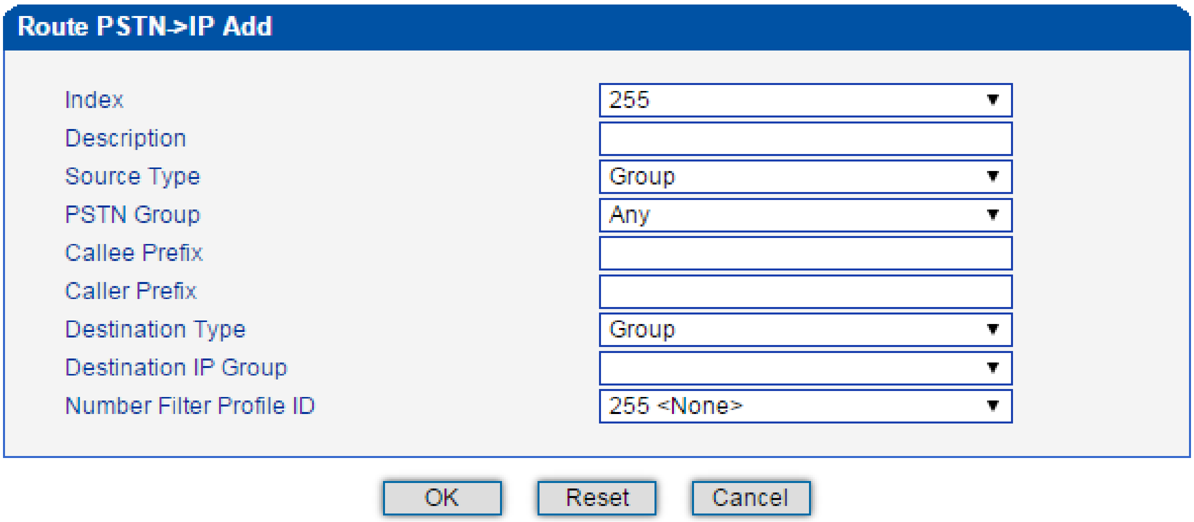
Incoming Calls from PSTN	Routing Priority	First PSTN →IP, then PSTN→PSTN
	Routing & Manipulation	There are two options: Routing before Manipulation Routing after Manipulation

4.12.2 PSTN→ IP Routing

On the **PSTN→ IP Routing** interface, you can set routing parameters for PSTN → IP calls.



Click **Add**, and the following interface will be displayed.

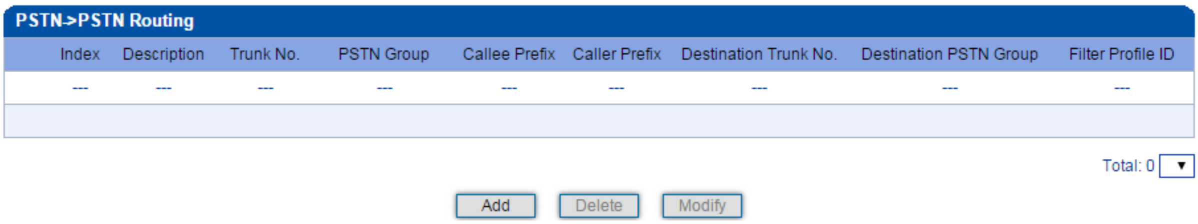


Parameter	Explanation
Index	The Index of the PSTN →IP route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the PSTN →IP route,
Source Type	Sources include PSTN group and PRI/SS7 trunk.
PSTN Group	If source is PSTN group, please select a specific PSTN group. If ‘Any’ is selected, it means the source is any PSTN group.
PSTN Trunk	If source is PSTN trunk, please select a specific PRI/SS7 trunk. If ‘Any’ is selected, it means the source is any PRI/SS7 trunk. .

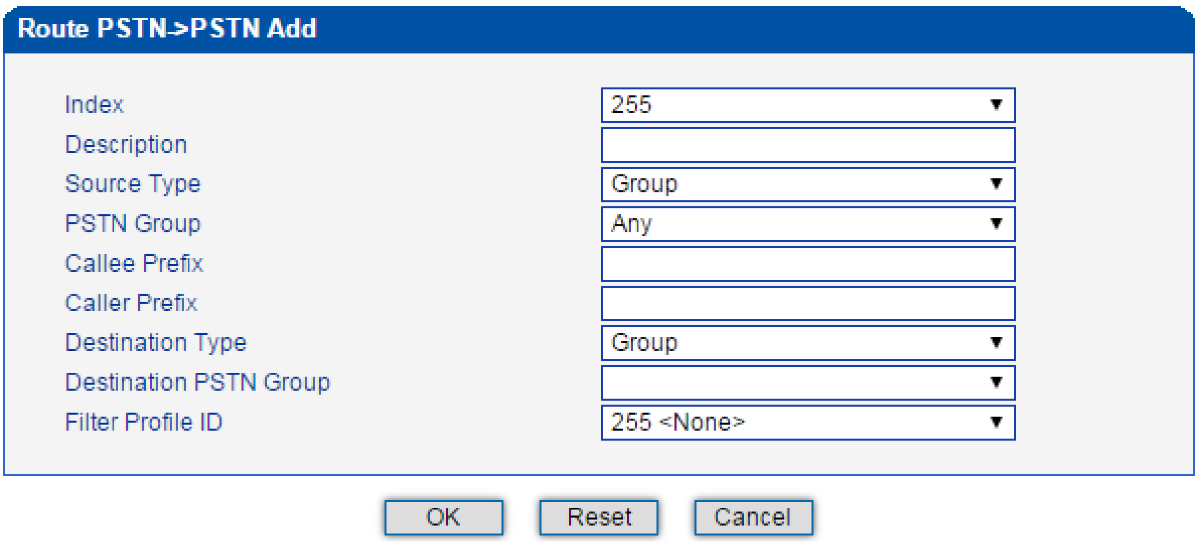
Callee Prefix	The prefix configured for callee number. When a callee number matches the prefix, this PSTN →IP route will be used. '.' is a wildcard, which means this PSTN →IP route will be used, no matter what the callee number is.
Caller Prefix	The prefix configured for caller number. When a caller number matches the prefix, this PSTN →IP route will be used. '.' is a wildcard, which means this PSTN →IP route will be used, no matter what the caller number is.
Destination Type	Destination is IP group or SIP trunk.
Destination IP Group	If source is IP group, please select a specific IP group.
IP Trunk No.	If source is SIP trunk, please select a specific IP trunk.
Number Filter Profile ID	The ID of filter profile. The white lists and black lists set in the filter profile will apply to this PSTN →IP route.

4.12.3 PSTN → PSTN Routing

On the **PSTN→PSTN Routing** interface, you can set routing parameters for PSTN → PSTN calls.



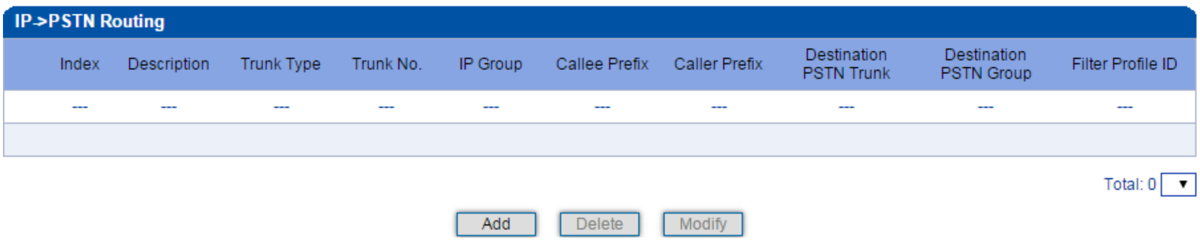
Click **Add**, and the following interface will be displayed.



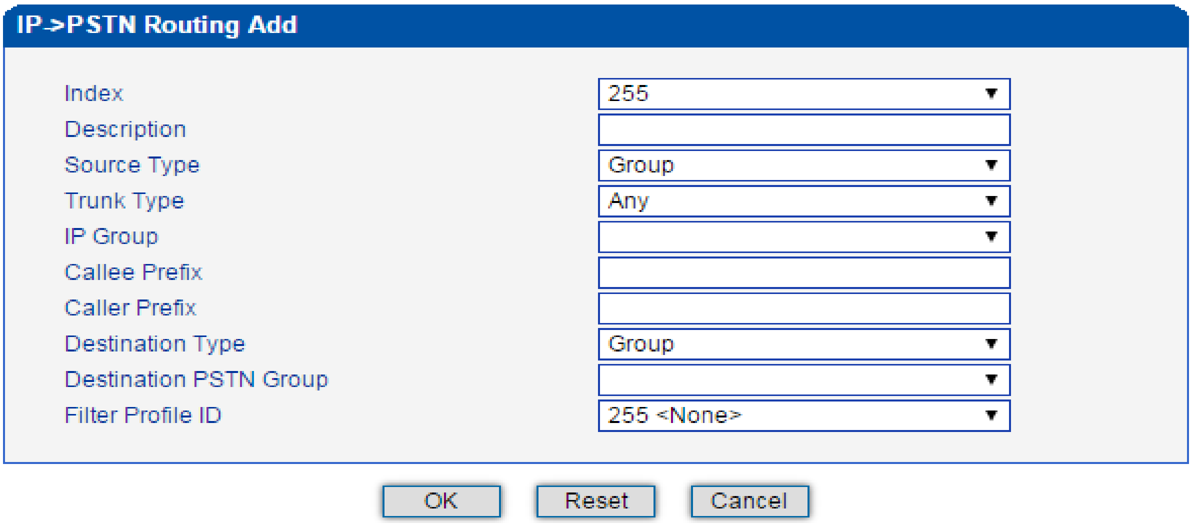
Parameter	Explanation
Index	The Index of the PSTN →PSTN route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the PSTN →PSTN route,
Source Type	Sources include PSTN group and PRI/SS7 trunk.
PSTN Group	If source is PSTN group, please select a specific PSTN group. If 'Any' is selected, it means the source is any PSTN group.
PSTN Trunk	If source is PSTN trunk, please select a specific PRI/SS7 trunk. If 'Any' is selected, it means the source is any PRI/SS7 trunk. .
Callee Prefix	The prefix configured for callee number. When a callee number matches the prefix, this PSTN →IP route will be used. '.' is a wildcard, which means this PSTN →PSTN route will be used, no matter what the callee number is.
Caller Prefix	The prefix configured for caller number. When a caller number matches the prefix, this PSTN →PSTN route will be used. '.' is a wildcard, which means this PSTN →PSTN route will be used, no matter what the caller number is.
Destination Type	Destination is PSTN group or PRI/SS7 trunk.
Destination IP Group	If source is PSTN group, please select a specific PSTN group.
IP Trunk No.	If source is PRI/SS7 trunk, please select a specific PRI/SS7 trunk.
Number Filter Profile ID	The ID of filter profile. The white lists and black lists set in the filter profile will apply to this PSTN →PSTN route.

4.12.4 IP → PSTN Routing

On the **PSTN→ IP Routing** interface, you can set routing parameters for IP → PSTN calls.



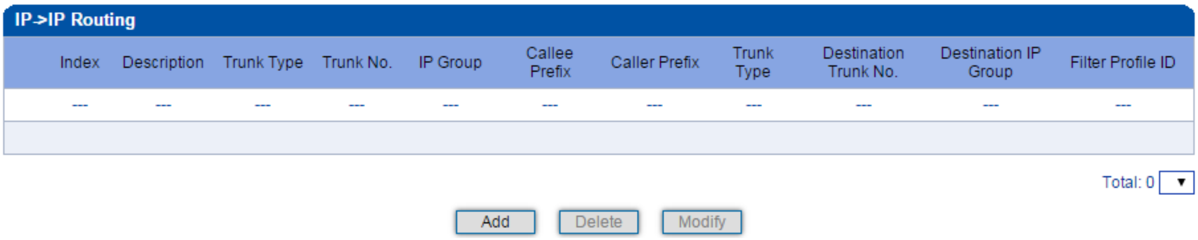
Click **Add**, and the following interface will be displayed.



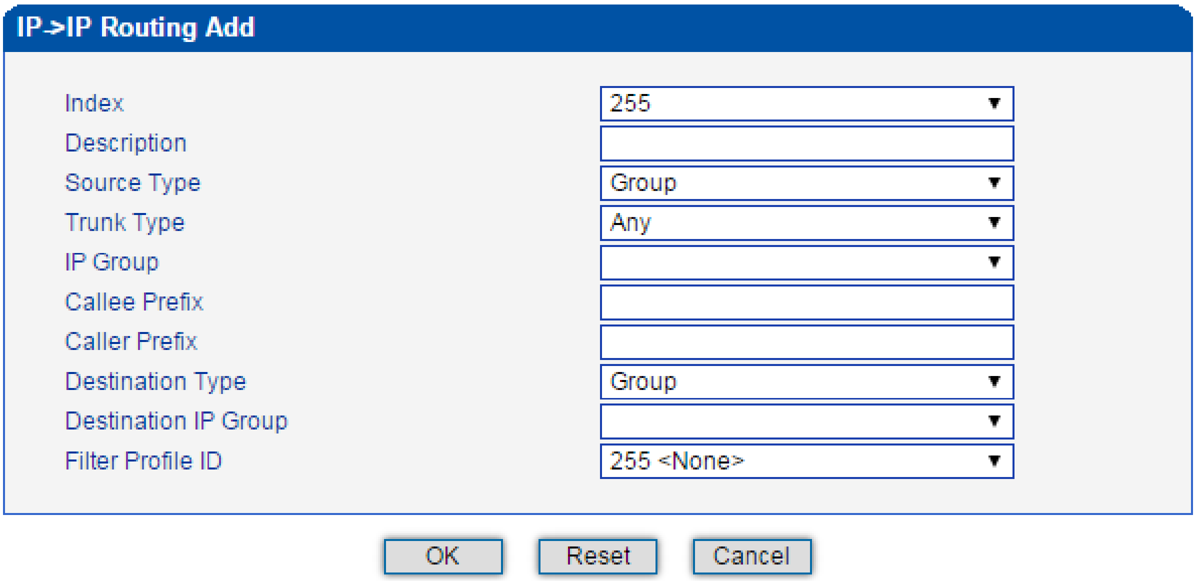
Parameter	Explanation
Index	The Index of the IP→PSTN route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the IP →PSTN route,
Source Type	Sources include IP group and IP trunk.
PSTN Group	If source is IP group, please select a specific IP group. If 'Any' is selected, it means the source is any IP group.
PSTN Trunk	If source is IP trunk, please select a specific SIP trunk. If 'Any' is selected, it means the source is any IP trunk. .
Callee Prefix	The prefix configured for callee number. When a callee number matches the prefix, this IP→PSTN route will be used. '.' is a wildcard, which means this IP→PSTN route will be used, no matter what the callee number is.
Caller Prefix	The prefix configured for caller number. When a caller number matches the prefix, this IP→PSTN route will be used. '.' is a wildcard, which means this IP→PSTN route will be used, no matter what the caller number is.
Destination Type	Destination is PSTN group or PRI/SS7 trunk.
Destination IP Group	If source is PSTN group, please select a specific PSTN group.
IP Trunk No.	If source is PRI/SS7 trunk, please select a specific PRI/SS7 trunk.
Number Filter Profile ID	The ID of filter profile. The white lists and black lists set in the filter profile will apply to this PSTN →PSTN route.

4.12.5 IP → IP Routing

On the **IP→IP Routing** interface, you can set routing parameters for IP → IP calls.



Click **Add**, and the following interface will be displayed.



Parameter	Explanation
Index	The Index of the IP→ IP route, from 0 to 255. Greater index value, higher priority for the route.
Description	The description of the IP → IP route,
Source Type	Sources include IP group and IP trunk.
PSTN Group	If source is IP group, please select a specific IP group. If 'Any' is selected, it means the source is any IP group.
PSTN Trunk	If source is IP trunk, please select a specific SIP trunk. If 'Any' is selected, it means the source is any IP trunk. .
Callee Prefix	The prefix configured for callee number. When a callee number matches the prefix, this IP→ IP route will be used. '.' is a wildcard, which means this IP→ IP route will be used, no matter what the callee number is.

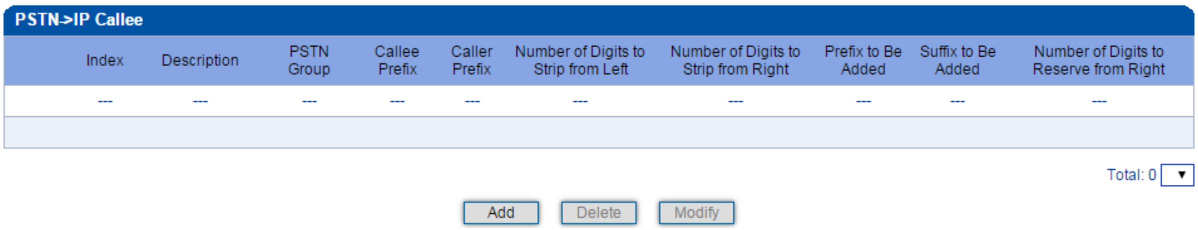
Caller Prefix	The prefix configured for caller number. When a caller number matches the prefix, this IP→ IP route will be used. '.' is a wildcard, which means this IP→ IP route will be used, no matter what the caller number is.
Destination Type	Destination is IP group or SIP trunk.
Destination IP Group	If source is IP group, please select a specific IP group.
IP Trunk No.	If source is SIP trunk, please select a specific PRI/SS7 trunk.
Number Filter Profile ID	The ID of filter profile. The white lists and black lists set in the filter profile will apply to this IP → IP route.

4.13 Number Manipulation

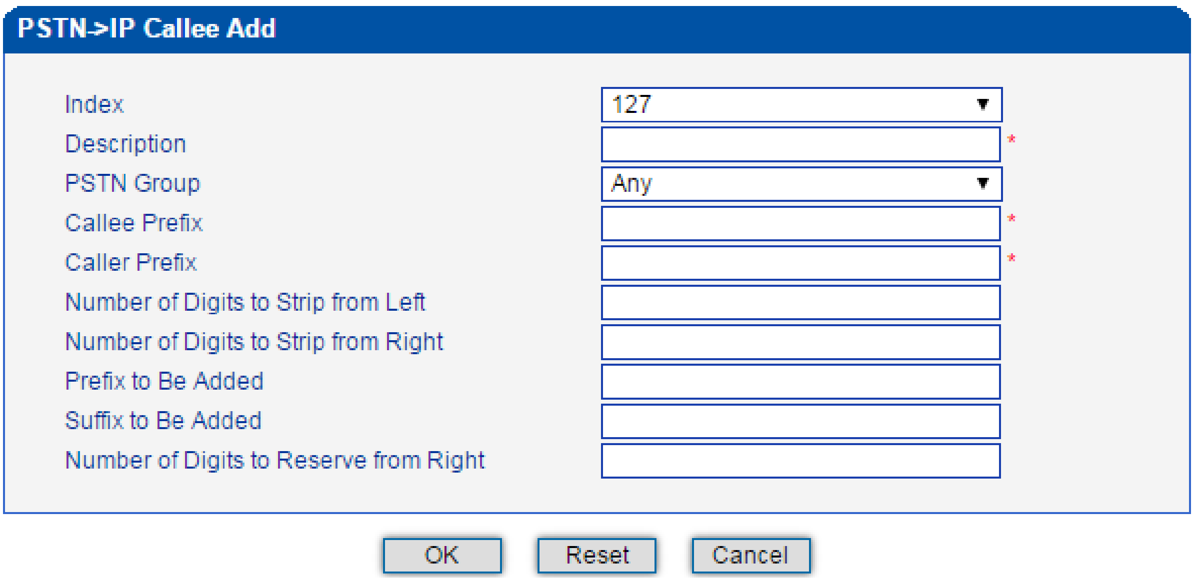
Number manipulation refers to the change of the caller number or callee number during calling process.

4.13.1 PSTN → IP Callee

On the **PSTN→ IP Callee** interface, you can set rules to change the actual callee number during PSTN → IP calling process.



Click **Add**, and the following interface will be displayed.



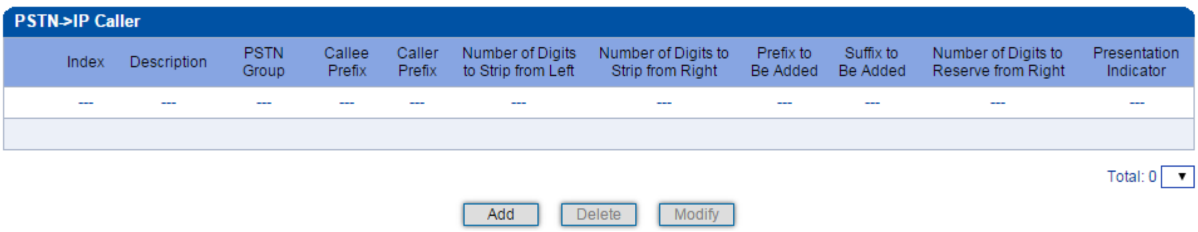
Parameter	Explanation
Index	The index of this PSTN →IP callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →IP callee number manipulation
PSTN Group	Select a PSTN group. The callee number will be manipulated when a call uses a trunk of this PSTN group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any PSTN group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the callee number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the callee number
Prefix to be added	The prefix added to the callee number after its digits are lessened.
Suffix to be added	The suffix added to the callee number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the callee number

For example:
If the called number is 25026531014, how do you change it into 026531014 ?
You can enter '3' in the value box for the 'Number of Digits to Strip from Left' parameter.

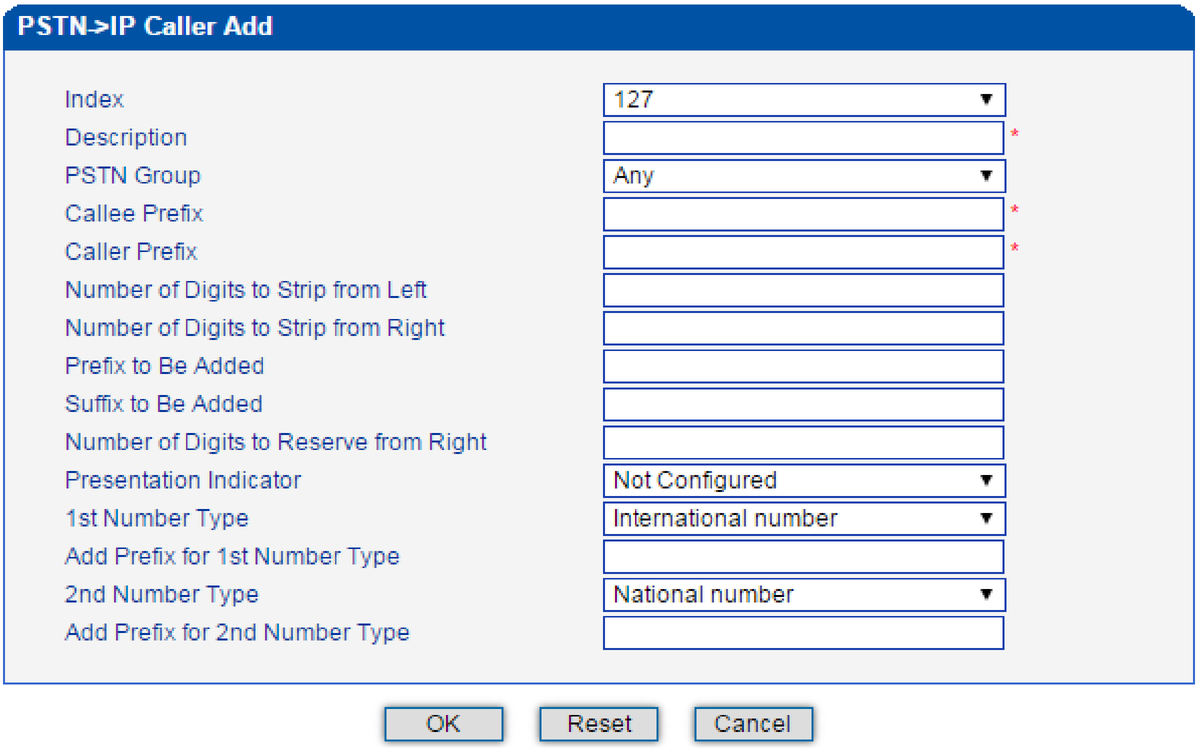
If the called number is 2653101413, how do you change it into 00912653101413?
You can enter ' 0091' in the value box for the 'Callee Prefix' parameter.

4.13.2 PSTN→IP Caller

On the **PSTN→ IP Caller** interface, you can set rules to change the actual caller number during PSTN → IP calling process.



Click **Add**, and the following interface will be displayed.

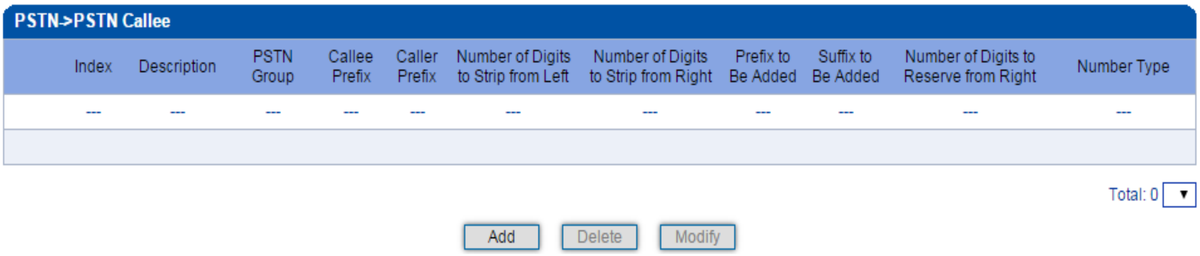


Parameter	Explanation
Index	The index of this PSTN →IP caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →IP caller number manipulation
PSTN Group	Select a PSTN group. The caller number will be manipulated when a call uses a trunk of this PSTN group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any PSTN group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the caller number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the caller number
Prefix to be added	The prefix added to the caller number after its digits are lessened.
Suffix to be added	The suffix added to the caller number after its digits are lessened.

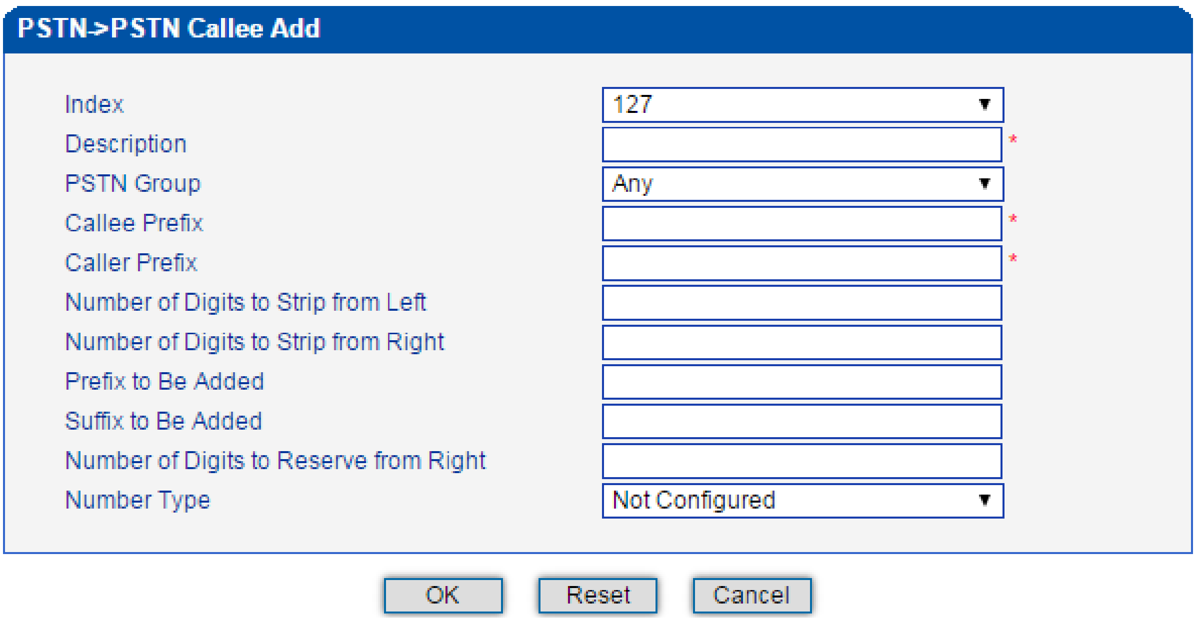
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the caller number
Presentation Indicator	If "Allowed" is selected, the calling number will be presented. If "Restricted" is selected, the calling number will not be presented. If "Not Config" is selected, the parameter does not work.
1st Number Type	If the caller number belongs to 1st number type, the set prefix will be added to the caller number.
Add Prefix for 1st Number Type	The prefix that will be added to those numbers that belong to 1st number type
2nd Number Type	If the caller number belongs to 2nd number type, the set prefix will be added to the caller number.
Add Prefix for 2nd Number Type	The prefix that will be added to those numbers that belong to 2nd number type

4.13.3 PSTN→PSTN Callee

On the **PSTN→ PSTN Callee** interface, you can set rules to change the actual callee number during PSTN → PSTN calling process.



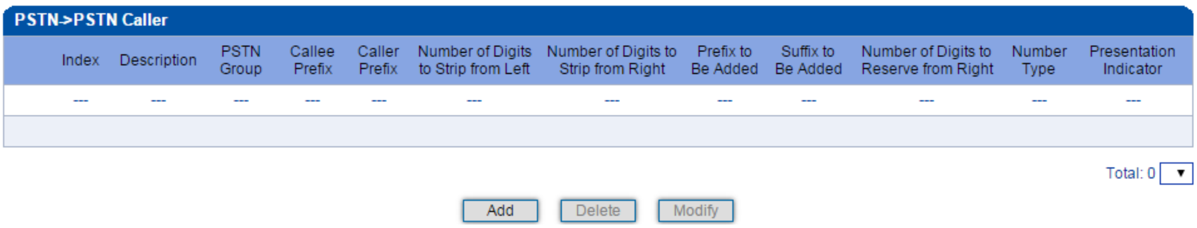
Click **Add**, and the following interface will be displayed.



Parameter	Explanation
Index	The index of this PSTN →PSTN callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →PSTN callee number manipulation
PSTN Group	Select a PSTN group. The callee number will be manipulated when a call uses a trunk of this PSTN group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any PSTN group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the callee number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the callee number
Prefix to be added	The prefix added to the callee number after its digits are lessened.
Suffix to be added	The suffix added to the callee number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the callee number
Number Type	The type of the callee number. Options include 'Not Config', 'International', 'National', 'Unknown', 'Network Specific', 'Subscriber' and 'Abbreviated'

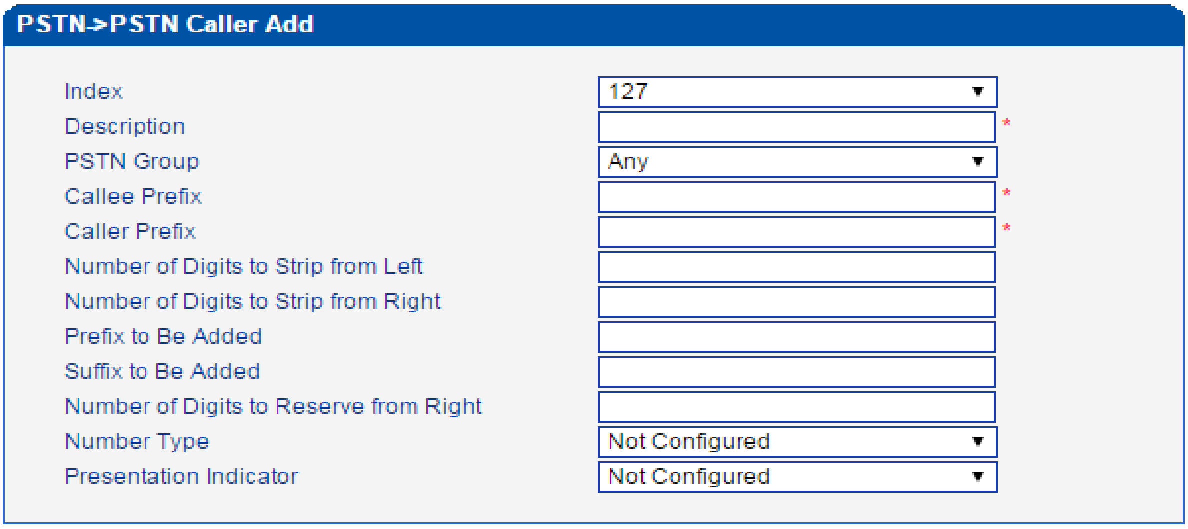
4.13.4 PSTN →PSTN Caller

On the **PSTN→ PSTN Caller** interface, you can set rules to change the actual caller number during PSTN → PSTN calling process.


PSTN->PSTN Caller

Index	Description	PSTN Group	Callee Prefix	Caller Prefix	Number of Digits to Strip from Left	Number of Digits to Strip from Right	Prefix to Be Added	Suffix to Be Added	Number of Digits to Reserve from Right	Number Type	Presentation Indicator
---	---	---	---	---	---	---	---	---	---	---	---

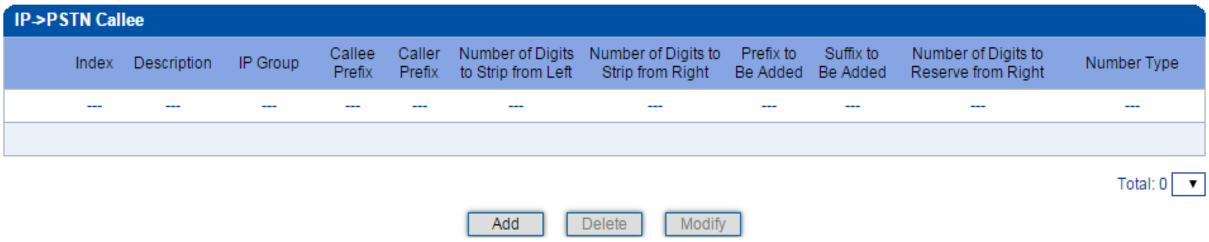
Click **Add**, and the following interface will be displayed.



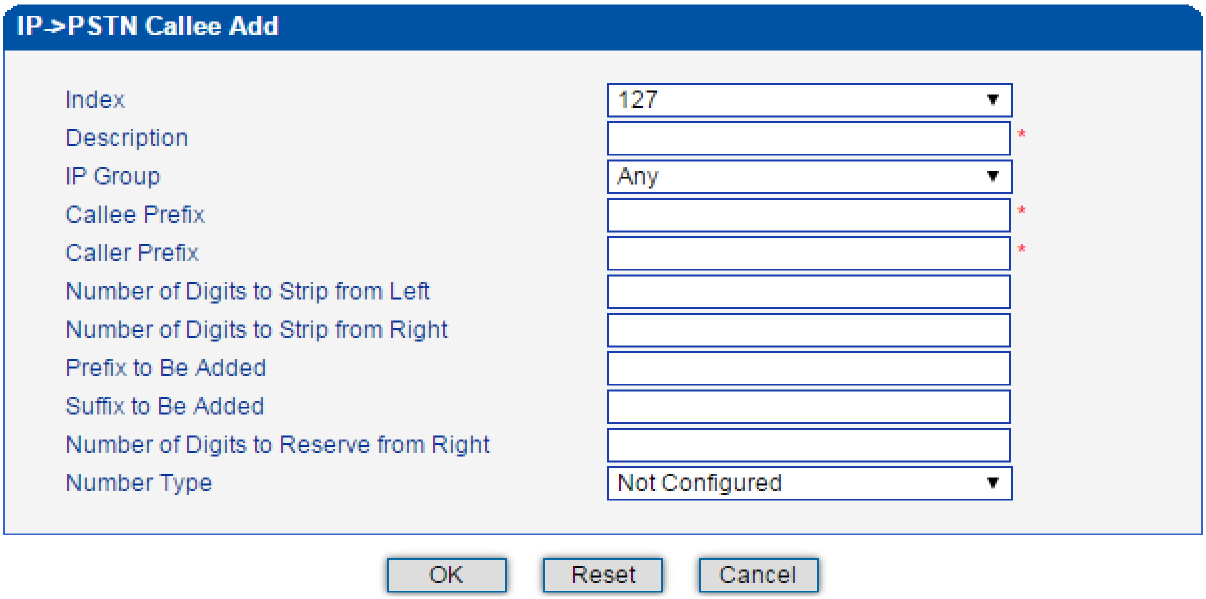
Parameter	Explanation
Index	The index of this PSTN →PSTN caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this PSTN →PSTN caller number manipulation
PSTN Group	Select a PSTN group. The caller number will be manipulated when a call uses a trunk of this PSTN group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any PSTN group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the caller number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the caller number
Prefix to be added	The prefix added to the caller number after its digits are lessened.
Suffix to be added	The suffix added to the caller number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the caller number
Presentation Indicator	If "Allowed" is selected, the calling number will be presented. If "Restricted" is selected, the calling number will not be presented. If "Not Config" is selected, the parameter does not work.
Number Type	The type of the caller number. Options include 'Not Config', 'International', 'National', 'Unknown', 'Network Specific', 'Subscriber' and 'Abbreviated'

4.13.5 IP→PSTN Callee

On the **IP→ PSTN Callee** interface, you can set rules to change the actual callee number during IP→ PSTN calling process.



Click **Add**, and the following interface will be displayed.

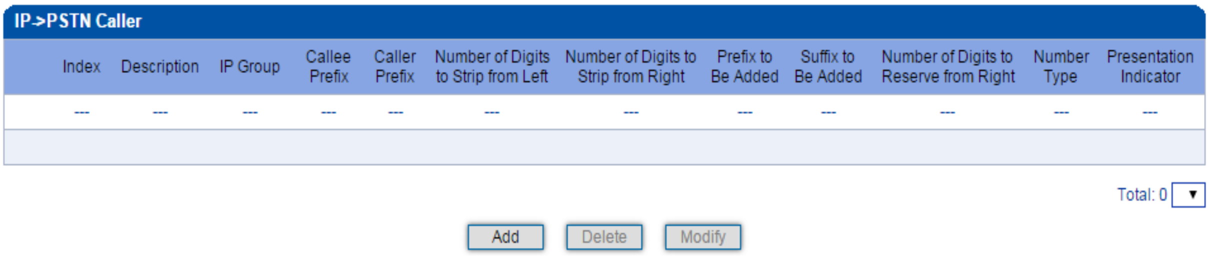


Parameter	Explanation
Index	The index of this IP →PSTN callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this IP →PSTN callee number manipulation
IP Group	Select an IP group. The callee number will be manipulated when a call uses a trunk of this IP group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any IP group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the caller number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the caller number

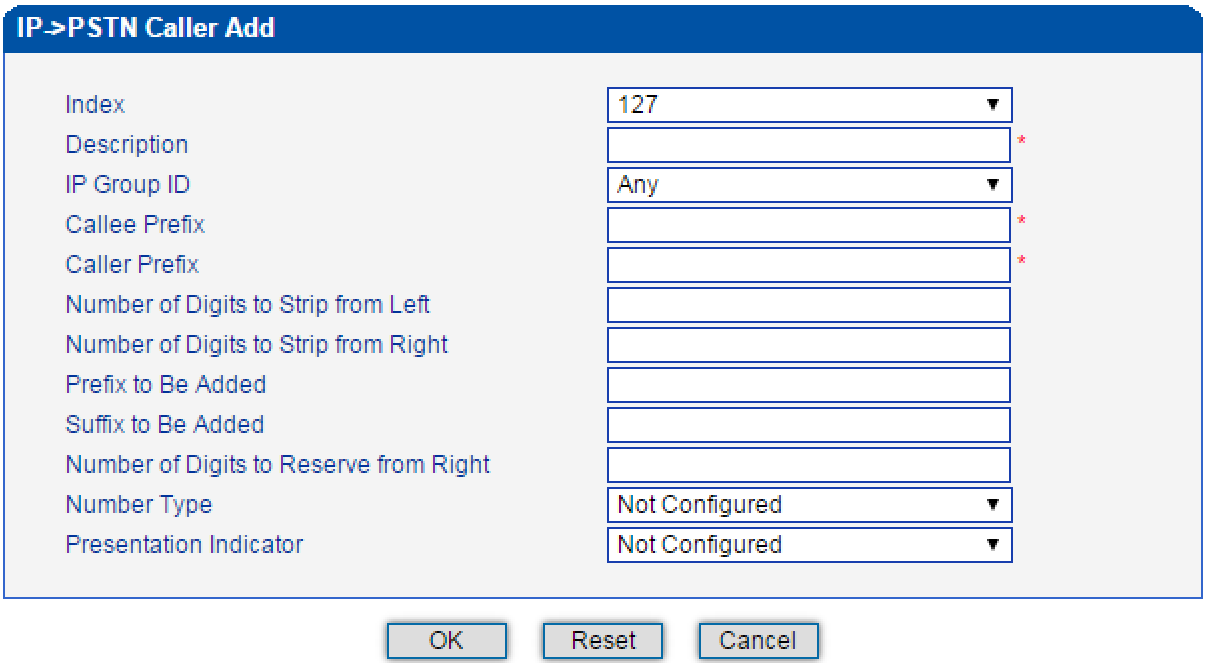
Prefix to be added	The prefix added to the callee number after its digits are lessened.
Suffix to be added	The suffix added to the callee number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the callee number
Number Type	The type of the callee number. Options include ‘Not Config’, ‘International’, ‘National’, ‘Unknown’, ‘Network Specific’, ‘Subscriber’ and ‘Abbreviated’

4.13.6 IP→PSTN Caller

On the **IP→ PSTN Caller** interface, you can set rules to change the actual caller number during IP→ PSTN calling process.



Click **Add**, and the following interface will be displayed.

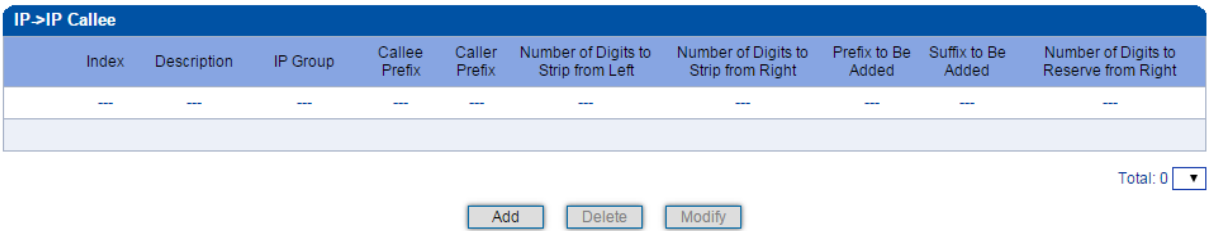


Parameter	**Explanation**
Index	The index of this IP →PSTN caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.

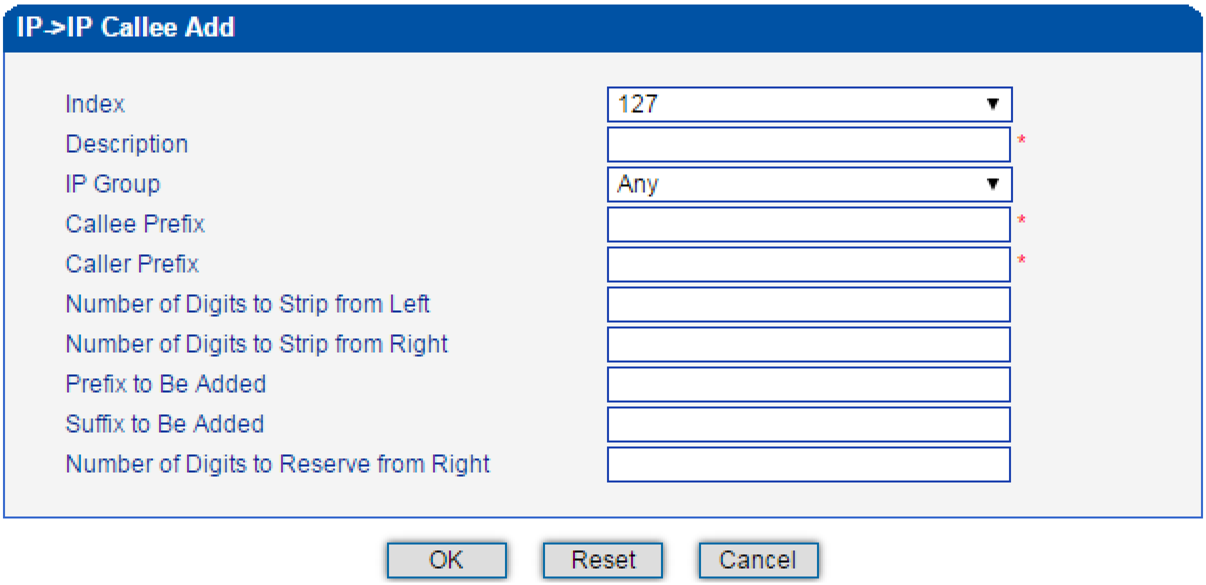
Description	The description of this IP →PSTN caller number manipulation
IP Group	Select an IP group. The caller number will be manipulated when a call uses a trunk of this IP group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any IP group.
Callee Prefix	Set a prefix for the callee number.
Caller Prefix	Set a prefix for the caller number
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the caller number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the caller number
Prefix to be added	The prefix added to the caller number after its digits are lessened.
Suffix to be added	The suffix added to the caller number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the caller number
Presentation Indicator	If "Allowed" is selected, the calling number will be presented. If "Restricted" is selected, the calling number will not be presented. If "Not Config" is selected, the parameter does not work.
Number Type	The type of the caller number. Options include 'Not Config', 'International', 'National', 'Unknown', 'Network Specific', 'Subscriber' and 'Abbreviated'

4.13.7 IP → IP Callee

On the **IP→ IP Callee** interface, you can set rules to change the actual callee number during IP→ IP calling process.



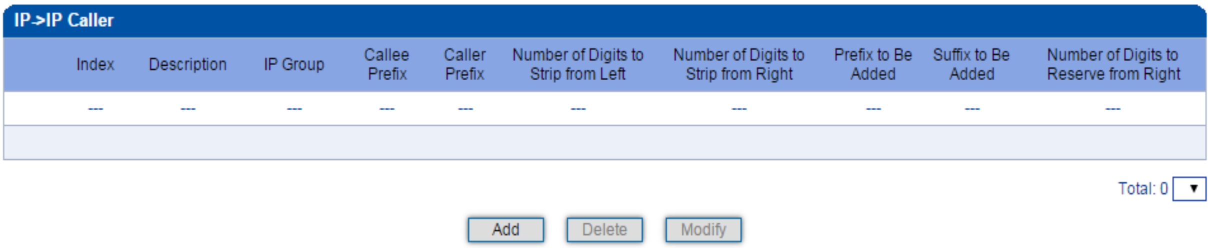
Click **Add**, and the following interface will be displayed.



Parameter	Explanation
Index	The index of this IP →IP callee number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this IP →IP callee number manipulation
IP Group	Select an IP group. The callee number will be manipulated when a call uses a trunk of this IP group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any IP group.
Callee Prefix	Set a prefix for the callee number. If the actual callee prefix matches this set callee prefix, the callee number will be manipulated.
Caller Prefix	Set a prefix for the caller number. If the actual caller prefix matches the set caller prefix, the callee number will be manipulated.
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the callee number
Number of Digits to Strip from Right	The number of digits which are lessened from the right of the callee number
Prefix to be added	The prefix added to the callee number after its digits are lessened.
Suffix to be added	The suffix added to the callee number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the callee number

4.13.8 IP → IP Caller

On the **IP → IP Caller** interface, you can set rules to change the actual caller number during IP→ IP calling process.

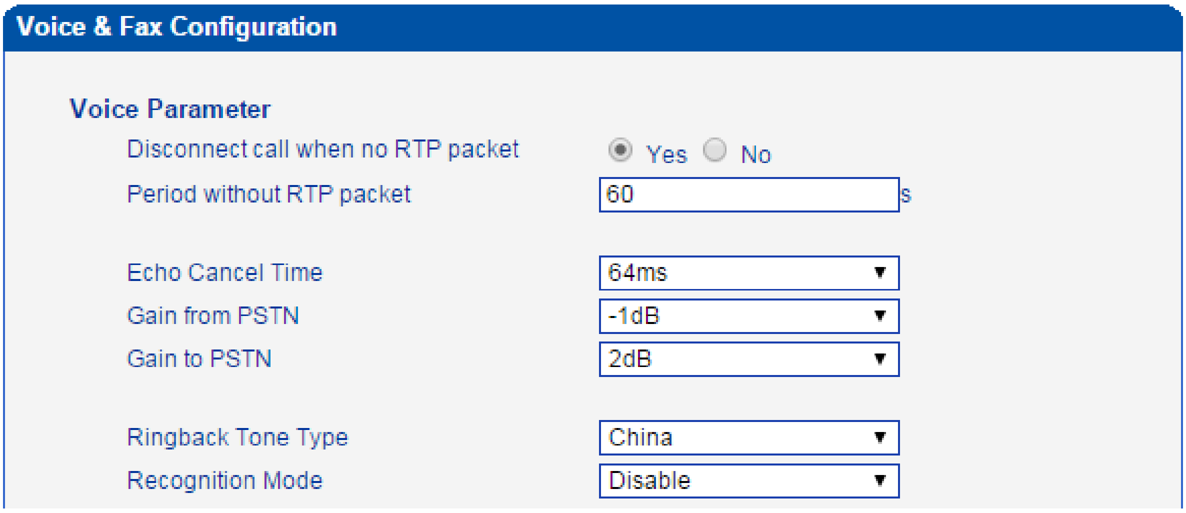


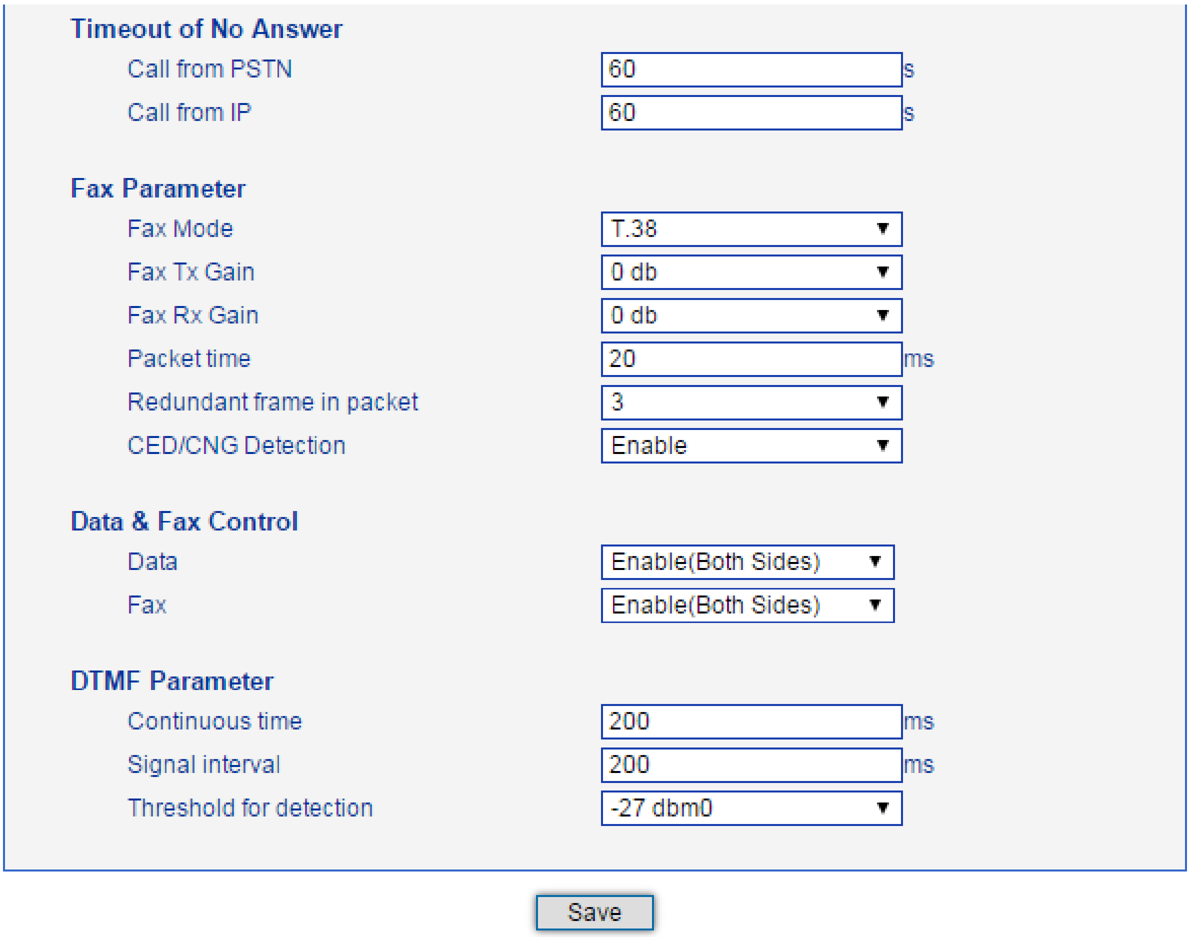
Click **Add**, and the following interface will be displayed.

Parameter	Explanation
Index	The index of this IP →IP caller number manipulation, from 0 to 127. Each index cannot be used repeatedly.
Description	The description of this IP →IP caller number manipulation
IP Group	Select an IP group. The caller number will be manipulated when a call uses a trunk of this IP group, actual callee prefix matches the set callee prefix, and actual caller prefix matches the set caller prefix. 'Any' means any IP group.
Callee Prefix	Set a prefix for the callee number. If the actual callee prefix matches this set prefix, the caller number will be manipulated.
Caller Prefix	Set a prefix for the caller number. If the actual caller prefix matches the set prefix, the caller number will be manipulated.
Number of Digits to Strip from Left	The number of digits which are lessened from the left of the caller number

Number of Digits to Strip from Right	The number of digits which are lessened from the right of the caller number
Prefix to be added	The prefix added to the caller number after its digits are lessened.
Suffix to be added	The suffix added to the caller number after its digits are lessened.
Number of Digits to Reserve from Right	The number of the retained digits which. are counted from the right of the caller number

4.14 Voice & Fax



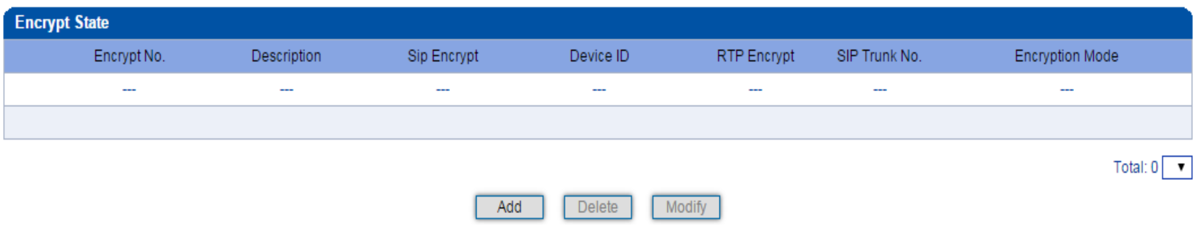


Belong to	Parameter	Explanation
Voice Parameter	Disconnect call when no RTP packet	Options include 'Yes' and 'No'. If 'Yes' is selected, the call will be disconnected when it is detected that the call's silence time is longer than the set maximum time without receiving RTP packets.
	Period without RTP packet	The set maximum time without receiving RTP packets. Default value is 60 seconds.
	Echo Cancel Time	The interval to remove echo from a voice communication. Options include 32ms, 64ms and 128ms.
	Gain from PSTN	The voice gain from PSTN to IP direction Default value is -1dB
	Gain to PSTN	The voice gain from IP to PSTN direction Default value is 2dB
	Ringback Tone Type	Local ringback tone

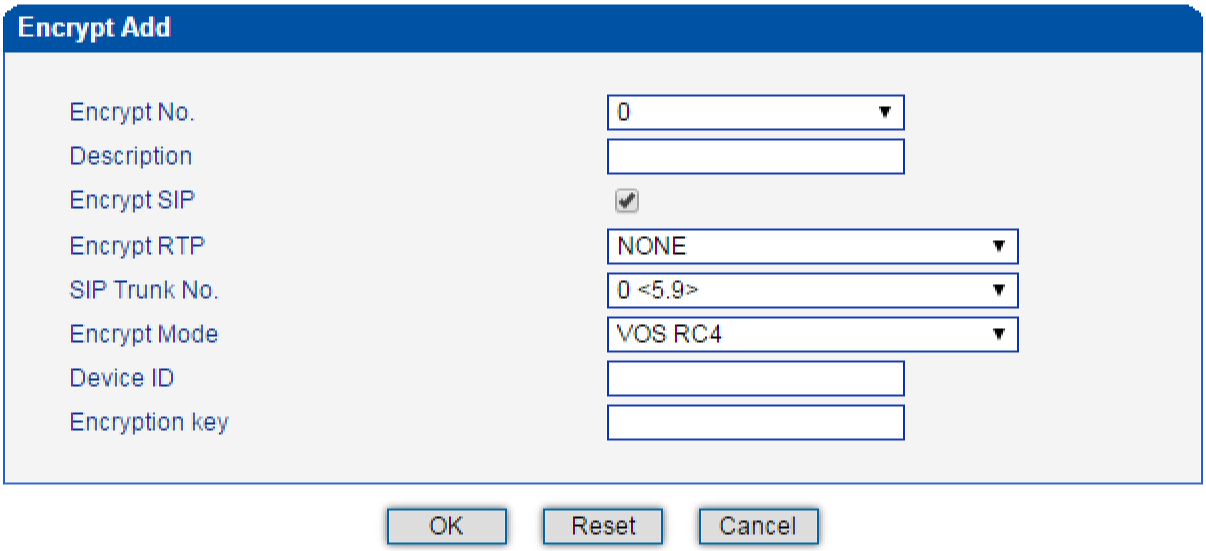
	Recognition Mode	Whether to recognize voice when prompt tone is played.
Timeout of No Answer	Call from PSTN	The maximum time of no answer for calls from PSTN
	Call from IP	The maximum time of no answer for calls from IP Network
Fax Parameter	Fax Mode	Options include T.38, Pass-through and Adaptive. Default value is T.38. Adaptive means auto negotiate with peer side.
	Fax Tx Gain	Gain of sending a fax
	Fax Rx Gain	Gain of receiving a fax
	Packet time	The time for data packing
	Redundant frame in Packet	The length of frame in RTP packet
	CED/CNG Detection	Whether to detect CED/CNG
Data & Fax Control	Data	Whether to enable voice data service on the MTG2000
	Fax	Whether to enable fax service on the MTG2000
DTMF Parameter	Continuous time	The duration of a DTMF signal
	Signal Interval	The interval between two DTMF signals
	Threshold for Detection	The signal detection threshold

4.15 Encrypt Config

On the **Encrypt Config** interface, you can set parameters related to encryption.



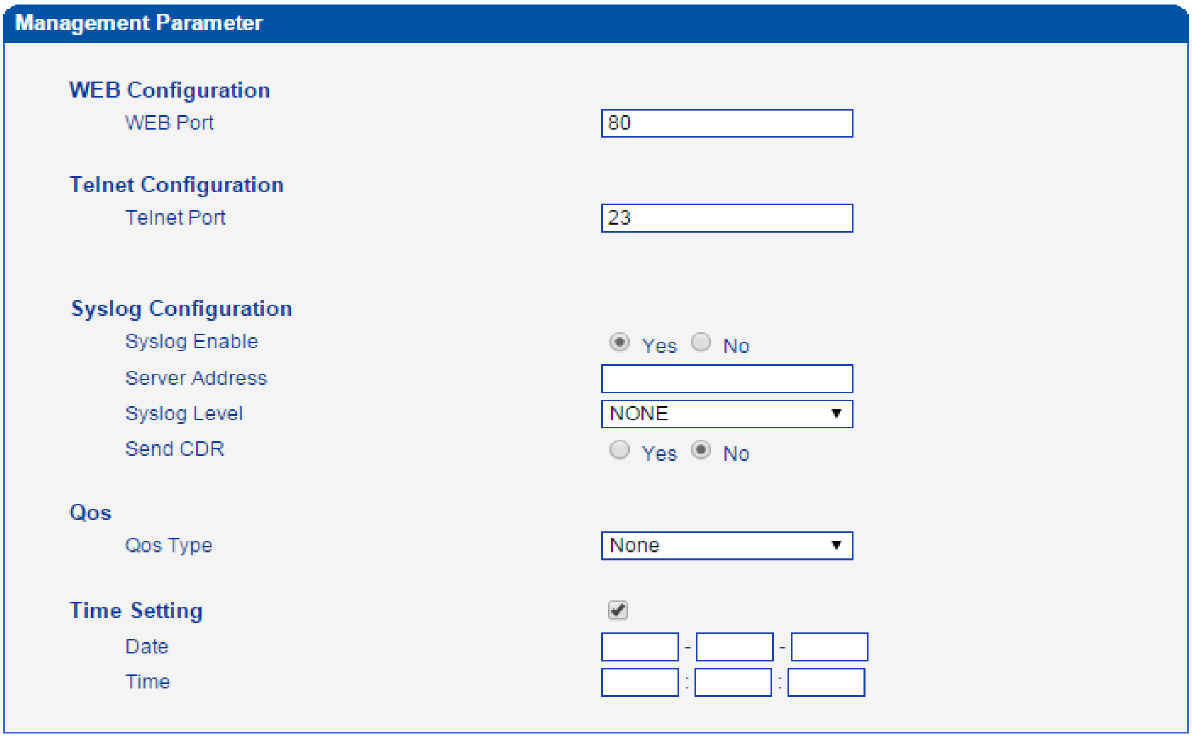
Click **Add**, and the following interface will be displayed.



Parameter	Explanation
Encrypt No.	The No. of this encryption
Description	The description of this encryption
Encrypt SIP	Whether to encrypt SIP message
Encrypt RTP	Whether to encrypt RTP packet
SIP Trunk No.	The No. of the SIP trunk that transmits the SIP message to be encrypted.
Encrypt Mode	Only support VOS RC4 at present
Device ID	The ID of the SIP account to which the SIP trunk belongs

4.16 Maintenance

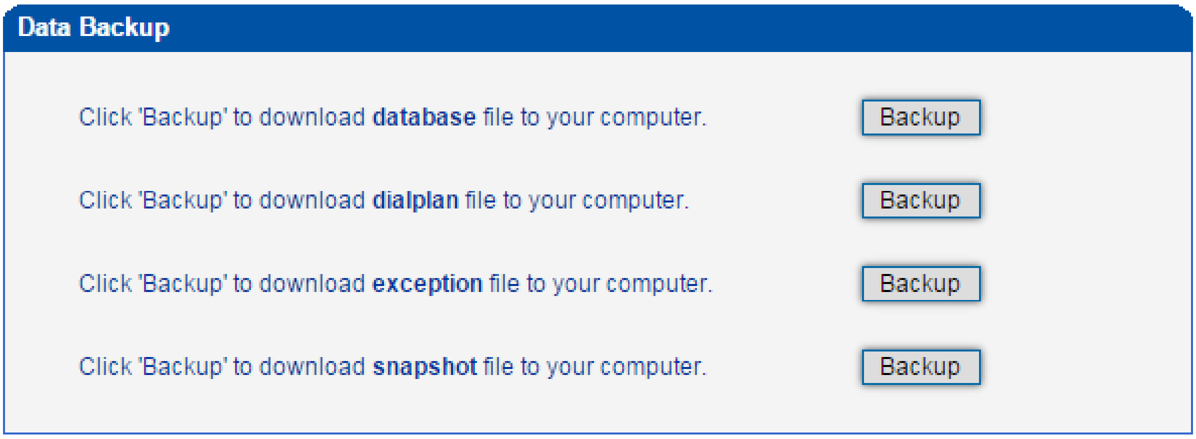
4.16.1 Management Parameter



Belong To	Parameter	Explanation
WEB Configuration	WEB Port	Listening port of local WEB service Default is 80.
Telnet Configuration	Telnet Port	Listening port of local Telnet service Default is 23.
Syslog Configuration	Syslog Enable	Whether to enable Syslog Default is No.
	Server Address	Address to save system logs
	Syslog Level	The system log type. Options include ‘Debug’, ‘Info’, ‘Notice’, ‘Warning’, ‘Error’ and ‘None’.
	Send CDR	Whether to send CDR (Call detail Record).
Qos	Qos Type	Options include ‘None’, ‘TOS’ and ‘DS’. TOS only supports IPv4.
Time Setting	Date	The date displayed on the WEB interface
	Time	The time displayed on the WEB interface

4.16.2 Data Backup

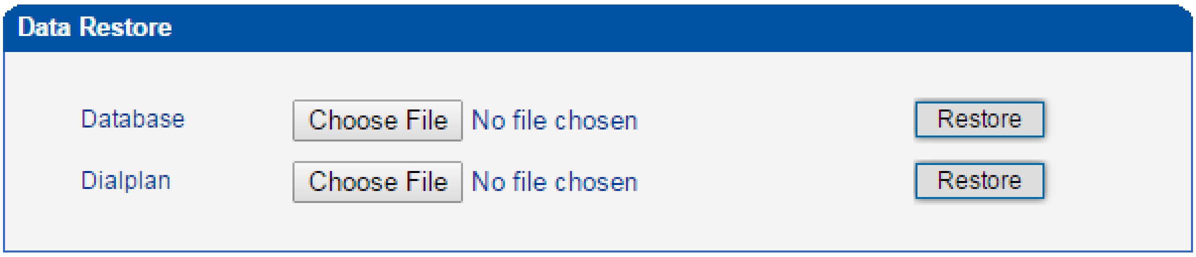
On the **Data Backup** interface, you can click **Backup** to download database file, dialplan file, exception file and snapshot file.



4.16.3 Data Restore

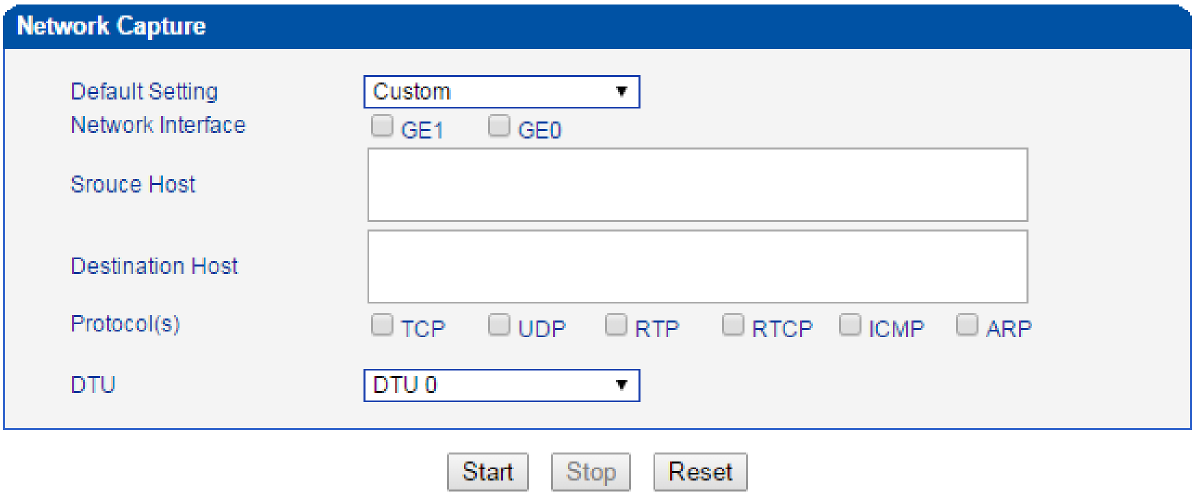
On the **Data Restore** interface, you can restore database and dialplan. If you upload a file that contains default configurations, the MTG2000 will be restored to default configurations. You can also upload a dialplan file to restore dialing rules.

Database herein refers to the database where configuration data are placed.



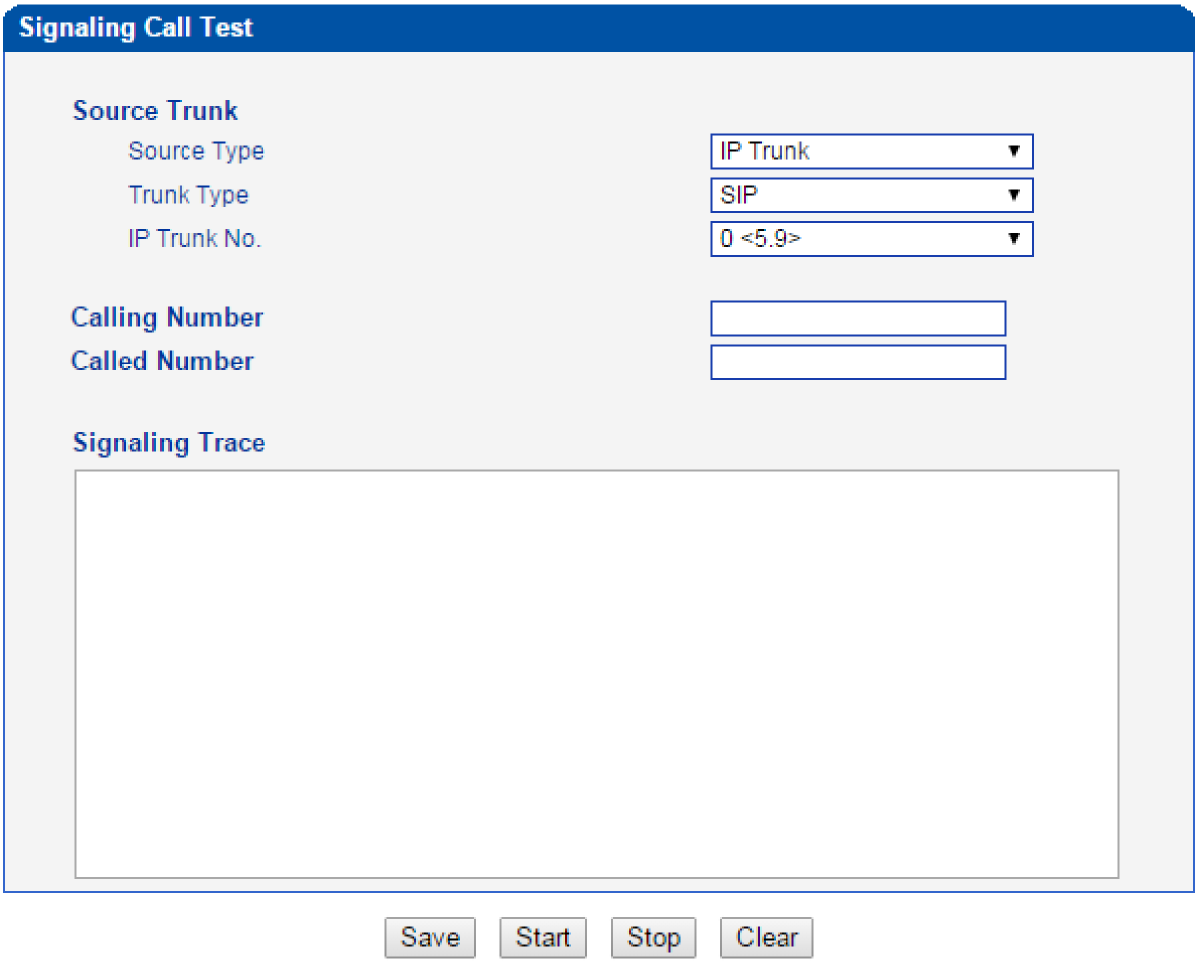
4.16.4 Network Capture

On the following interface, you can capture data packages of the GE0 port or GE1 port of a selected DTU board. You can also set a specific protocol to capture the packages that you want.



4.16.5 Signaling Call Test

On the **Signaling Call Test** interface, you can test whether the signaling of a call is successfully connected. You need to select the source type, trunk type and IP trunk No. of the call, and enter the calling number and called number. If the signaling of a call fails, you can find out where errors have occurred through the messages returned in the **Signaling Trace** box.



4.16.6 ModFile Information

ModFile is voice recognition file which provides the number of total calls recognized, the number of calls currently recognized, the number of the calls recognized in the past one minutes, the number of the calls recognized in the past 15 minutes, the number of the calls recognized in the past one hour and the number of the calls per day.

ModFile Infomation

Customer	Version	Date Built	Sample Count
UCComm	100	20141212	19

Recognize Statistics

RecogCallCnt	CurActiveRecogCnt	AvgRecogCntPerMin	AvgRecogCntPer15Min	AvgRecogCntPerHour	AvgRecogCntPerDay
0	0	0	0	0	0

Sample List

Sample Id	Sample Describe	Sample RecogCnt	RecogRatio
1	rec1	0	0%
2	rec2	0	0%
3	rec3	0	0%
4	rec4	0	0%
5	rec5	0	0%
6	rec6	0	0%
7	rec7	0	0%
8	rec8	0	0%
9	rec9	0	0%
10	rec10	0	0%
11	rec11	0	0%
12	rec12	0	0%
13	rec13	0	0%
14	rec14	0	0%
15	rec15	0	0%
16	rec16	0	0%

Total: 19 Page 1

Refresh Clear

4.16.7 Version Information

On the **Version Information** interface, the version information of the software, database, Web, FPGA, DSP and DTU boards are displayed.

Version Information

File Type	Version	Date Built	Time Built
Software	2.05.01.03	2015-07-14	16:51:46
Database	2.03.04	2015-07-14	15:27:12
Web	2.05.01.03	2015-07-14	11:09:35
FPGA	1.02.09	2015-04-14	19:14:28
DSP	2.01.02	2015-03-19	22:05:56
CardDTU	2.01.11	2015-05-20	16:23:14

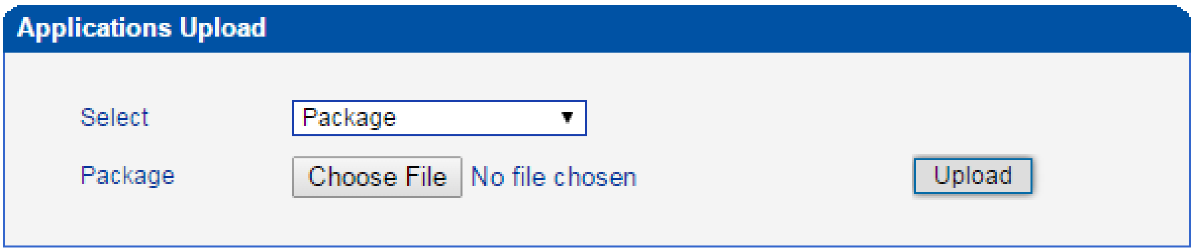
CardDTU Version Info

Slot Num	Current Version
0	v2.01.11
1	v2.01.11
2	v2.01.11
3	v2.01.11
4	v2.01.11

Refresh

4.16.8 Firmware Upgrade

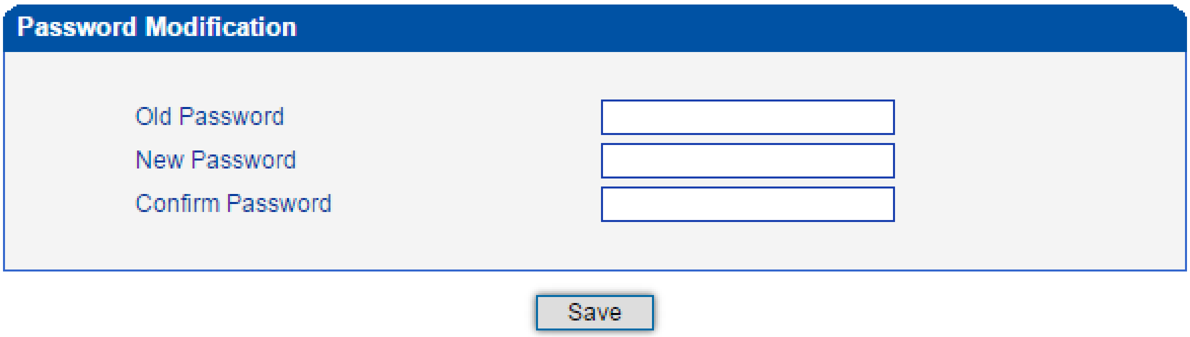
On the **Applications Upload** interface, you can upload files to upgrade the software, the Web, and the Modfile of MTG2000. If you select ‘Package’, it means the upgrading files of the software and Web are packaged and then uploaded.



4.16.9 Password Modification

On the **Password Modification** interface, you can modify password for logging in the MTG2000 device. Default password is admin, so it is advised to modify it for security consideration.

The abovementioned password is also used to log in Web Interface, Telnet and SSH.



4.16.10 Device Restart

Click the Restart button, and you can restart the MTG2000 device.

5 Abbreviation

Abbreviation	Full Name
PRI	Primary Rate Interface
DND	Do-not-Disturb
FMC	Fixed Mobile Convergence
SIP	Session Initiation Protocol
DTMF	Dual Tone Multi Frequency
USSD	Unstructured Supplementary Service Data
PSTN	Public Switched Telephone Network
STUN	Simple Traversal of UDP over NAT
IVR	Interactive Voice Response
ISUP	ISDN (Integrated Services Digital Network) User Part
NTP	Network Time Protocol
PBX	Private Branch Exchange
RTP	Real Time Protocol
RTCP	Real Time Control Protocol
SNMP	Simple Network Management Protocol
SS7	Signaling System Number 7
TUP	Telephone User Part
LOS	Loss of Signal
RAI	Remote Alarm Indicator

AIS	Alarm Indication Signal
LFA	Loss of Frame Alignment
ISDN	Integrated Services Digital Network
CIC	Circuit Identification Code
SPC	Signaling point code
PCM	Pulse Code Modulation
CLI	Calling Line Identification

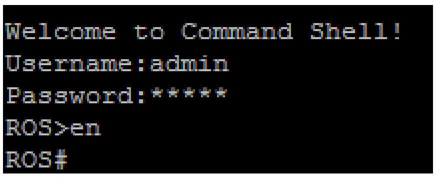
6 Commands

6.1 Commands under en Mode

This section is aimed to guide customers to get more details of MTG2000 gateway through command lines. It introduces the command lines that are commonly used.

6.1.1 Login Command

Run the PuTTY software, and login MTG2000 gateway through Telnet. Enter **username** and **password**, and then run command **en** to activate the privileged commands.



6.1.2 Query IP Address

Enter the command **show int**, IP address, MAC address and Netmask of GE1 are displayed.

```
ROS#show int
eth0        Link encap:Ethernet  HWaddr 00:5A:4E:56:38:04  MAC
                               IP Address                            Netmask
GE1         inet addr:172.16.222.2  Bcast:172.16.255.255  Mask:255.255.0.0

            UP BROADCAST RUNNING MULTICAST  MTU:1400  Metric:1

            RX packets:222562 errors:0 dropped:0 overruns:0 frame:0

            TX packets:71386 errors:0 dropped:0 overruns:0 carrier:0

            collisions:0 txqueuelen:532

            RX bytes:66441300 (63.3 MiB)  TX bytes:23649487 (22.5 MiB)

            Interrupt:11
```

6.1.3 Query Statistics about DTU

Enter the command **show card**, and statistics about DTU are displayed.

```
ROS#show card
CardNum  RemoteMAC          ConnectState  LinkOk  queue  RegCnt   LastRegTick  CurTick   LastOffTick  LinkFailCnt  Version
  0      00-11-22-33-44-01  Active        OK      0      1        10309        2347576   0            0            v2.01.11
  1      00-11-22-33-44-11  Active        OK      0      1        10786        2347576   0            0            v2.01.11
  2      00-11-22-33-44-21  Active        OK      0      1        11262        2347576   0            0            v2.01.11
  3      00-11-22-33-44-31  Active        OK      0      1        11739        2347576   0            0            v2.01.11
  4      00-11-22-33-44-41  Active        OK      0      1        12214        2347576   0            0            v2.01.11
```

6.1.4 Query DSP Information

Enter the command **show dsp info**, and DSP information is displayed.

```
ROS#show dsp info
Dsp No:0,         Status:DSP_LOADING_INIT_SUCCESS
                  Dsp Cap:2480
                  Dsp Mac:00-11-22-33-44-02
               Ip Address:172.30.20.4
              Arm version:Branch_7_25_K2
          Load Fail Count:0
     Cmd NoResponse Count:0
 ==========================================================
Dsp No:1,         Status:DSP_LOADING_INIT_SUCCESS
                  Dsp Cap:2480
                  Dsp Mac:00-11-22-33-44-03
               Ip Address:172.30.20.4
              Arm version:Branch_7_25_K2
          Load Fail Count:0
     Cmd NoResponse Count:0
 ==========================================================
Dsp No:2,         Status:DSP_LOADING_INIT_SUCCESS
                  Dsp Cap:2480
                  Dsp Mac:00-11-22-33-44-12
               Ip Address:172.30.20.4
              Arm version:Branch_7_25_K2
          Load Fail Count:0
     Cmd NoResponse Count:0
 ==========================================================
```

6.1.5 Query CPU Performance

Enter the command **show perf,** the CPU performance is displayed.

```
ROS#show perf
    performance now :0
    performance 5s  :0
    performance 60s :0
    performance 600s:0

    performance now user(%%):0
    performance now system(%%):0
```

Performance now	CPU load at current time
Performance 5s	Average CPU load in recent 5 seconds
Performance 60s	Average CPU load in recent 60 seconds
Performance 600s	Average CPU load in recent 600 seconds

6.1.6 Query SS7 Trunk Status

Enter the command **show ss7 sta**, and the status of SS7 link is displayed.

```
ROS#show ss7 sta
  grpId  linkState  mainLink  backupLink  currentCalls  maxCalls  failCalls  tot
alCalls  failRatio
--------------------------------------------------------------------------------
```

6.1.7 Query SS7 Link Statistics

Enter the command **show ss7 link**, and statistics about SS7 link are displayed.

```
ROS#show  ss7 link
  linkId hdlcNo type    revErrs cc       rc    lsc            iac      poc         txc   aerm    suerm
 daedt daedr
----------------------------------------------------------------------------------------------------
```

6.1.8 Query SS7 Call Statistics

Enter the command **show ss7 call**, and statistics about SS7 calls are displayed.

```
ROS#show ss7 call
grpId: interface ID   userId: CC call ID   callId: SS7 call ID
online total calls: 0
```

6.1.9 Query SS7 Errors

Enter the command **show ss7 err**, and errors about SS7 trunks or SS7 links are displayed.

```
ROS#show ss7 err
error cnt:14
[07-15 11:06]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:06]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:07]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:08]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:08]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:08]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:08]erro - hard_init()->tm_connect_ss7_e1 failed!
[07-15 11:08]linkId[2] erro - ###-- Error: Abnormal Flag -> 127 <= 21 <= 127
[07-15 11:08]linkId[2] erro - ss7_pkt_discard()->fsn error! previous:51 ,new:127 len:6
```

6.1.10 Query PRI Trunk Status

Enter the command **show q931 sta**, and statuses of PRI trunks are displayed.

6.1.11 Query PRI Link Statistics

Enter the command **show q931 link**, and PRI link statistics are displayed.

6.1.12 Query PRI Call Statistics

Enter the command **show q931 call**, and statistics about PRI calls are displayed.

6.1.13 Query Packet Statistics of HDLC Channel and Related Error Codes

Enter the command **show mcc x** (x refers to the port No. of HDLC channel), and the packet statistics and error codes (if there are any) of the HDLC channel are displayed.

6.1.14 Query Status of E1 Port

Enter the command **show e1 x** (x refers to the E1 port No.), and the status of the E1 port is displayed.

6.1.15 Query Statistics of All Calls

Enter the command **show cc calls**, and the statistics of all calls are displayed.

6.2 Commands under config Mode

6.2.1 Login Commands

Welcome to Command Shell!
Username:admin
Password:*****
ROS>en
ROS#
ROS#^config
ROS(config)#

6.2.2 Other Commands

Used For/To	Command
Query version information	ROS(config)# load show
Call tracing	ROS(config)#deb cc detail all ROS(ada)#turnon 27
SIP signal tracing	ROS(config)#deb sip msg all ROS(ada)#turnon 71
Query SS7 Signal	ROS(config)#deb ss7 <lnkId> <level> ROS(ada)#turnon 96
Query PRI Signal	ROS(config)#deb q931 detail

	ROS(ada)#turnon 64
Restart MTG2000	ROS(config)#reset gmpu [ipaddr]

6.3 Commands under ada Mode

6.3.1 Login Commands

Welcome to Command Shell!
Username:admin
Password:*****
ROS>en
ROS#
ROS#^ada
ROS(ada)#[119-17:35:18:040]ADA CONNECTED …,WELCOME!
ROS(ada)#

Used For/To	Command
Query the records about exceptions or errors	ROS(ada)#cmd 3 30 0
Query the records about exceptions or errors before the restart of MTG2000	ROS(ada)#cmd 3 30 1
Disable the printing of SIP messages	ROS(ada)#turnoff 71
Disable the printing of SS7 messages	ROS(ada)#turnoff 96
Disable the printing of PRI messages	ROS(ada)#turnoff 64
Disable the printing of CC messages	ROS(ada)#turnoff 27